ABSTRACT

VoIP is a kind of communications technology that can transmit data and voice on IP networks promises several advantages such as lower cost of conversations and simplicity of the system compared to a conventional telephone. Congestion can be affected to the sound quality. VoIP quality adaptation can be a solution to keep the sound quality on VoIP technology. The mechanism is adapted the delivery bit rates, based on the information packet loss, delay, and jitter.

VoIP quality adaptation is a voip mechanism that can adapt to network conditions in controlling the quality of the sound. One of the methods in this thesis is by conduct perform codec switch that has a different bit rate value by using the function on the SIP re-invite. So, if there is a congestion on the network conditions, the system will automatically downgrade the codec into the smaller bit rate, then when conditions return to normal , the system will upgrade performancy of the codec into more larger bit rate to ensure the quality of the sound. Trasehold parameters are used to change the codec using the Extended E-model, which can predict the value of MOS.

It will be evidenced according to the initial hypothesis, that the measurement of voice quality on VoIP in the IMS network with switches codec adaptation has better performance compared to VoIP that can not do the adaptation mechanisms.

Keywords : VoIP, VoIP quality adaptation, Extended E-Model, SIP, MOS, IMS