Abstract

The development of VoIP technology has brought business into a new era of communication that offers a unification of all communication that are routed through the multimedia and internet. Further development of the internet is the emergence of the concept known as internet telephony. This concept allows the incorporation of all IP application and services that exist in internet and telephony, so the concept is expected in the future will be widely used., combined with existing telephony infrastructure and can be predicted.

Protocol that is used for VoIP called H323 and SIP protocols. Both Protocols are issued by different standardization organization. For H323 issued by ITU-T, while for SIP issued by IETF (nternet Engineering Task Force). To run VoIP application is commonly used Asterisk-based server. Asterisk is open sources that is very important for world telecommunication. Asteris is implementation of the telephone Private Branch Exchange (PBX). In this final task will be made using the VoIP application server asterisk on LAN network to be able to measure the performance parameter (delay, *jitter*, *packet loss*, *throughput*) and analyze the signaling. That is call setup delay.

Keyword : VoIP, Asterisk, H323, SIP, Delay, Jitter, Packet Loss, Throughput, Call Setup Delay