Abstract

VoIP is a communications technology that can transmit voice data in IP network. This technology is growing and mostlyused in daily activities because it has many advantages such us can reduce the cost of communication and simplicity of the system needed. The problem that arises is the speech quality is very dependent of conditions internet network traffic. If network happen congestion, it will cause increasing the value of packet loss and delay. So that is can effect the speech quality of services provided. Therefore the communication system that is able to adapt to network traffic conditions become alternative solution that can be implemented.

VoIP quality adaptation is a mechanism of voip that can adapt basednetwork conditions to control the sound quality. One method used in this thesis is to change the codec that has a different bit rate values using the method re-invite in SIP. So that when the congestion network condistions occur then the system will perform codec downgrade to a smaller bit rate, then when the network return to normal condition, the system will upgrade codec bit rate to ensure the speech quality. Trasehold parameter are used to change the codec using E-model, which is one of method for voice quality prediction.

The result showed that voip quality adaptation has batter performance as compared to voip that not using adaptation mechanism. This is evidenced by the magnitude of the larger MOS value for all scenario tested.

Keywords : VoIP quality adaptation, codec, re-invite, SIP, E-model, MOS