ABSTRACT

Nowadays communication using VoIP (Voice over Internet Protocol) shows that quality to be one of the many ways of people to communicate. The technology which transforms analog voice signal to digital voice signal then perform packet grouping and transmit over IP network, become choice that compete to PSTN. VoIP offers the lower price of communication.

ITU – T recommended H.323 signaling protocol as first protocol for VoIP application. As ITU – T recommendation, H.323 is signaling protocol mainly for conference that includes voice, video and data. H.323 has widely implemented in the world but still have much problem when to be implemented at WAN (Wide Area Network). IETF knows that condition. They released SIP as choice of signaling protocol of H.323. It has slim code that have good quality in WAN. With their stability and supported by WAN, SIP look can substitute H.323 protocol signaling. The Asterisk® programmers develop new protocol signaling that offers new feature which SIP and H.323 wasn't supported. IAX can easily through packet pass the NAT (Network Address Translation). Which SIP or H.323 needs device more.

As a result from analysis give information about VoIP quality at Local Area Network STT Telkom using IAX protocol. The experiment show value of one way delay is 0-150ms. It means still in good range according to ITU-T recommendation. Jitter also have good value 0-20ms with IAX has the bigger value (18-20ms). IAX perform quality nearly with SIP. IAX can solve NAT problems. It is shows with two RTP directions in a single call. All of the experiment and MOS give result that VoIP service at LAN STT Telkom using IAX protocol is good enough.

Keyword : VoIP, H.323, SIP, IAX, NAT.