

Some organizations consider taking the plunge off of big iron PBX platforms into IP telephony as being pretty daring, but that's nothing compared to what Sam Houston State University (SHSU) is doing. The south Texas school is boldly moving thousands of users off a Cisco VoIP platform to an open-source VoIP network based on Asterisk.

SHSU is in the process of moving its 6,000 students, faculty and staff off of Cisco CallManager IP PBXs and a legacy Nortel Meridian PBX over to Linux servers running Asterisk, which includes call processing, voicemail and PSTN gateway functionality. The driver for this project was cost, says Aaron Daniel, senior voice analyst at Sam Houston State University.

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"We thought that it will be more cost effective in the long run to go with an open source solution, because of the massive amounts of licensing fees required to keep the Cisco Call Manager network up and running," says Daniel, who this week gave a presentation on his migration project at the VON show in Boston. In the Cisco model, each phone attached to the Call Manager required a separate licensing fee to operate, Daniel says. In SHSU's Asterisk/Cisco model, where it will keep its existing Cisco phones but attach them to Asterisk servers on the back end, the phone licensing costs are eliminated.

SHSU so far has moved 1,600 IP phones from Cisco Call Managers to Asterisk, which runs the IETF-standard version of SIP. The Asterisk functions are spread across six redundant Dell servers: two act as redundant PSTN gateways (and are outfitted with four-port T-1 cards from Digium, which commercially distributes Asterisk); two more servers handle call processing; another set provides voicemail.

The Cisco 7940 and 7960 IP phones the school had deployed were updated with a standard SIP software image replacing the proprietary Cisco Skinny Call Control Protocol (SCCP, or "Skinny"), which was used to connect the phones to the Call Managers. When the IP phones were upgraded with the SIP image about a month ago "all we had to do was reboot the phones," in order to register them with the Asterisk server, he says.

More control over the IP PBX software and servers was another reason SHSU made the Asterisk jump, Daniel says. "We felt we were more susceptible to hacks," since only Cisco-approved servers updates and patches could be installed on the Windows Server 2000-based CallManagers, he says. "We have a lot more peace of mind with the open-source system. If a bad exploit is found in SIP, we can fix it ourselves."

Besides the phones, Cisco gear still comprises a large chunk of the IP telephony infrastructure at SHSU. The entire WAN and LAN is based on Cisco routers and switches. The Catalyst switches already installed support power over Ethernet (for powering IP phones) as well as QoS for voice traffic. All voice traffic on the campus network runs separate from data traffic in its own VLAN segment. Additionally, Cisco VG228 gateway devices, which can connect up to 24 copper/analog phones to an VoIP network, is used in dormitories and other areas where just a basic phone is needed instead of a more costly IP handset, Daniel says.

So far, SHSU has been able to operate the Asterisk/Cisco IP phones at one-third the cost of Call Manager/Cisco IP phones, Daniel says. When the digital Nortel handsets are migrated to SIP-based Cisco phones, or analog sets, another large chunk of savings will come just by shutting down the electrical and cooling resources required to keep the old PBX running. "The Meridian takes up an awful lot of power itself. The room it's in has to be cooled to 60 degrees, and it has to have its own generator," Daniel says.

While Asterisk and the SIP protocol lack some of the more extensive features on the Cisco Call Manager, the university community has handled the transition with few glitches. The only major feature missing in the Asterisk/Cisco phone network is secretarial functions, which allow an administrator to manage and answer phone extensions for multiple end-users. To fix this, Daniel is looking into extensions to the SIP protocol that allow for multiple-line handling, he says.

In another potential issue with open-source VoIP, SHSU loses the technical support from Cisco with its Asterisk migration. But Daniel says he has so far been able to keep up with support issues through mailing lists and the online community that develops and supports Asterisk. Dell provides support on the server hardware, and Digium supports the T-1 cards installed in the boxes.

"We try to have checks and balances," among the IT staff that supports the Asterisk system, Daniel says. "We try to keep the [the Linux and Asterisk server images] as pristine as possible." Daniel has also created copious documentation on all the Asterisk configurations and changes he's made to the software. "Basically if someone were to have to come in and take over my job, they'd have a pretty quick turnaround on learning what needs to be done," he says.

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