Abstract

The Session Initiation Protocol is an application layer control signaling protocol for creating, modifying and terminating sessions, these include internet multimedia conferences, internet telephone calls and multimedia distribution. SIP is text-based protocol and can be developed with additional feature and service, these include call control service, presence, instant messages, mobility, and interoperability with other telephony system. This technology can fulfilled the needed of call connection to other area in internet coverage. The Real-time Transport Protocol provides end-to-end network transport functions suitable for applications transmitting real-time data such as audio, video or simulation data, over multicast or unicast network services.

By using Session Initiation Protocol for control signaling protocol and RTP for controlling media transport on a system that will be implemented. Internet Telephony technology with SIP and RTP will be integrated with administration system will be more effective in the building and maintenance of the system. Analize processing to SIP and RTP with using of type of service and codecs that used by client in order to connecting to the system so it can be obtain SIP message behavior and call flow and RTP delay, jitter, and bandwidth compsumptions.

Keywords: Internet Telephony, SIP, RTP, linux.