NGI/CE/03

Voice over IP

FY 2004 Proposal to the NOAA HPCC Program

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Voice over IP

Proposal for FY 2004 HPCC Funding

Prepared by: Alex Hsia

Executive Summary:

Data networks are quickly becoming one of the key components to support NOAA's mission. We can leverage the required performance and reliability of the data networks by deploying Voice over IP (VoIP) for certain applications. This will allow NOAA to benefit from the vast features that VoIP provides such as toll bypass, cost reduction, flexible dialplan, rich feature base, rapid deployment and reconfiguration to name a few.

Problem Statement:

For certain applications such as a small remote branch office, or to support a field mission, it is cost prohibitive to deploy a typical PBX to support the telephone needs of the personnel. NOAA can either rely on obtaining phone service from a local host organization, or obtain dedicated lines from the local telco. Neither of these solutions provide a convenient, cost effective way for NOAA employees to make calls to other NOAA offices including the home office. Furthermore, features such as voicemail, voice menus, conferencing and integrated voice response are costly to implement in these traditional analog or digital phone solutions.

The cost savings are obvious when a flexible dial plan can be implemented where local calls can use the Public Switched Telephone Network (PSTN) and calls to other NOAA offices can be routed over Internet trunks to avoid costly long distance charges.

Proposed Solution:

The proposal will evaluate the utility of a Open Source PBX called Asterisk which leverages the low cost of Linux servers coupled with low cost Linux telephony hardware and rich extensible software features to provide a platform to facilitate the interoperation of the PSTN and VoIP networks. A variety of telephones will be evaluated including analog phonesets connected via VoIP telephone adapters, full featured VoIP telephones with displays capable of displaying dynamic content, and software phones on desktop computers.

Asterisk acts as a full featured PBX supporting virtually all conventional call features on station interfaces such as Caller*ID, Call Waiting, Caller*ID on Call Waiting, Call Forward/Busy, Call Forward/No Answer, Call Forward Variable, Three-way Calling, Supervised Transfer, Unsupervised Transfer, Voicemail, Meet-me Conferencing, Least Cost Routing, VoIP Gateway, Call Detail Records, etc.

Asterisk supports a variety of hardware interfaces for bringing telephony into a Linux box ranging from a single phone line to a quad T1 PCI board. These interfaces can be used to connect analog phones as well as standard analog phone lines into the Asterisk server and provide the gateway between the VoIP and PSTN networks.

In a typical remote mission configuration the Asterisk server can be configured to automatically route calls. The flexible dialplan can be configured to route local calls over to the PSTN while sending calls destined for the home office over the VoIP network to avoid long distance charges.

Security is provided ranging from static defininition of clients to the use of RSA encrypted passwords for roaming clients. Dialplans can also be segmented allowing for certain extensions to only be accessible for certain interfaces and/or VoIP users. Thus local phones can be permitted to make outgoing calls, but inbound callers would be prevented from accessing the "9" extension for an outbound line.

Analysis:

Unlike many modern "soft switches", Asterisk can use both traditional TDM technology and packet voice protocols. Calls switched on TDM interfaces provide lag-less TDM call quality, while retaining interoperability with VoIP packetized protocols.

Asterisk supports both H.323, the ITU standard for VoIP, as well as SIP, the IETF standard for VoIP. Asterisk also supports the defacto industry standard, IAX for Asterisk networking which allows transparent interoperation with NAT and PAT firewalls, including placing, receiving, and transferring calls and registration. In this way, PBX's and phones can be totally portable. Just plug them in anywhere on the Internet and they'll register with their home PBX, instantly routing extensions to them appropriately.

VoIP phones will be tested at the NOAA-Boulder remote Table Mountain Test Facility to evaluate call quality and performance over WAN links. Typical configurations to support a remote field site as well as remote field missions will be documented.

Performance Measures:

The project can be successfully accomplished according to the following timetable:

Milestones

Month 02 - Purchase Linux Telephony hardware and VoIP phones
Month 04 - Configure Asterisk software/hardware
Month 06 - Test out interoperability between VoIP and PSTN networks
Month 08 - Deploy VoIP phones at remote location for testing
Month 10 - Configure/test diaplan configurations

Month 12 - Document installation and a few example configurations

Deliverables

Provide a list of the final products from this project

- Asterisk system to bridge the PSTN and VoIP networks
- VoIP capability to remote NOAA offices
- Cost effective telephone support for remote NOAA locations