NGI/CE/03 (do not delete this) **Voice over IP**

HPCC FY 2004 Final Report

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Proposal Theme: **NGI**

Funding Summary: FY 2004 $ 31,000

Voice over IP

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Performance:

The main deliverables of multiple Asterisk <http://www.asterisk.org> VoIP servers were provided to bridge the PSTN and VoIP networks and provide VoIP capability to a test remote NOAA office out at the Table Mountain test facility in Longmont, Colorado. Testing included the evaluation of the following:

- Multiple hardware and software phones based on Session Initiation Protocol (SIP)
- Voicemail
- Unified messaging
- Interactive Voice Response (IVR) system
- Conferencing system
- Call routing
- Different Compression/Deccompression (CODECs) standards
- Network Address Translation (NAT) and Firewall traversals
- Interoperability with the Public Switched Telephone Network (PSTN)

While all of the above was tested, full interoperability with the existing PBX was not accomplished due to coordination issues with the NIST telecommunications group, the managers of the NOAA-Boulder PBX system. As a result, outbound calls from the Asterisk server to the PSTN were available, but inbound calls from the PSTN to the Asterisk server were limited to a single PSTN number. When full bidirectional capability is established more extensive testing and VoIP telephones will be deployed.

The documentation for the test setup, which includes the various configuration files, is included at the following URL <http://boulder.noaa.gov/noc/hpcc/2004/voip>.

Questions regarding this project can be directed to the principal investigator, Alex Hsia <Alex.Hsia@noaa.gov>.

Project 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 including toll bypass, cost reduction, flexible dialplan, rich feature base, rapid deployment and reconfiguration to name a few.

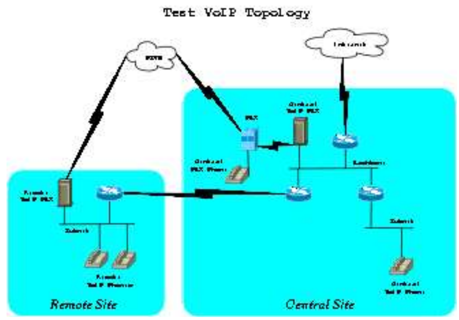
The project started with the purchase of an inexpensive telephony PCI card which provided one FXO port to connect to an analog phone line, and one FXS port to connect up an analog phone to a host which then became the test Asterisk server. From this test box configurations were explored and various VoIP phones ranging from dedicated hardware based phones from Cisco and Polycomm, to Analog Telephone Adapters (ATA) from Budgetone and Sipura and software based phones from SJ Phones and Xten were also tested.

Further testing included voicemail functionality which has the capability of sending the message as a WAV attachment to an e-mail address, and provides the capability of accessing the voicemail over the web.

Other features that were evaluated included an IVR system which included a directory application, and the capability to modify the menu system based on time of day. The Asterisk PBX can also function as a conferencing system provided there is enough capacity to support the teleconference. For those dialing in via the PSTN there would need to be adequate PSTN interfaces and for those dialing in over IP adequate bandwidth to support the number of callers would be required.

Call routing is one of the more interesting and flexible portions of the Asterisk server. You can establish least call routing where you could direct international calls to whichever VoIP provider has the lowest rates. The Asterisk server can also direct calls over different channels if primary paths are unavailable. This feature is important in cases where the Internet connection is down so the Asterisk server can direct calls over FXO interfaces to the PSTN. Call routing can also be source dependent. For instance, calls from the director to the IT support number can be routed directly to the support group manager.

After adequate testing was accomplished additional telephony hardware was purchased. This included a T1 PCI interface card and a rack mount Dell PowerEdge 1750 which became the main site VoIP PBX with a T1 connection to the existing Department of Commerce Boulder Labs PBX. A three port FXO interface card was also purchased which was installed in a host to act as the remote site VoIP PBX. This configuration was deployed as indicated by the following figure.



Call routing was configured such that VoIP calls from the remote site used the Internet if it was available to reach the main site VoIP phones and for general PSTN access, and resorted to the FXO interfaces to traverse the PSTN if the WAN was down. A similar configuration was used on the main site VoIP PBX except backup calls are directed to the PSTN number on the remote site VoIP PBX where calls are then directed by the IVR system.

One of the lessons learned is that even though network links may not be congested, Quality of Service (QoS) is still an important network feature to implement. Without QoS, the jitter on the voice conversations could get adversely high which reduces call quality. Additionally the installation of an Asterisk server has recently been greatly simplified through the Asterisk@Home <http://asteriskathome.sourceforge.net> CentOS based CD distribution.

An additional lesson learned was not to underestimate the amount of coordination required with the telecommunications people. Unfortunately the telecommunications people in Boulder are in a completely separate organization so getting cooperation in a timely manner was problematic.

The Asterisk software is also not the most stable software platform and requires occasional reboots to restore functionality so I would not recommend it for a large scale installation.

Expenditure Summary:

Category	Detailed Description	Amount	Matching
Personnel Compensation			
	NOAA-Boulder NOC	$ 25,000	$ 10,000
Capital Expenses			
	Asterisk Servers	$ 1,500	$ 2,000
	Linux telephony hardware	$ 2,500	
	VoIP phones	$ 2,000	
Total Requested:		$ 31,000	$ 12,000

Future Direction:

As indicated earlier, when full bidirectional capability is established between the VoIP and traditional PBX more extensive testing and VoIP telephones will be deployed.

The principal investigator also plans on investigating what would be required to participate in the SIP.edu <http://www.internet2.edu/sip.edu/>. This would allow people using SIP phones to make calls with familiar URLs such as <sip:Alex.Hsia@noaa.gov> rather than having to remember or look up a phone number.

Others in the David Skaggs Research Center (DSRC) have expressed interest in forwarding their desk phone to a wireless SIP phone so they could have roaming access throughout the building to overcome the bad cell phone coverage that is present in the DSRC.

SIP is not inherently a voice application. It's use includes video capability such that if two endpoints have matching capabilities they could have videoconferencing over IP.