ABSTRACT

One of the demands on the mobile communication service is voice service that has good quality, large capacity, and a small error. Capacity depends on the availability of bandwidth. However, problems arise due to limited bandwidth. One way to overcome the above problems is using proper speech coder.

In this thesis discussed about the implementation of AMR and ADPCM voice coding on the IEEE 802.16e WiMAX network in AWGN channel and fading. It aims to identify the coding performance of the voice on IEEE 802.16e WiMAX network with measurement parameters such as Mean Square Error (MSE) and the channel capacity.

The results of this thesis indicate that ADPCM provides better performance than AMR because it has an average value of MSE is smaller. But the side channel capacity, AMR has the advantage of being able to reduce the maximum bandwidth of 88.1% in the comparison condition AMR 4.75 kbps with 40 kbps ADPCM.