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Implementation of Session Initiation Protocol for Better QoS in Next Generation Networks

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Abstract -**The Network model designed for of SIP-IMS provides the real time optimized QoS Parameter like jitter, , End to End delay, MOS and Packet Delay are discussed together with thesedesign models, and are simulated using the OPNET Modeler. Several results related to link utilization, end-to-end delay, packet loss and network convergence are obtained by using already available statistical methods and modules from OPNET Modele**

***Keywords*-soft switch,ngn,sip.ims,ip,QoS**

INTRODUCTION

Next Generation Network or NGN is the packet switched IP based network service that provides both using wire line and wireless communication systems and provide high grade of services based on Internet Protocol Version 6(IPV6). Next generation network is a mile stone in telecom industry and new era of technology. Before there were separate networks for Voice, data, internet service which is quiet costly for the subscriber. Next generation network is for convergence. The word "convergence" in next generation network means that Convergence of Application services, network services, network facilities and content services. Next generation network will provide all services and facility under one plate form with reasonable and economical rates. It will provide IP-telephony, IP-VoIP mobile and IP based internet services. next generation network will provide good quality of service(Quos).It will provide good capacity, High bandwidth, good coverage of service, performance and efficiency and low rate of packet loss.

II.QUALITY OF SERVICE (QOS)

In IP networking, quality can mean many things. In VoIP, quality simply means being able to listen and speak in a clear and continuous voice, without unwanted noise. QoS (Quality of Service) is a major issue in VoIP implementations. The issue is how to guarantee that packet traffic for a voice or other media connection will not be delayed or dropped due interference from other lower priority traffic. Parameter we have concentrated for QoS are **Packet delay variation, End to End delay Jitter and MOS.**

III. Session Initiation Protocol (SIP):

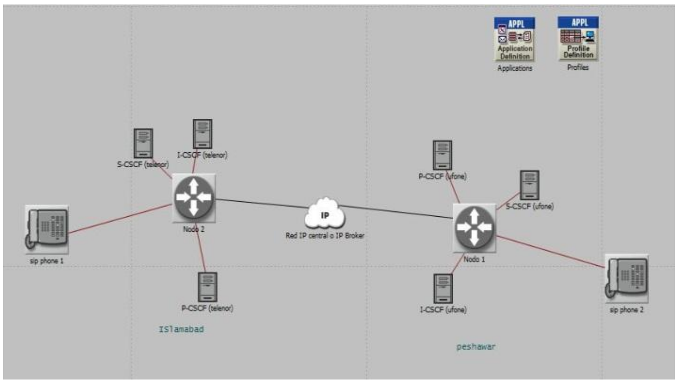


Figure 1:Design of SIP IMS Network

is a peer-peer signaling protocol for VoIP, developed by the IETF MMUSIC Working Group and defined in *RFC 2543.*

It is a proposed standard for *initiating, modifying, and terminating an interactive user session that involves multimedia elements* such as video, voice, instant messaging, online games, and virtual reality. SIP requires a simple core network with intelligence embedded in endpoints; thus it is highly scalable. It closely resembles HTTP and SMTP; thus SIP sits comfortably alongside Internet applications.

IV. SIP IMS(IP-Multimedia Subsystem)

is core architecture. Servers are running with using SIP registration protocol called CSCF. The Call Session Control Function (CSCF) is a central component to signaling and control within the IP Multimedia Subsystem (IMS) network. Subdivided into three separate parts, the CSCF is responsible for all signaling via Session Initiation Protocol (SIP) between the Transport Plane, Control Plane, and the Application Plane of IMS. The CSCF consists of the Proxy CSCF (P-CSCF), Interrogating CSCF (I-CSCF), and the Serving CSCF (S-CSCF), which each have unique functions within IMS.

The P-CSCF is responsible for interfacing directly with the Transport Plane layer and is the first point of signaling within IMS for any end-layer.

The main function of the I-CSCF is to simply proxy between the P-CSCF as entry point and S-CSCF as control point for applications found in the Applications layer.

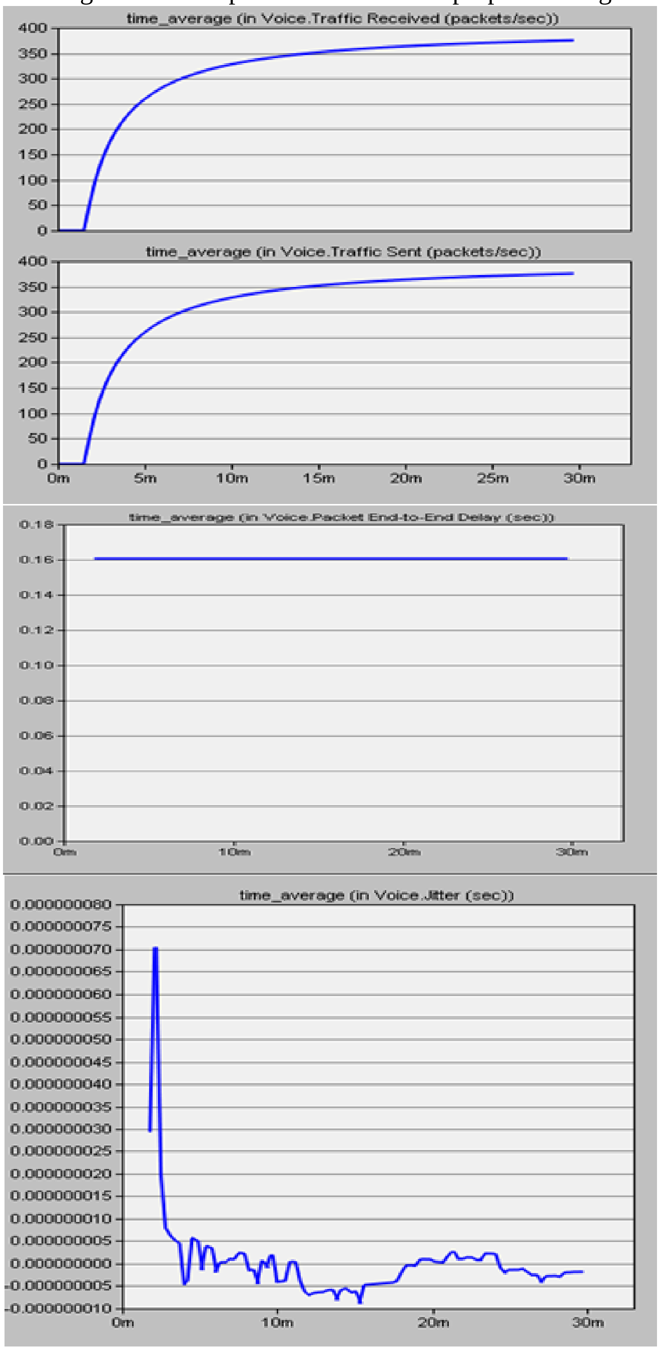
The S-CSCF is responsible for interfacing with the Application Servers (AS) in the Application layer. Upon receiving a registration request SIP message from an I-CSCF, the S-CSCF will query the HSS via Diameter protocol to register the terminal as being currently served by itself.

V. SIP-IMS NETWORK DESIGN:

The design is shown in figure 1.

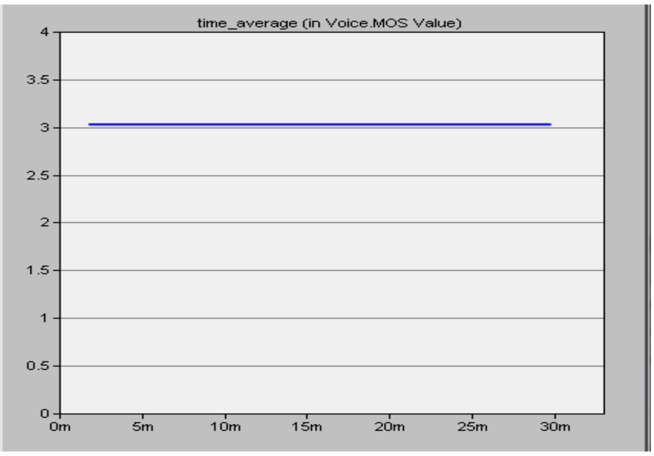
VI. Results:

Following are some important results of the proposed design.



VII. CONCLUSION:

It is concluded from the results is that SIP Network is the secure, economical and reliable network for communication. It's Network architecture is Simple as compared to other network architecture in NGN. It have good QoS for Jitter, End to End delay, MOS and packet delay variation. It will reduce the CAPEX and OPEX of any organization. It is convergent Network.



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