Implementation of VoIP in University of Kelaniya

T. Pathirana[1*], I. Solangaarachchi[1], D. Weerasinghe[1]

In many enterprise level environment, it was a habit to inherit Private Branch Exchanges or PBX to reduce telephone costs by not using public switched telephone networks (PSTN) for internal voice calls. Earlier days traditional POTS (Plain Old Telephone System) PBX severed this purpose. In modern technical era, calling through Internet or Internet Protocol (IP) Telephony or Voice-over IP (VoIP) is a technology that allows delivery of audio and other multimedia content over the Internet. With the advances of Internet and VoIP systems, many enterprises are migrating from POTS PBX to VoIP PBX as those systems promises high quality voice transferring plus many added services like video conferencing, MMS, Chat, easy billing, monitoring, etc. With new additions of multi-story buildings and with the high data speed network implementations it was a mere truth that incorporating VoIP will further reduce costs in implementing and maintaining voice call systems.

This paper describes the implementation of a VoIP based communication solution designed for University of Kelaniya. It is quite evident that larger portion in university budget is accounted as the PBX maintaining charges. But with introduction of VoIP, we can use the existing Local Area Network and the single network connection terminating at the user premises for connecting both telephone handset and the computer. Therefore, avoids the cost for cabling of new connections and maintenance. Additionally, the quality of the calls would be significantly elevated.

As the main component of this proposed system, an free and open source Asterisk based SIP system was compiled on an existing Ubuntu server, here SIP or Session Initiation Protocol refers to a signaling protocol designed to create, modify and terminate a multimedia session over the internet protocol. Basic requirement to initiate SIP PBX is to identify a suitable numbering plan which can be used with future provisioning, therefore a four-digit number system was utilized for the task. Also the customization of Asterisk according to University needs, implementation of Interactive Voice Responses and user metering was done. Then the system was connected to the traditional PBX through an E1 trunk so that it will enable users to call within two systems. Softphones were used between IT staff, as they allowed roaming capability if used in a wireless network. Hard wired phones were also used for testing. After the implementation the only costs accounted were, for the new hard phones and E1 trunk.

In its implementation we had to focus on performance, quality of service, reliability and availability, scalability, obsolescence and service life, security and regulatory issues, electricity and backup power, network traffic and bandwidth.

When looking at the financial benefits to university, maintaining traditional PBX cost more than using this stated system as it totally depends on the existing LAN. Because of the maintenance is done by the university IT staff as their daily duty, no additional costs will apply for installing, commissioning and maintaining of VoIP system. Even though the initial costs will be high, it is an added advantage that VoIP users are able to call freely within the National Research and Education Network if they are connected in such a method. But all outgoing calls to the PSTN will still be billed as we are legally bound not to route voice calls through commodity Internet.

Keywords: *Voice over IP, SIP server, Implementation of VoIP, High Definition Voice Calls, Asterisk*

[1] University of Kelaniya, Sri Lanka *thilina@kln.ac.lk