# The Contributory Effect of Latency on the Quality of Voice Transmitted over the Internet

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**Abstract**: Deployment of Voice over Internet Protocol (VoIP) is rapidly growing worldwide due to the new services it provides and cost savings derived from using a converged IP network. However, voice quality is affected by bandwidth, delay, latency, jitter, packet loss e.t.c. Latency is the dominant factor that degrades quality of voice transfer. There is therefore strong need for a study on the effect of Latency with the view to improving Quality of Voice (QoV) in VoIP network. In this work, Poisson probability theorem, Markov Chain, Probability distribution theorems and Network performance metric were used to study the effect of latency on QoS in VoIP network. This is achieved by considering the effect of latency resulting from several components between two points in multiple networks. The NetQoS Latency Calculator, Net-Cracker Professional® for Modeling and Matlab/Simulink® for simulating network were tools used and the results obtained compare favourably well with theoretical facts.

**Keywords**: VoIP, Latency, QoV, Network, Modeling

## 1 Introduction

The use of voice over Internet Protocol (VoIP) is the process of transmitting voice as data over the Internet equipped with coding/decoding devices (CODEC series) that helps in converting sound waves into digital packets so that the packets can be transmitted across a digital line and at the other end decode back to sound. Basically, it involves converting analog voice signals to digital format in form of a data through the Internet Protocol (IP) [1]. The Public Switched Telephone Networks (PSTN technologies) architecture built primarily for voice is not flexible enough to carry data hence this is largely incompatible with the convergence of data/voice/video. Evolution of audio coding technologies has allowed voice to be transmitted over data links that resulted to emergence of VoIP which has been most sought after device for companies looking to take advantage of IP services. There is a cost benefit to be derived from using this new technology which is also prone to challenges.

The main challenge is improving the Quality of voice (QoV) in VoIP to avoid degradation of the service. The fidelity of Legacy telephone has not been achieved by VoIP yet hence the infrastructure of this new technology must be able to support Latency that degrades voice transmitted over the Internet. Latency stands as the delay that occurs when a packet crosses a network connection, from sender to receiver i.e. end-to-end-delay that occurs in information exchange between two nodes [2], [3]. There are many sources of delay in VoIP systems that add up to the total latency. Among these are the following: Algorithmic delay which is related to the speech codec used; Processing delay which is related to the signal processing performed and depends on the available CPU performance; Hardware delay and network delay caused by physical delay in the transmission lines; buffers in Routers which is time varying delay. Jitter, known as time variable delay as packets streams travel through an IP network in different paths and resulted in varying arrival time. Too much traffic in the network causes packets drop that result in packet loss [4].

It is important to predict expected voice quality under various network conditions and traffic loads so that steps can be taken in advance to prevent potential problems. In this case, the quality of service (QoS) must be capable of prioritizing traffic types; interpreting traffic types (applications running over IP) and then conveying them over the network. Research efforts in the area of studying the effect of Latency on QoS specifically in VoIP network performance were carried out by several researchers which include: Agnihot et al, May et al, Naser et al, S. Sahu et al, Antos et al to mention but a few.

Therefore it becomes necessary to conduct a study on the effect of Latency in VoIP network with the primary objective of improving on Quality of voice transmitted over the Internet toward achieving optimum toll voice quality and high network throughput.

## 2 Related Works

Research efforts in this area have been reported by several researchers. [5] used time scale modification algorithm, to handle losses that impact quality of voice delivery and jitter but resulting queuing delay from this packet arrival delay variation which cause packet drops after time to live (TTL) must have expired was not considered. [6] used a given buffer size, and two-bit architecture traffic load to estimate the expected delay of packets for constant bit rate and this aim was achieved.

Path Switching carried out by [7] showed that path quality estimator based on International Telecommunication Union (ITU-TE)-model was developed for voice quality assessment and an application driven path switching ‘algorithm’ ‘that’ applied the time scales over which path switching decisions are made to voice quality. Though, network emulation and

experiments over a wide area indicated that with sufficient path diversity, path switching can yield noticeable improvement in voice quality by utilizing the inherent path diversity of the internet. The limitation of path switching is that it can also introduce performance degradations through transient disruptions that may results due to switching from one path to another, particularly when the network path differs significantly in their propagation delay and topology.
Integrated Service/ RSVP architecture was influenced by the work of [8] as a signaling protocol for application to reserve resources for aggregation of flows to set up explicit routes (ERs) with QoS requirements. However the requirement on routers is high. All routers must have RSVP, packet scheduling and admission control. This is tasking to the internet core. Despite all these research works the problems of call quality in VoIP remains apparent and the need for further research on effect of latency with the view to minimize the voice degradation in VoIP.

# 3 Methodology

In this work Poisson probability theorem, Markov Chain and Probability distribution theorems were used to study the effect of latency on QoS in VoIP. This was achieved from the concept of the effect of delay components between end-to-end networks in a multiple networks environment. NetQoS Latency Calculator [9], NetCracker Professional, Matlab/Simulink and NS-2 were tools used in performing the network validation experiment.
In order to achieve accurate results, the conventional legacy data network was first examined to determine the characteristics of the network. The network monitoring tool used for the parameter measurement was the Solar wind Simulation Software® [10] capable of given detail result of the network performance.
The software were modified to measure the latency, throughput, retransmission time (RTT), and packet loss for a single user, then to 30 users along the link between Lagos and the Abuja contact centre in Nigeria, with the results taken at the Cisco® 7201 border router of the Lagos contact centre. The codec for compression used were the uncompressed 64Kbps G.711 protocol, while the compression on the WAN link was G.729 protocol with 8Kbps of bandwidth.

# 4 Delay Components

The time it takes for a voice to be digitized, packetized transmitted, routed and buffered over the internet is known as delay. It poses one of the major threats to QoS mechanisms. Propagation and processing delay are directly responsible for the major factors that affect quality of service in VoIP. There are several types of delays in an IP network which differ from each other as to where they are created. The delay components are classified based on the place of their creation, mechanism or some other attributes [11] as shown in Figure 1. The VoIP delay components are coder and packetization delay originated from the end-to-end transmission through the communication channel, thus their components affect the result of voice packets in the network.

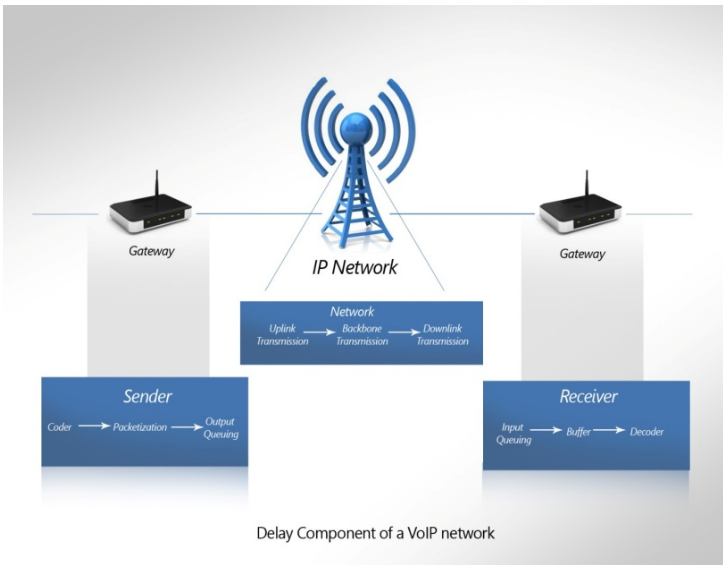


Figure 1. Delay Components in VoIP Network of a Communication Channel.

The detailed description of individual delay components as it goes with the VoIP network are as follows: Queuing delay, serialization delay, propagation delay in transmission network, access / codec delay and packetization delay.

# 5 Modeling Network Infrastructure of Voice over IP

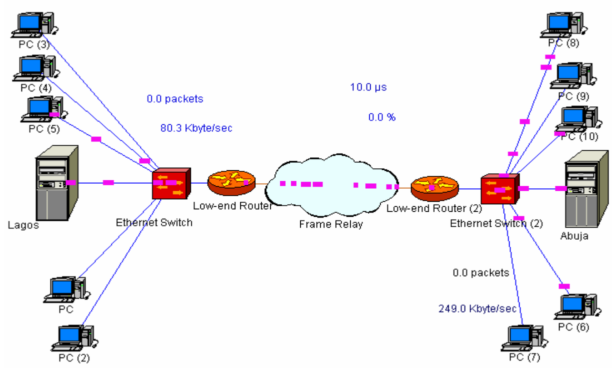


Figure 2. Communication Network of Multiple Nodes.

Communication network of multiple nodes shown in Figure 2, illustrates the developed mathematical model of the research work. The communication network channel is established between two Local Area networks with multiple nodes transferring voice packets along the network media. The transmitting end and the receiving end are depicted in the model and the scenario presents real life network environment as a test-bed for the research using Net-Cracker Professional simulation software®.

# 6 Mathematical Model

We consider a large network with many stations able to communicate using CSMA/CD protocol. Assume infinite number of host with packet arrival in Poisson stream rate, $\sigma < 1$,for time, t, in an ascending order t = 1, 2 … We applied traditional Poisson model, with the assumption that a large number of packets can arrive at a buffer with a Poisson distribution. The probability Pn (t) of exactly n packets arriving during a time interval of length t is given by $P_n(t) = \frac{(\sigma t)^n e^{-\sigma t}}{ni}$ *where* $n = 0, 1, 2,$ σ is the average packet arrival rate [12].

This shows that the probability that for a Poisson data flow stream in a given small network area with change in time (ΔT) an event will occur. i.e. the amount of packets which are necessary to be sent at an interval of time, ΔT will be a Poisson data flow. Each station transmits one packet only. It is considered that the time slot for successful transmission of a packet is given as: *(t, t + 1).*

With the above mathematical expression we come into conclusion that the total delay which comes from the backlog of the systems during data/voice performance within the channel can be considered as the net delay and expressed as follows:

***Net delay = Propagation delay*** $(T_p)$ ***+ Serialization Delay*** $(T_s)$ ***+ Queue Delay*** $(T_q)$

Thus, the total delay on the network can well be represented as:

$T_{total} = T_A + T_D \in (T_p + T_s + T_q)$, where $T_A$ = Encoding time, $T_D$ = Delay,

The derived model for the net latency in the $n^{th}$ network is given as [13]:

$$Latency\_n = \frac{\sum x_n}{0.667c} + \frac{\sum N_n \mu}{(\mu - \lambda_n)B} + \frac{\sum \lambda_n^2}{(\mu - \lambda_n)B\mu}$$

## 6.1 Results and Analysis of the Network Mathematical Model

From the mathematical model, the net delay was obtained by considering the total delay experienced by the network i.e. *propagation delay, serialization delay, and queue delay* [13]. Data generated from the mathematical model and real world network setup were analyzed with the corresponding result obtained. In the relationship between responses of packet sizes against the throughput of the network, Figure 3, it was observed that the increment noted was on a gradual increase which suggests that the network has not reached a congested state.

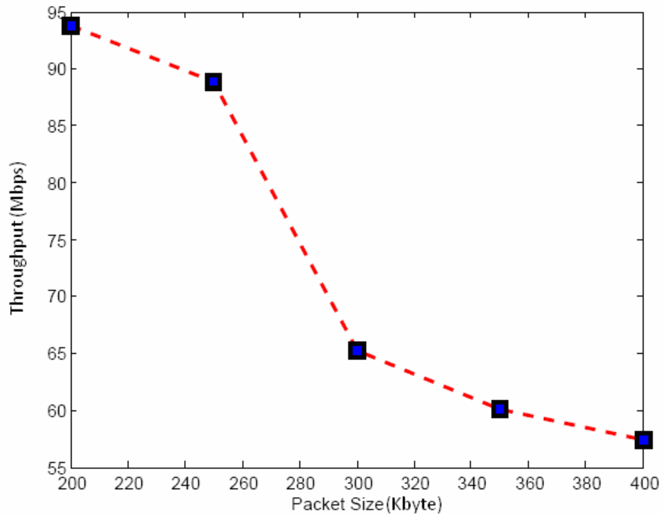


Figure 3. Graph of response of packet sizes against the network throughput

An increase in packet sizes result into the reduction in throughput, A sharp change was observed in the throughput when the packet size was increased to 300Kbytes, having a reduction in value from 89Mbps to 65Mbps.With the utilization value of ρ=0.4 in Figure 4, the packet loss shows a steady increment, though mostly negligible, however it is of considerable value when compared in respect of the packet size between 200 and 400Kbytes. A similar situation is recorded when the utilization factor becomes ρ=0.9 with the Bandwidth Capacity of 100Mbps.

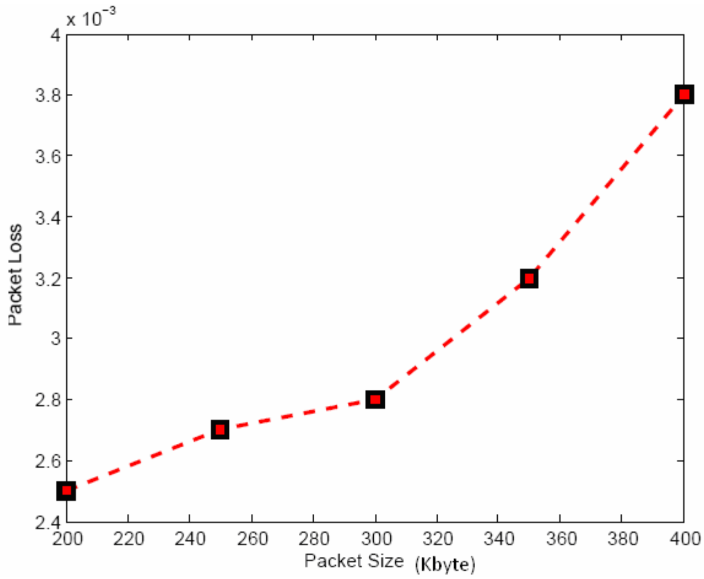


Figure 4. Graph of packet size against packet loss with utilization factor of 0.4 with 100Mbps bandwidth.

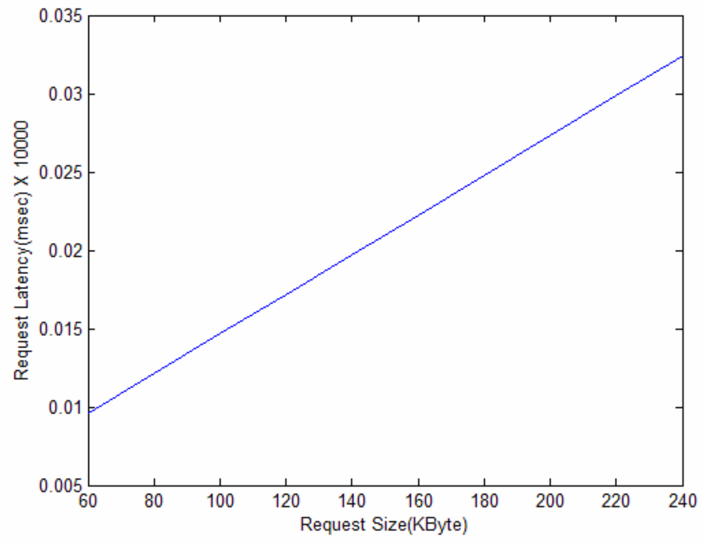


Figure 5. The graph of packet request size against Latency

The Figure 5 shows the graph of request size against overall request latency. As the packet size transmitted increases, so does the latency accordingly hence, the linear relationship. However it is obvious that as packet size increases latency increases such that above 200msec voice becomes impaired degraded. The optimum was attained with packet size of 190Kbyte and request latency of 0.025 x $10^4$msec. Beyond this point, toll call quality starts experiencing degradation well notice in Figure 6 graph of packet request size against request Latency indication of decline in network performance.

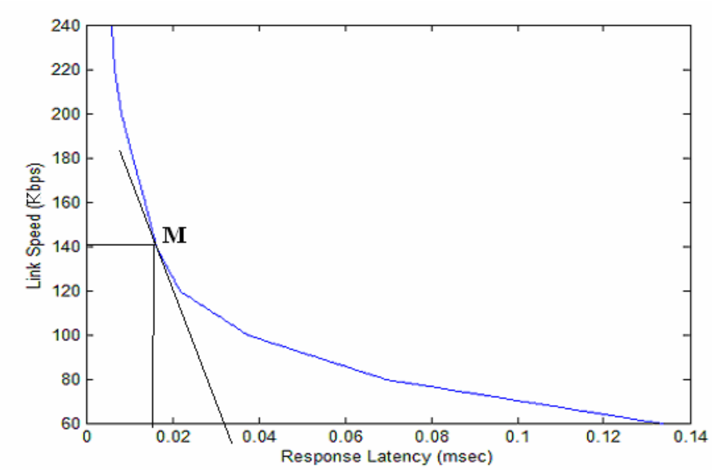


Figure 6. The graph of response Latency against Link Speed

Considering the result in Figure6, as the link-speed drops, latency increases gradually until latency get to the threshold [14] of the throughput on the point M of the tangent (142Kbps and 0.18 x $10^3$msec). At this point, latency increases exponentially to approach its peak of (180msec.) for the workload. From the result as the link-speed drops further irrespective of the codec used, prioritization method deployed; with a lot of packet drops at this region, latency tends to a high value that degraded the voice quality. At this point, no amount of bandwidth will minimize the latency for toll call quality to be achieved. Also, we observed that the packet loss becomes so much that throughput has no effect on the degraded voice quality hence the asymptotic increase of latency which results into voice impairment. This follows that at about 142 Kbps no amount of throughput of the network can impact latency positively.

# 7 Conclusion

Use of developed mathematical model (Poisson probability theorem and Probability distribution theorems) in analyzing and studying the contributory effect of latency on the quality of voice transmitted over the internet. Improved QoS was achieved from the research result:

Toll quality of 5.0 (MOS) => $0 \leq$ Latency $\leq$ 182kbps. By controlling the workload and bandwidth, the efficiency of the network was achieved with minimum latency on the network.

Experimental results obtained correspond to the simulated modeling. Provide fundamental insight to the contributory effect of latency on the amount of bandwidth can change the network performance when the system gets to the threshold point "$0 \leq$ Latency $\leq$ 182kbps" as obtained in our result.

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