



## VOIP Experience

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### SOFT SWITCH

Softswitch is generic name for new approach to IP telephony switching; it allows enjoying the economic advantages of offering multiple services over IP network and has potential to address the shortcomings of traditional switches. The Softswitch works with IP gateways to perform call control functions such as call setup and tear-down. Softswitch interconnects existing PSTN via Signaling gateways and media gateways and also H.323 networks and SIP networks. This was developed on VxWorks on the X86 platform.

### IMPLEMENTATION OF A VOIP TEST BED

This project aims at developing a VOIP test bed. ITU-T H.323 is a standard for sending voice (audio) and video using IP on the public Internet and within an intranet. In addition to IP, VoIP uses real-time protocol (RTP) to help ensure that packets get delivered in a timely way. The project includes developing RTP, RTCP and H.245 signaling protocols. This test bed contains one gatekeeper, four H.323 servers established communication between the servers using gatekeeper. The development was done on Linux on X86 platform.

### DEVELOPMENT OF MPLS PROTOCOL STACK

In this project we developed the Prototype of MPLS protocol architecture for traffic engineering with Label Distribution Protocol (LDP) as the underlying signaling protocol. The MPLS router will analyze the packet at the edge of the network and then assign a label and in subsequent routers only label swapping will be performed. In this project Bandwidth is considered as the constraint for Constraint based Routing. LDP consist of four sets of messages for establishing the Label Switched Path. Discovery messages maintain the presence of the MPLS router in the network. Session messages are used to establish and maintain the session between peers. Advertisement messages are used to establish the Label Switched Path by creating the Label Information Base. The notification messages are used to send the error notification and advisory notification between the routers. This was developed on VxWorks.

### DEVELOPMENT OF A SIP ENABLED VOIP VPN

This project, done for Nortel, involved the development of Call processing, provisioning and Services support for Nortel's Call server (CS2K) for its SIP based VOIP VPN solution. It involved Design, Coding and testing of CallP component in Nortel's Call server. The technologies used on this project include Protel (Nortel's Proprietary Language), PRI CALLP, SIP and SOS.



## **SPECTRUM MG4K APPLICATION CALL PROCESSING**

This project, done for Nortel, involved the sustenance of its Application Call Processing (PRI, ISUP) Software of Spectrum Peripheral Module for its DMS family of Switches and MG4K, Nortel's media gateway. The main task involved was to Design, Code and test fix for issues raised in Nortel's customer sites. The technologies used here were Protel, PRI, ISUP CALLP, C++, UNIX, Clearcase and SOS.

## **FEATURE ENHANCEMENT OF AVAYA IP PHONES**

This project involves designing and implementing feature enhancements and maintains existing AVAYA IP telephones. The AVAYA IP phone is VOIP based phone which uses H323/SIP stacks with basic phone operations plus extra features like web browser, http download etc. Some of the tasks involved include providing multilingual support, Make use of WindML tool to support Graphics in existing phones, Migrating web APIs' from different versions of releases, and sustenance of RAS. This development was done on VxWorks using C++.

## **SIP STACK DEVELOPMENT**

SIP Stack is an Application Layer Protocol Stack, which helps in initiation, modification and Termination of IP session. The Project involves development of a flexible SIP stack User Agent, as per RFC 3261. Our SIP Stack interoperated with more than 25 SIP stack in SIPIT forum held in Sydney. This implementation was done on Linux using C, and was made OS independent, so the code could be compiled for most OS.

## **DEVELOPMENT OF A SIP NETWORK SERVER**

This project involves the development of a SIP Network Server, which is a carrier-grade, high performance, feature rich offering that serves as a core infrastructure router by providing the transaction-stateful proxy, registrar and location server functionality. The various software tools used include g++, Val grind, purify, pure-cov, on Mont vista Linux Platform.

## **DEVELOPMENT OF A SIP B2BUA FRAMEWORK**

Session Initiation Protocol is an industry standard signaling protocol for Voice over IP technology. We built a B2BUA server framework on top of a SIP Stack with the aim to provide OSA interface for further application development on the SIP server framework. This project providing centralized Back to Back User agent B2BUA server with call picking ,call parking and click to dial services. The software tools used include g++, purify, pure-cov, on Solaris Platform.



## **DEVELOPMENT OF A STATIC LOAD BALANCER**

Load Balancer distributes the incoming SIP calls to backend servers. It is a critical component in networks that meet call-handling requirements by distributing calls to the multiple proxies. It ensures that the backend server is highly available. The software tools used are g++, Val grind, purify, pure-cov, make utility and python on Linux Platform.

## **DEVELOPMENT OF IMS SERVER**

This project involved developing a Unified IP-based architecture enabling converged network for fixed, wireless and cable access using VoIP. Implementation of this was based on a 3GPP standardized implementation of SIP. We also developed the P-CSCF and S-CSCF over the SIP server framework. The software tools used were g++, Val grind, purify, pure-cov, make utility and python on Linux Platform.