

VoIP Test-bed

The Networks Research Lab in its quest for state of the art research and development begun to build a small VoIP pilot to examine signaling and Quality of Service issues. The pilot started six months ago after a study on VoIP signaling and especially Session Initiation Protocol (SIP). In its quest the lab team selected to examine several Open Source SIP Proxy servers and select one to be used for the pilot. This led to the selection of Asterisk Open Source PBX.

The first step was to setup an internal network that was able to manage calls from one phone to the other as in any voice infrastructure. After this was completed the team connected Asterisk with the University's main PBX by using a line card that supported FXO channels. The University provided us with two analog ports that we utilized to make calls from our VoIP network to other University extensions and the outside world. The dial plan was configured in such way that the users of the IP devices could dial as they would normally dial from the any existing university telephone. For example if a user wants to reach extension 2687 he just dials 2687. The problem that still remains is that since we are not natively connected to the University of Cyprus PBX, but rather two analog ports are used, we are not able to display caller id from the system. In order to check whether the system could provide caller id we connected the system to a standard telephone line that supported caller id and the system responded by displaying the number. If the number was stored in the global address book then the name of the number holder was displayed.

The second step of the pilot was to give access to the system to outside IP phones. The firewall was configured to allow SIP signaling to pass towards the asterisk system and an IP phone was plugged in the Computer Science Department LAN. This was successful so the team decided to configure PBX systems to remote locations in order to examine quality of service issues in low bandwidth connections. The first remote location selected was a researcher's house. An asterisk system was setup at his location and an IP phone was connected to it. A soft phone was installed on his PC and on his PDA, turning his PDA to a wireless phone. After that the two asterisk boxes were interconnected using the internet and his ADSL connection. The voice quality at first was not adequate and some QoS parameters were added to his ADSL router to give higher priority to Voice calls. After that the two dial plans were merged in order to allow calls from one location to the other. Now the researcher can reach any University extension just by dialing the extension he needs and talk with anyone at acceptable voice qualities, at zero cost if we exclude his ADSL connection he was already paying for. Due to the limitations of the ADSL network another location was selected to build a similar network in order to evaluate the ability to sustain voice calls from an ADSL connection to another ADSL connection. This was completed and successful calls were made but the uplink speed of the ADSL network limited that only up to three simultaneous calls could be achieved.

To evaluate how the lab can save money during travel the firewall was setup to allow VPN connections to the internal lab, thus giving any traveling user the ability to connect to Asterisk via a soft phone and be reachable from within the lab. This gave the ability to call anyone he wanted, as long as he was connected to the internet. This was put to the test during a meeting in Brussels where the hotel offered free WLAN access to its residents. A researcher was able to communicate with his co workers via a VPN connection and his PDA. This saved a respectable amount of money, which he would have to pay to talk with his co workers.

The next step in the pilot system evaluation was to take advantage of the free VoIP providers that are easily found on the internet. Free World Dial up (FWD) was the first provider we selected. After we signed up for an account the asterisk system was configured to connect to FWD network and allow the VoIP users to dial to other FWD users and to its peering providers. This allowed the system to dial for free to toll free numbers in the UK, USA, Germany, Norway, Netherlands by utilizing IP telephony over the internet. A test was made by one of the researchers when he used his IP phone to dial to the Conferencing Bridge in order to participate to telephone meeting for an EU project. He reported that the quality was as good as the quality experience he had when he was using the university's telephone system. Other providers were later added like VoIPBuster with excellent results as well as voiptalk.org and interconnection with the local Skype account.

After the successful implementation and testing of the pilot some new applications were selected in order to enhance productivity. These applications were unified messaging, fax to email and visa versa, call auditing and reporting and CRM integration. At this point unified messaging, fax to email and call auditing and reporting are completed giving the users the ability to receive their faxes to their inbox and their voicemails as attached audio files and the system administrators the ability to take call statistics for cost analysis and least cost routing decisions. Also an extra feature was added that gives the ability of the users to check their voice mails from a web interface from anywhere in the world and forward a message to a colleague.

The following figure shows the current VoIP network setup.