Abstract

In recent years, Session Initiation Protocol (SIP) developed by the Internet Engineering Task Force (IETF) has gained significant popularity in the Voiceover- IP (VoIP) arena and is competing with the Internet Multimedia protocol H.323. SIP is also selected by Third Generation Partnership Project (3GPP) as a standard signaling protocol for service control in Third Generation (3G) wireless network.

SIP is a communication control protocol capable of running on different transport layers, e.g., Transport Control Protocol (TCP), User Datagram Protocol (UDP), or Stream Control Transmission Protocol (SCTP). Today's SIP application is mostly operating over the unreliable transport protocol UDP. In lossy environment such as wireless networks and congested Internet networks, SIP messages can be lost or delivered out of sequence. The SIP application then has to retransmit the lost messages and re-order the received packets. This additional processing overhead can degrade the performance of the SIP application. To solve this problem, researchers are looking for a more suitable transport layer for SIP.

SCTP, a transport protocol providing acknowledged, error-free, non-duplicated transfer of messages, has been proposed to be an alternative to UDP and TCP. The multi-streaming and multi-homing features of SCTP are especially attractive for applications that have stringent performance and high reliability requirements.

In this final measurements to determine the results obtained performance comparison of UDP and SCTP transport protocol is measured from the delay which both showed results that are not much different, still in accordance with the limits set by the ITU at around 150-400 ms. Jitter measurements show they have a performance that exceeds the standards of CISCO which is less than 30 ms, caused the jitter buffer at the time of testing. And while the current is measured by the packet loss parameter indicates that the SCTP is superior in terms of dealing with packet loss of about 10-20%.

Kata kunci: VoIP, SIP,UDP, SCTP, multistreaming, *delay,jitter,packet loss*.