ABSTRACT

There are currently two main standards for VoIP signaling control protocols; H.323 and SIP. Both of them provide the same basic signals, call control services, and supplementary services. A document from "Nortel Network" stated that H.323 was designed with ATM and ISDN signaling in mind, H323 does not suite for signaling control for VoIP, because H.323 is inherently complex and thus inefficient for VoIP. SIP is text based therefore it simplifies the implementation task. The SIP is more flexible, it can be modified easier while still maintain the original characteristics. The implementation of SIP is still in the beginning state. There are only few implementations of SIP. This work studies the SIP in the way to extend its functions to make it be able to replace PBX. Although the SIP has been developed for a period of time, its current functionalities do not cover all of the services that can be provided by current PBX system. The services are Call Pickup and Call Reservation services. To make the SIP capable to replace the PBX, it should be able to provide the services. This thesis designs the extended SIP and implements the extended SIP server, for the PBX functionality. The implemented PBX server can provide most of the important functions similar to the traditional PBX machine.