ABSTRACT

SIP protocol (session initiated protocol) is a signaling protocol that is used to exchange real-time data on internet services such as VoIP (Voice over IP). This protocol uses the identity of the device in the form of IP (internet protocol) to determine the paths that each user will use to communicate with each other. However, there are problems with the SIP protocol that includes the IP address on the payloads in the signaling packet that is sent, so it cannot reach the client on a LAN (Local Area Network) that is behind NAT (Network Address Translation) when routed on a public network. This causes real-time communication such as VoIP cannot be used.

In this final project, a design simulation will be done to overcome this. That is by using the Elastix server and the Linphone softphone that supports the STUN protocol (Session Traversal Utilities for NAT) and TRUNK information is added to the server. STUN binds the users to the public IP that they use and maps their IP address. On the other hand TRUNK serves to find information about the public IP of the neighbor server where other users are. With this, each user can communicate with each other even if they are behind NAT.

In this final project, it was found that by using Elasitx server and Linphone softphone that supports STUN and TRUNK mechanisms, VoIP communication such as voice, conference calls, and video calls can be used. For QoS (Quality of Service) with parameters of delay, jitter, packet loss and MOS values, the results are very good for voice calls and good for video calls. This can be seen from the voice call delay obtained from 10 calls which is 19.97 ms, the jitter obtained is 2.2 ms, the obtained packet loss is 0% and the MOS value is 4.4. For video call delay obtained from 5 calls, which is 19.97 ms, jitter is 3.7 ms, packet loss is 0% and MOS value is 4.1.

Keywords *: VoIP, SIP, NAT, STUN, TRUNK.*