

1.1 Background

Voice over Internet Protocol (VoIP) is a technology that allows users to make telephone calls using a broadband Internet connection instead of an analog phone line. VoIP holds great promise for lowering the cost of telecommunications and increasing the flexibility for both businesses and individuals [1]. VoIP leverages existing IP-based packet-switched networks to replace the circuit-switched networks used for voice communications since the invention of the telephone. A VoIP application is gaining an ever increasing popularity in the Internet community [2].

The VoIP protocol suite is generically broken into two categories, control plane protocols and data plane protocols. The control plane portion of the VoIP protocol is the traffic required to connect and maintain the actual user traffic. It is also responsible for maintaining overall network operation (router to router communications). The data plane (voice) portion of the VoIP protocol is the actual traffic that needs to get from one end to another. VoIP offers several advantages over traditional circuit switching networks. These include:

- Low cost: Utilizes the widespread availability of Internet to make phone calls using standard, readily available computer parts.
- Possibility for greater bandwidth efficiency: Due to the vast amounts of codecs available for VoIP networks (which may also be available to circuit-switching networks but are not implemented due to cost restrictions), voice data may be transmitted at rates different from the standard 64kbps, including variable bit rates.

Voice over Internet Protocol (VoIP) systems can be built up in numerous forms and these systems include mobile units, conferencing units and telephone handsets. Along with this equipment of end users, VoIP stands for Voice over Internet Protocol. It is also referred to as IP Telephony or Internet Telephony. It is another way of making phone calls, with the difference of making the calls cheaper or completely free. The 'phone' part is not always present anymore, as you can communicate without a telephone set.

VoIP has a lot of advantages over the traditional phone system. The main reason for which people are so massively turning to VoIP technology is the cost. VoIP is said to be cheap, but most people use it for free. Yes, if you have a computer with a microphone and speakers, and a good Internet connection, you can communicate using VoIP for free. This can also be possible with your mobile and home phone.

There are many ways of using VoIP technology. It all depends on where and how you will be making the calls. It could be at home, at work, in your corporate network, during a travel and even on the beach. The way you make calls varies with the VoIP service you use.

Real-time Application — To add a special challenge for the deployment of a VoIP network, integration and detailed information about the characteristics of each, is to maintain the position of the phone. This information is often performed manually in a spreadsheet or database of some type. This information is required to keep on working hard and expensive, making the time to manage the VoIP deployment, represents a significant labor cost. Phones to connect to a network increases as the number of time and effort required to keep accurate information about the growth. As a result, the rule will no longer have to wait for new phones or change times will move to approve by an inability to control systems, and the entire process. [3]

1.2 Problem Statement

The problem is the sensitivity of VoIP quality to delay, jitter and packet loss.

VoIP is supported by several protocols and it has its own set of characteristics which are not common in other types of applications. These include the use of Transport Control Protocol (TCP), User Datagram Protocol (UDP) and, Real-time Transport Protocol (RTP).

1.3 Objectives

To maintain VoIP quality during voice transfer process, to achieve

- **Simulate the propose scheme in any network simulator.**
- **Measure the QoS parameters of transmission protocol.**
- **Compare the proposed scenarios of TCP UDP and RTP network with other.**
- **Evaluate either protocol that batter for transmission the VoIP packets.**

1.4 Research Methodology

NS-2 is the simulator tool used for designing the network and deploying VoIP technology. Attached with TCP, UDP and RTP to get result from Mat-lab for throughput, end to end delay, packet loss and Jitter.

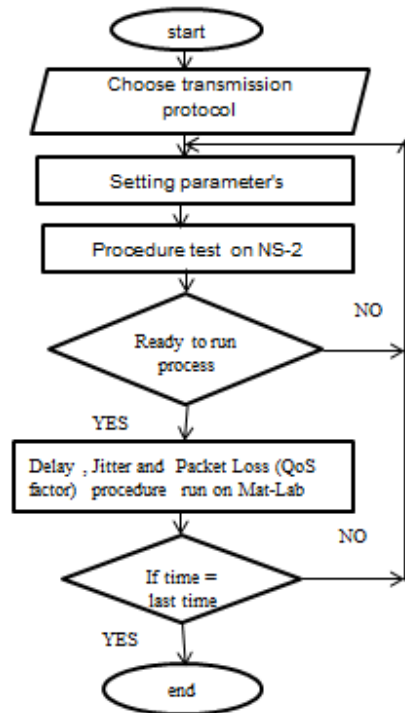


Figure (1-1): Methodology flow chart

1.5 Project Scope

There have been numerous research efforts regarding the performance of TCP and UDP in VoIP. In this project we have included and reviewed various performance studies carried out for RTP, in particular the ones those focuses the voice communication trends in today's IP network. We have also compared QoS factor when these are transported over TCP, UDP and RTP scenarios. We are also analyzed the traffic when it runs over TCP, UDP and RTP with constant level and compared the results.

In this research we well thought-out as follows:

- Chapter 1 : an introduction to research idle and background of Voice over IP.

- Chapter 2: literature review and related work for VoIP over TCP, UDP and RTP.
- Chapter 3: QoS factor and the Methodology of applied scenarios of simulation design.
- Chapter 4: Results and Discussion of comprises of TCP, UDP and RTP refer to throughput, delays, packet loss and jitter.
- Chapter 5: conclusion and remarks presented the conclusion of the research along with the contributions from this thesis.