

Abstract

In this time, the needs of data and voice communication have an equal priority, if the network built of them done by their each self, it must be need a big cost. Including the needs for audio conference. There are several audio conference application technology that can be use at IP network, which is data network. The kind of audio conference is the one that based on SIP (Session Initiation Protocol) and multicast.

In this final task, has done a performance comparison between audio conference technology that based on SIP and multicast. The performance is about the voice quality and bandwidth efficiency. The measured parameters for voice quality are the delay and packet loss. The experiment based on the kind of the using codec, there are G.711 A-Law, G.711 U-Law and GSM 06.10. And also based on the network traffic condition. To create this condition, the network was given a traffic load.

From the experiment results with the same codec, for the network that wasn't given a traffic load, the voice quality that produces by both application is same and including into a good level. When the network was given an increasingly traffic load, the decreasing voice quality that produce by Multicast audio conference is smaller than the SIP. Beside that, Multicast audio conference application is more efficient in the use of bandwidth compare with SIP with the efficiency about 37-39 %.

Keywords: audio conference, SIP, multicast, codec, voice quality, bandwidth.