ABSTRACT

RTP, the Real Time Transport Protocol, has gained widespread acceptance as the transport protocol for voice and vide on the internet. Its companion control protocol, the Real Time Control Protocol (RTCP), is used fop loose session control, QoS reporting such as delay, jitter, and calculation packet loss, and media synchronization. The RTP specification describes an algorithm for determining the RTCP packet transmission rate in a multicast RTP session which could used in sessions with anywhere from one to a million members.

However, have several problems with this algorithm when used with very large groups with rapidly changing group membership. One problem is the flood of RTCP packets which occurs when many users join a multicast RTP session at nearly the same time.

To solve this problem, in this Final Project I will used a timer reconsideration which have two mode, conditional reconsideration and unconditional reconsideration. Applying of timer reconsideration assumed for network without delay and loss. While for network with delay and loss will be enhanced by a calculation for network delay and the loss.

At this mathematical simulation, all user incircuit to network with bandwidth downstream assumed equal to each every user. Network assumed to own very big bandwidth upstream, so that congestion only be became of just downstream. Congestion caused since packet of RTCP Report from all user, what sent over each every user which follow to join that session

In this Final Project I present a mathematical analysis of this algoritma, and demonstrate that performs extremely well. Beside that, also be given data result of mathematical analysis program to assist to show the performansi timer reconsideration. With exploiting this timer reconsideration at real application, expected problems of congestion of effect from flood packet RTCP is solvable

Keywords : RTP, RTCP, Conditional Reconsideration, Unconditional Reconsideration