ABSTRACT

VoIP is a technology with narrow band system which is useful to replace traditional telephony for voice services in packet switch based. The internet world has applied VoIP for most application in IPv4 networks. Since IPv6 has expanded and used by common people, there would be a transition where some people ready with the network of IPv6 while some of them still working on IPv4 networks.

There are lots of mechanisms for IPv4/IPv6 transition. By adding SIP Proxy and media gateway that can be mentioned as NAT traverse method in VoIP technologies where one side can be seen as intra network and the others can be seen as external networks with bridging mode one to another to communicate, hopefully could be an efficient method for this translation because no device needed such as a specific router only to route 6 to 4 networks and vice versa.

This project will implement integration of VoIP SIP in IPv4/IPv6 networks using MSP (Mini SIP Proxy) with dual IP on bridging mode where SER act as SIP Proxy and RTP Proxy that works as UFWDD (UDP Forwarding Daemon) for the RTP stream.

The measurements of analyzing performance show that proxy has an ability to modify header to forward message to destination. IPv6 with the longer header has no influence in transmission network. Even the measurement shows that IPv6 has a better performance compare to IPv4. Parameters for analyzing performance show that method of inserting proxy between IPv4/IPv6 networks has no influence for quality of VoIP itself. This has been proved by measuring the performance of this project.

Keywords: VoIP, SIP, IPv4, IPv6, SIP Proxy, and UFWDD.