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To Evaluate the Performance based on Delay and Jitter of Wired VoIP using a network of assorted number of Routers and Hosts.

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***Abstract*— In this paper, we study experimentally the performance of VoIP on wired communication with varied combination of Routers and hosts. We concentrate on two factors of QoS – Delay and Jitter. NS3 simulator is used to study the pattern and effect on the two factors by varying the number of Routers and hosts in the given setup. By this study the user can decide on what combination is best for his application and what factors can be compromised and which factors can be enhanced. Also, we have concluded by giving a glimpse of using the results of this experiment for QoE. The study can further be extended to including all the factors that come under QoS.**

Keywords*— TCP, UDP, VoIP, QoS, NS3

I. Introduction

Looking at the current trend of wireless communication, you would question why wired? Why TCP? Even though combination of wireless and UDP type of commiuncation being widely used in todays' communication world, there are applications which need wired and TCP. Although UDP has its advantages, we need to decide which transport layer protocol to use on the basis of application. TCP is best suited in applications where high reliability in sending data packets is needed. Other major advantages of TCP are – in-order packet delivery, data remains intact, error checking & error recovery and acknowledgement for packets received.

VoIP is now important in communication for making phone calls over internet. i.e. sending voice data packets over internet. VoIP uses the network that uses packet switching over the customary telephonic communication – Public Switched Telephone Network (PSTN). PSTN is a circuit switching technology i.e. our landlines, that needs a dedicated line for making a telephone call. This is major hurdle in packet based communication. VoIP is implemented by sampling the analog voice signals, encoded by using an encoding technique – codec, this data is put into packet format – encapsulation of data by IP header, then transmitted over Internet in the same way as data packets are communicated. VoIP has become simpler to use due to technological advancements in the hardware used & protocol development in this area also tackled the packet formation, error checking and security issues to a greater extent. One of the biggest advantage of VoIP is making long distance calls at very less cost and there is no need to use different phone numbers in different countries. [1],[2],[4]

The second section of paper covers overview of QoS in VoIp. Why we have chosen NS3 over NS2 is discussed in the third section. In the fourth section of the paper we discuss the experimental results of a network with different number of routers and hosts for wired VoIP using TCP as transport layer protocol. For evaluating performance of this network we use QoS. QoS is used to ensure high-quality performance which can be applied for critical applications. QoS depends on network characteristics like – Delay, Jitter, Packet Loss, Echo and Throughput.[3],[5] And in the last setion we confer the experimental study and evaluate the results.

II. Quality of Service (QoS) in VoIP

Quality of Service gives a guideline to the users about quality of the service they will be using for their application. It is important not only from customer point of view but also from network administrator point of view. The network administrator can make use of different network characteristics to mould the factors affecting them to achieve high quality. Following are the characteristics that make up the QoS :

Performance Calculations

Throughput

Throughput is the average rate of successful packet pass over the communication link. It measured in bytes/second. In the trace file, "r" represents receive in normal.

Instantaneous Throughput= bytes received / one second

$$\text{Average Throughput} = \frac{\text{Total number of bytes received}}{\text{Simulation runtime}} \quad (1)$$

Throughput in VoIP depends on the number of users simultaneously using the facility and the codec used.

Packet loss

Packet loss is defined as when one or more packets of data moving through a network fail to reach its destination. Packet loss is mainly caused by network congestion. Packet loss is calculated as loss of packets as opposed to number of packets sent. The cumulative packet loss is given as follows:

$$\text{Cumulative packet loss} = \frac{\text{Total packets dropped}}{\text{Simulation runtime}} \quad (2)$$

The requirement of packet loss in VoIP is less than 1% to ensure minimal audible error. This depends on the codec used by VoIP and the common codec used by VoIP is G.729**.**

Latency

Latency is a measure of time delay happened in a system. In this simulation delay is measured in taking the time difference between when packet is sent from source node and when it reaches its destination. Every packet, to be identified in the network, is assigned with a packet identification number. This number will recognise the packet that will travel through the network through every node. Instantaneous latency= Receive time destination node – send time of source node.

$$\text{Average Latency} = \frac{\text{cumulative total of instantaneous latency}}{\text{Simulation runtime}} \quad (3)$$

Jitter

Jitter is an informal name given to IP packet delay variation; it is commonly used in electronics and telecommunication. This is a unwanted diversion from cyclic nature of signal in the network.

Instantaneous Jitter = Current Latency – Previous Latency

The average jitter is calculated as follows:

$$\text{Average jitter} = \frac{\text{sum (current latency-previous latency)}}{\text{Simulation runtime}} \quad (4)$$

Jitter has a negative effect on the voice quality which is not suitable to VoIP. Also, jitter may result in loss of voice packets leading to disrupted communication in telephonic communication or missing words in the telephonic call. To have minimal voice quality jitter must be less than 100ms.

Delay

Network delay is one of the important performance characteristic of a network. The delay is defined as amount of time taken for the packet to travel across the network to reach from one host to another host at the destination. Delay is measured in milliseconds or nanoseconds. Delay is dependent on distance between the two communicating nodes, speed of the communication link and network traffic. [1],[3]

III. NS3 SIMULATOR FOR EXPERIMENTAL STUDY

Simulation is best method through which researchers can study a scenario without involving hardware. Simulation imitates the real world scenario which is to be implemented. When a researcher is studying a particular environment it is better to implement it using a simulator and confirm that the results are as expected. After confirming the workability of the system, the actual hardware can be employed. This will not only save time but also cost which will be beared by the researcher if the scenario does not work. Simulator gives the flexibility to make any changes in the system without worrying about the hardware cost.

Network simulator is boon to the computer network domain. It gives an environment where in the real world network can be imitated. Network Simulator is not only useful to academic researchers but also to industrial development and network administrators. Network simulator is best used to design, implement, simulate, analyse and conclude about performance of different networks, protocols, different topologies and network scenarios. Researchers can experimentally study a given network design from QoS and QoE point of view. These studies help them to understand the implement ability of a network and also confer the quality of the network by studying the factors of QoS – Delay, Jitter, Packet loss and Throughput.

Network simulator is a discrete event simulator. It is useful in efficiently management of cost for network design, provides ease of any addition and modification of network structure to existing network, enables user to model different network topologies, model flow of packets (network traffic) between nodes, provides animated view of packet flow to better visualise the network and finally gives output in the form of performance attributes.[6],[7], [9]

Following is the comparison of NS2 and NS3 [8] :

Table 1 : Comparison of NS2 and NS3

NS2	NS3
Was released in 1996	Was released in 2008
It is supported by DARPA, VINT,SAMAN, NSF AND CONSER	It is supported by NSF and INRIA
It is built in C++, but scripted in OTcl	It is built in C++, but scripted in Python

Simulation output is given as NAM	Simulation output is given as NS-3-viz, pyviz, NAM
Models used – **Application Layer :** Ping, vat, telnet, FTP, multicast FTP, HTTP, probabilistic and trace-driven traffic generators, webcache **Transport Layer :** TCP(many variants), UDP, SCTP, XCP, TFRC, RAP, RTP Multicast: PGM, SRM, RLM, PLM **Network Layer :** Unicast: IP, MobileIP, generic dist vector and link state, IPinIP, source routing, Nixvector Multicast SRM, generic centralized MANET: AODV, DSR, DSDV, TORA, IMEP **Link Layer:** ARP, HDLC, GAF, MPLS, LDP, Diffserv Queuing: DropTail, RED, RIO, WFQ, SRR, Semantic Packet Queue, REM, Priority, VQ MACs: CSMA, 802.11b, 802.15.4(WPAN), Satellite Aloha **Physical Layer :** TwoWay, Shadowing, OmniAntennas, EnergyModel, Satellite Repeater	**Models used –** Application Layer : OnOffApplication, asynchronous sockets API, packet sockets **Transport Layer :** UDP, TCP **Network Layer** : Unicast IPv4, global static routing Multicast static routing MANRT:OLSR Network Layer : PointToPoint,CSMA.802.11 MAC low and high and rate control algorithm **Physical layer :** 802.11aFriis propagation loss model, log distance propagation loss model, basic wired(loss, delay)

Output of NS3 : following are some snapshots of output of NS3

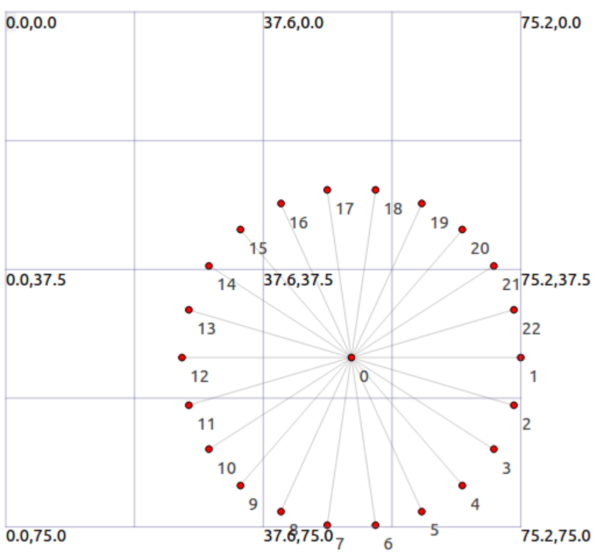

Figure 1: NS3 animator output

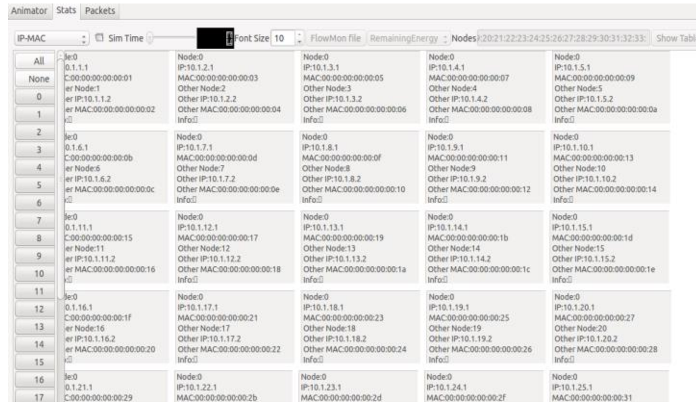

Figure 2 : NS3 Statistical Output


-<FlowMonitor>
-<FlowStats>
<Flow flowId="1" timeFirstTxPacket="+1000000000.0ns" timeFirstRxPacket="+1002092800.0ns" tin
timeLastRxPacket="+10004172798.0ns" delaySum="+575529597.0ns" jitterSum="+444803.0ns" lastI
rxBytes="46706" txPackets="250" rxPackets="250" lostPackets="0" timesForwarded="0"> </Flow>
<Flow flowId="2" timeFirstTxPacket="+1000000000.0ns" timeFirstRxPacket="+1002092800.0ns" tin
timeLastRxPacket="+10004172798.0ns" delaySum="+575529597.0ns" jitterSum="+444803.0ns" lastI
rxBytes="46706" txPackets="250" rxPackets="250" lostPackets="0" timesForwarded="0"> </Flow>
<Flow flowId="3" timeFirstTxPacket="+1000000000.0ns" timeFirstRxPacket="+1002092800.0ns" tin
timeLastRxPacket="+10004172798.0ns" delaySum="+575529597.0ns" jitterSum="+444803.0ns" lastI
rxBytes="46706" txPackets="250" rxPackets="250" lostPackets="0" timesForwarded="0"> </Flow>
<Flow flowId="4" timeFirstTxPacket="+1000000000.0ns" timeFirstRxPacket="+1002092800.0ns" tin
timeLastRxPacket="+10004172798.0ns" delaySum="+575529597.0ns" jitterSum="+444803.0ns" lastI
rxBytes="46706" txPackets="250" rxPackets="250" lostPackets="0" timesForwarded="0"> </Flow>
<Flow flowId="5" timeFirstTxPacket="+1000000000.0ns" timeFirstRxPacket="+1002092800.0ns" tin
timeLastRxPacket="+10004172798.0ns" delaySum="+575529597.0ns" jitterSum="+444803.0ns" lastI
rxBytes="46706" txPackets="250" rxPackets="250" lostPackets="0" timesForwarded="0"> </Flow>
<Flow flowId="6" timeFirstTxPacket="+1000000000.0ns" timeFirstRxPacket="+1002092800.0ns" tin
timeLastRxPacket="+10004172798.0ns" delaySum="+575529597.0ns" jitterSum="+444803.0ns" lastI
rxBytes="46706" txPackets="250" rxPackets="250" lostPackets="0" timesForwarded="0"> </Flow>
<Flow flowId="7" timeFirstTxPacket="+1000000000.0ns" timeFirstRxPacket="+1002092800.0ns" tin
timeLastRxPacket="+10004172798.0ns" delaySum="+575529597.0ns" jitterSum="+444803.0ns" lastI
rxBytes="46706" txPackets="250" rxPackets="250" lostPackets="0" timesForwarded="0"> </Flow>
<Flow flowId="8" timeFirstTxPacket="+1000000000.0ns" timeFirstRxPacket="+1002092800.0ns" tin

Figure 3 : NS3 FlowMonitor Output

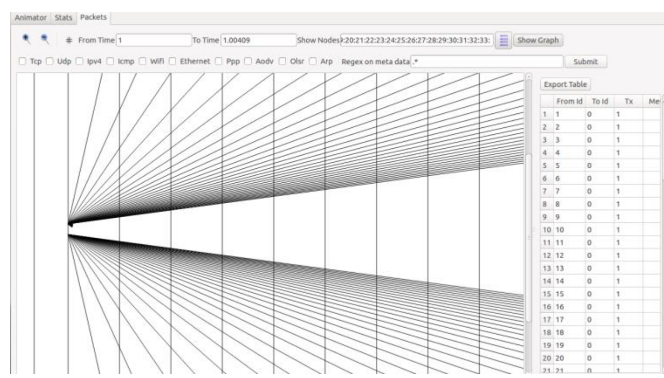

Figure 4: NS3 Packet Information

IV. EXPERIMENTAL SCENARIOS ITS RESULT AND DISCUSSIONS

We have worked with VoIP on wired communication with TCP as transport layer protocol. Different combinations of Routers and hosts were made for comparative study. Example of one of the network topologies is given :

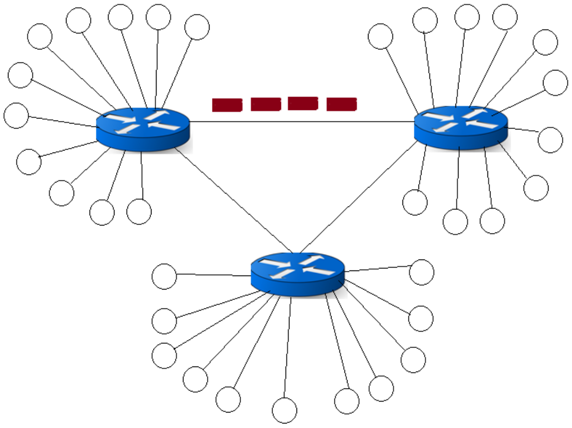

Figure 5: Network Topology Example

The results were observed as given in the table:

Table 1.2 : Comparison of QoS Factors – Jitter & Delay

No. of Routers, Hosts	Jitter (nS)	Delay (nS)
2,10	777300	1902342
3,33	444803	2086399
5,40	1954	2192364

Comparison of the factors is shown in the graph below:

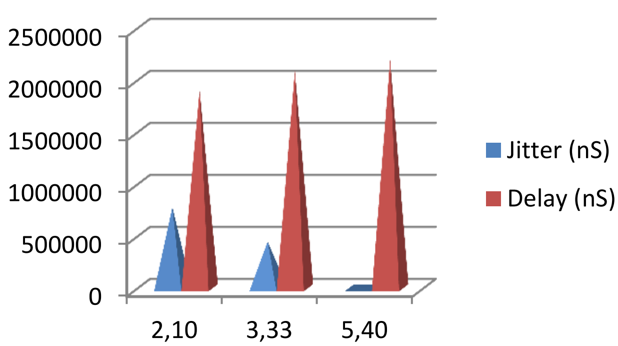

Figure 6 : Graphical Representation for Jitter & Delay

From the diagram it is observed that as number of hosts and routers go on increasing the delay increases but the jitter decreases.

V. Conclusion

VoIP communication is popularly used for making calls over internet. The greatest concern is poor quality of voice, which leads to unrecognisable words or complete loss of voice packets. QoS is maintained by jitter within 100ms, latency must not be more, packet loss must be less than 1% to avoid error in voice, and throughput must be high. From the experiments done it is concluded that as the number of nodes increase the jitter increases and tends to 100ms reducing the quality of voice. Thus to keep jitter to cross this value we have to ensure that there are optimal number of nodes in the path of communication.

QoS measures the values of performance of any network, whereas QoE is subjective to the experience of the user while working with a network. Customer or end users are more interested in usability of the network in given working conditions, which is given by QoE. Thus QoE is important and more useful to the end user. In our study we can conclude that wired VoIP quality can be maintained if number of hosts in the path of communication is such that jitter is maintained below 100ms. Thus choosing a optimal path becomes necessary.

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