

## ABSTRACT

RTP, the Real Time Transport Protocol, has gained widespread acceptance as the transport protocol for voice and video on the internet. Its companion control protocol, the Real Time Control Protocol (RTCP), is used for 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 be used in sessions with anywhere from one to a million members.

However, there are 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 use a timer reconsideration which has two modes, 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 users are connected to network with bandwidth downstream assumed equal to each every user. Network assumed to own very big bandwidth upstream, so that congestion only be caused by just downstream. Congestion caused since packet of RTCP Report from all users, what sent over each every user which follow to join that session

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

Keywords : RTP, RTCP, Conditional Reconsideration, Unconditional Reconsideration