EDUCAUSE Center for Applied Research

Research Bulletin

Volume 2007, Issue 7

March 27, 2007

Open Source IP Telephony:

A Strategic Choice

Jose Joaquin Valdes, Jr., Colorado State University

4772 Walnut Street, Suite 206 ◆ Boulder, Colorado 80301-2538 ◆ www.educause.edu/ecar/

Overview

According to Paul Mockapetris (2006), there are approximately one billion hardwired telephone lines in the world. Most of these lines are connected to legacy or traditional time division multiplexed (TDM) private branch exchanges (PBXs). During the past few years, however, voice over Internet protocol (VoIP) and VoIP-enabled PBXs have entered the market. Although based on different technologies, these VoIP PBXs are still proprietary, and this limits the choice of features and telephone instruments to those offered by the PBX manufacturer.

Since customers are restricted to the functionality designed into the PBX by the manufacturer, they are unable to select the best-in-class telephone instruments or add services tailored to their institutional requirements. This is the traditional paradigm for PBXs, and, in a technology culture that is becoming more accustomed to having choices, this paradigm is outdated. Imagine having to purchase music CDs that are limited to a manufacturer's unique player hardware. In such a world, the hardware manufacturer would determine both the price for those CDs and the music you can purchase. In this model, customers either have their choices limited to those offered by the manufacturer, or they must purchase multiple players to expand those choices.

Open source IP telephony offers an alternative to the proprietary PBX paradigm that might enhance the strategic agility of your college or university. This research bulletin discusses the evolving open source IP telephony environment. It focuses on rationale, risks, and rewards associated with open source IP telephony and implications for higher education. Further, it's intended to challenge the reader to contemplate and question the traditional telephony paradigm.

Highlights of Open Source IP Telephony

Open source IP telephony distinguishes itself in the telephony world by adhering to the open source software paradigm that is familiar in the information technology (IT) environment. In its purest form, open source is predicated on having access to the source code. This access allows an institution to modify an application and then contribute the modification to the open source community for reuse. In practice, the community collaborates in the development of applications and shares in and benefits from the results. Further, the open source model allows an institution to use commodity or off-the-shelf hardware and nonproprietary standards, which can lower development and implementation costs. These distinctions, applied to telephony, provide higher education with the means to

- directly develop and/or extend telephony services,
- select best-in-class and off-the-shelf telephone instruments and servers,
- maintain control over service creation and deployment, and
- control operational and capital costs.

Use of open source for telephony fundamentally transitions control from the manufacturer to the user institution.

Standards-based open source IP telephony enables for voice services the flexibility that has historically been associated with standards-based data networking for computers. In networking, the Ethernet standard allows compliant network cards from various manufacturers to be used in personal computers from different manufacturers to provide network connectivity. Customers can select the network card best suited for their needs.

Approaches to Open Source IP Telephony

The spectrum of evolving open source IP telephony models currently ranges from the unsupported, downloadable software PBX to the turnkey, commercial-grade, ready-for-service, supported PBX. Colleges and universities can select the model that best meets their institutional requirements, or they can cost-effectively experiment with a downloadable version prior to making a purchasing commitment. This flexibility is one of the strengths of open source IP telephony.

There are important common elements among the predominant open source IP telephony models. Session Initiation Protocol (SIP) is the predominant signaling protocol. Basic telephone service, voicemail, and an array of traditional telephone features and telephone instruments are universally supported. Further, open source IP PBXs predominantly use Linux as the operating system. The standardization on SIP and Linux is logical within the context of open source, since both the SIP protocol and the Linux operating system are open standards.

The most obvious differences among open source IP telephony models relate to the number of protocols that are supported. One major open source IP PBX provider posits that its product is native SIP, while its competition has elected to support multiple protocols, including SIP and legacy, and additionally has created its own proprietary protocol. The fact that a manufacturer has elected to develop a proprietary protocol is interesting in light of the focus on nonproprietary solutions associated with open source.

Strategic Architecture Options

Those responsible for providing IT/telecommunication services in colleges and universities must consider the open source IP telephony models carefully, since the strategy selected will have an impact on the migration process from legacy telephony to the open source IP telephony solution.

The multi-protocol model may inherently offer a less radical migration path because it supports legacy protocol interfaces. This architectural option presumes to provide a bridge between legacy and open source by allowing institutions to retain existing telephone instruments and network connections to the public switch telephone network (PSTN). This distinction may enable a gradual, incremental migration from the traditional telephony environment to open source IP telephony.

The SIP PBX or proxy model presumably requires a replacement of the legacy environment in a more dramatic fashion. Because it supports only SIP, it renders legacy telephone instruments and network connections useless. Gateways may be required to interface to the PSTN and to other legacy PBXs or peripherals, such as stand-alone voicemail systems.

These distinctions will affect what the telephony environment looks like after transition. The multi-protocol model may be a hybrid environment of legacy and IP, while the SIP-only model is simply SIP. However, the SIP-only choice may ultimately be more in line with the "end point" IP telephony environment that is envisioned for the institution.

The relatively low cost of implementing open source IP telephony, at least in a prototype environment, allows institutions to cost-effectively conduct hands-on investigations of both evolving architectural models under various configurations. Recognizing that much of the development in this space has occurred during the past few years, it is reasonable to expect rather rapid evolution of the open source architectural models in the near future. In fact, Andrew Garcia's December 2006 piece in *eWEEK* references the explosion of choices in the VoIP market: "As with all things, eWEEK Labs expects to see corresponding retraction in the market, likely towards the end of 2007" (Garcia, 2006, para. 2). Further, *eWEEK* states that it expects to see "casualties" in the open source IP telephony space, where only a few companies have distinguished themselves sufficiently by developing viable business models and management of open source IP telephony distributions.

Reward and Risks: Colorado State University

Colorado State University (CSU) has elected to invest in an open source IP telephony strategy because it supports the institution's overarching strategy of convergence. We predict that telephony services will be one of numerous services provisioned over a converged wired and wireless/cellular network. With open source IP telephony, CSU expects a reduction in costs to provision and maintain telephone services at a time when revenue from basic telephone service are flat and long-distance usage is rapidly eroding. CSU appreciates the ability to modify and create services as a result of access to the open source code. Although not risk-free, open source IP telephony provides service and cost-containment possibilities that are not feasible with proprietary telephony systems.

CSU has used open source IP PBX code as the basis for the current voicemail system in its residence halls. We intentionally focused on the voicemail application after determining that the downloadable open source IP PBX we elected to test supported basic telephony services. The basic voicemail service on the open source IP PBX was sufficiently robust to suggest that a customized solution was viable. A single CSU telecommunications programmer was able to build a production-class voicemail application for the residence halls on top of the basic service.[1] This success gave us confidence that other applications can be developed rapidly and cost-effectively.

The voicemail application provides both a traditional telephone keypad–based interface and a Web-based interface for managing voicemail, including user authentication, end-user message management, storage, and retrieval. The Web interface virtually eliminates the need for telephone access to the voicemail service and offers remote access to voicemail messages from any networked computer. The Web interface has become a personal favorite of this author as a result of the e-mail alerts and the ability to manage voicemail from home.

CSU's concerns relating to maintenance costs, managing message storage capacity, and single point of failure on the legacy voicemail system have been mitigated by this open source solution. Commercial, moderately priced, off-the-shelf servers provide system redundancy, fail-over capability, and relatively inexpensive message storage capacity. With this configuration, voice messages can be stored similarly to e-mail messages. The open source IP voicemail system is currently connected to our legacy PBX, in tandem with the legacy voicemail system used by faculty and staff. We are confident that it can natively interface with a SIP proxy server once it replaces the legacy PBX. However, it can also remain connected to the legacy PBX as a replacement to the legacy voicemail system.

Interestingly, CSU has yet to fully exploit the primary purpose of the open source IP PBX—voice service. To date, we have used it for voicemail, campus announcements, music-on-hold, and as a gateway for call routing to Internet2-affiliated institutions. We have plans to expand the network gateway functionality to support SIP trunking to the PSTN. We continue to monitor the evolution of the open source IP telephony space to ensure that our ultimate model is sustainable and institutionally appropriate. We monitor the progress of other institutions engaged in open source IP telephony and expand our own architectural model based on their and our experiences.

CSU Is Not Alone

The University of Pennsylvania is migrating to open source IP PBX from its carrier-based telephone service. The project, entitled iPhone (http://www.upenn.edu/computing/voice/voip/), uses Cisco's IP Phone 7940G and has four pilot phases scheduled from October 2005 to summer 2007. The final phase promises full production service to faculty and staff who meet the program prerequisites. The production service is anticipated to offer, among other services, voicemail, authorization codes, Web-based customer feature management, and soft (IP) phones.

According to a September 2006 article in *LinuxWorld*, Sam Houston State University (SHSU) was in the process of replacing an existing VoIP PBX, the Cisco Call Manager, with an open source IP PBX (Hochmuth, 2006). This is an interesting twist on the more common strategy of replacing an existing legacy telephony PBX with an open source IP PBX. SHSU plans to convert all 6,000 faculty and staff telephones from its current Cisco Call Manager and Nortel Meridian PBX to Linux servers running an open source IP PBX application. Its strategy is to deploy open source IP PBX servers for call processing, voicemail, and a PSTN gateway.

SHSU cites a common reason for implementing an open source IP PBX solution. According to senior voice analyst Aaron Daniel, "The driver for this project was cost" (Hochmuth, 2006). SHSU expects to save costs by eliminating the licensing fees required to keep the Cisco Call Manager network operating. Each Cisco IP telephone connected to the Cisco Call Manager requires a separate license. However, SHSU will be able to attach its Cisco IP telephones to the open source IP PBX without a concern about licensing fees. SHSU has already moved 1,600 telephones to the open source solution. In addition, SHSU expects to save by decommissioning the Nortel Meridian PBX, eliminating the costs of cooling and electrical resources required to operate the Meridian PBX. However, Daniel acknowledges that accurate and comprehensive documentation is essential because the open source IP PBX installation is only supported internally, except for the Dell servers. He also recognizes that some major features are missing and must be internally developed, such as secretarial functions that allow an office administrator to manage and answer multiple telephone extensions. To fix this, SHSU is investigating development of a routine to handle multiple lines.

These are just two examples, among several that we are aware of, where institutions of higher education are engaged in open source IP telephony.

What IP Open Source Telephony Is NOT

Our experience indicates that open source IP telephony is not as mature as legacy PBX technology, which has a long and documented history of reliability and support. With open source IP PBXs, we have reservations regarding reliability, scalability, and robustness. CSU is still in the exploratory and architectural definition stages on its open source IP PBX environment, however, and we fully expect that our experiences and observations combined with those of other institutions and commercial entities that are engaged in open source IP telephony will provide us with a good roadmap toward emerging services and technologies. Clearly, we will not implement a production PBX that cannot meet the expectations of CSU.

Fully managing an open source IP telephony environment will pose additional challenges. We will increasingly be undertaking responsibilities traditionally reserved for PBX manufacturers. I believe, however, that the adjustments will be offset by the agility to create services, adhere to standards, exercise control, and lower the overall life-cycle costs of converged services.

What It Means to Higher Education

Understanding your institution's requirements and motivations should be the first step in considering alternative technologies. Without a doubt, when it comes to open source IP telephony, the devil—along with the risk—is in the details. For example, unless you elect to purchase support services from the open source IP PBX vendor or software distribution provider, if these are available, your staff will assume full responsibility for this new telecommunications environment. In many institutions, staff members are already fully engaged, and additional support requirement and technical demands might become an overwhelming burden.

Further, because open source IP telephony represents a true crossover between IP technologies that have typically been the purview of the IT department and legacy systems from the "telephone guys," running open source IP telephony can accelerate the need for the two technology cultures to collaborate or identify gaps in requisite skills. The skills required to support an open source IP PBX, its voicemail, or its gateway application, for example, might not be found within the traditional telecommunications staff, and, at the same time, the requisite telephony skills associated with PSTN trunking and station work are typically not present within the traditional IT staffs. Regardless of whether these two organizations are structurally integrated, their collaboration should be seamless to ensure a viable large-scale implementation of an open source IP PBX. In addition, customizing an application inherently renders outside support more difficult and more expensive. The risks traditionally associated with customized software are applicable to open source IP telephony—for example, maintaining up-to-date documentation; finding a solution for loss of institutional memory as a result of personnel retirements or transfers; developing robust change-management, testing, and upgrade policies and procedures; and so forth.

While the price tag for open source IP telephony may be very attractive compared to the cost of a traditional PBX installation, it is crucial that institutions consider the hard and soft costs of support, organizational impacts, security issues, and procedural changes. Further, electing to increase in-house staff, hire consultants, or contract for support will certainly have an impact on the total cost. An institution also assumes the responsibility for monitoring and executing unavoidable upgrades to the hardware and software associated with the open source IP telephony environment. We understand that there are costs and operational perils that CSU has yet to discover.

While there are clearly implications for IT/telecommunications staff operations, one must also consider the impact on the end-user experience. After one year of open source–based voicemail service to the CSU residence halls, we have experienced little negative feedback. If we were to extend this service to the entire campus, however, the results might be somewhat different. We know that retaining the “look and feel” of the current legacy telephone–based voicemail interface is very important for some faculty and staff members. However, aside from learning to use a new telephone instrument, we expect the end user to have a transparent experience.

CSU has yet to explore whether an open source IP PBX can scale to support thousands of users, although we have designed an architecture that can test it. We will be exploring the ability to route large call volumes and support Automatic Call Distribution groups that are critical to our campus operation. Lastly, we will be investigating E911 support, since CSU has its own Public Safety Answering Point. We view these tests as the next check point in a systematic investigation and implementation of open source IP telephony.

Once open source IP telephony is deployed, basic telephone service will be a service on the convergent network rather than a discrete service on separate network infrastructure. Further, once the monolithic legacy PBX infrastructure is replaced by a server-based open source IP telephony model, telecommunications departments become an “integrated IT-centric” operation that provisions telephone services on a network nominally shared with video and data services.

Can an open source IP PBX be as stable and reliable as the legacy PBX? It will depend on the architectural and operational choices made. For example, institutions may be well advised to implement redundant and/or clustered Linux servers, as well as institutional security, upgrade, and operational policies. Some may argue that servers are less reliable and more susceptible to operating system and network vulnerabilities than are legacy PBX systems, but once open source IP telephony is treated as a mission-critical application, then relevant industry and/or institutional strategies and policies can be employed to improve stability, reliability, and availability.

In fact, will future “desktop” telephony have to adhere to the traditional five nines (99.999 percent) reliability? The proliferation of mobile devices and soft phones that can use the wireless and/or cellular networks may minimize the reliance on desktop telephones. Transitioning to an open source IP PBX in a convergent environment may challenge the reliance on the desktop telephone. The use of text and instant messaging on mobile devices and computers yet again increases the options for communicating. In a convergent environment, the key characteristics will be multiple modes of seamless and collaborative communications. Reliability and availability will then be a function of the collective convergent network rather than a function of a discrete service or device.

I believe that the emergence of open source IP telephony is an opportunity for higher education decision makers and technology professionals to venture, at least intellectually, outside the box regarding telephone services and the concept of convergence. The opportunities and challenges at this juncture are likely to hold intellectual and practical appeal in an academic environment. We voluntarily reside in an environment that seeks to expand human knowledge through exploration and discovery. We have a charter to support the missions of our institutions through the application of technology. Within the context of either of these paradigms, should efforts to avoid reasonable risks be a deterrent to seeking the best technological solutions for our institutions?

The strategic choice is ours to make.

Key Questions to Ask

- What are the risks and rewards for our institution to pursue open source IP telephony?
- How can we compare stability and reliability between an open source IP PBX and a legacy PBX?
- What are the critical factors for achieving a successful open source IP telephony implementation?
- What is the institution’s vision for convergence?
- With the emergence of a broad range of ubiquitous communication strategies, how critical is the traditional telephone concept of five nines (99.999 percent) reliability?

Where to Learn More

- Brisard, A. (2006, Spring). Leveraging SIP within existing networks: Reflecting on the university network. *ACUTA Journal of Communications Technology in Higher Education, 10*(1) 24–26.
- Hutcheson, Z. (2004). Is open source IP telephony ready for prime time? YES. *Network World*. Retrieved January 2, 2007, from http://www.networkworld.com/columnists/2004/0607faceoffyes.html
- Kapoor, T. (2006, March). SIP: A key to convergence in the enterprise. *SIP Magazine*, *1*(2).
- Malik, O. (2006, February 23). Here comes open source telecom. *GigaOM*. Retrieved January 2, 2007, from http://gigaom.com/2006/02/23/here-comes-open-source-telecom/
- SIPfoundry. (2006, May 19). How to compare sipX ECS with the Asterisk PBX (sipX vs. Asterisk). Retrieved January 2, 2007, from http://sipx-wiki.calivia.com/index.php/Comparing_sipX_with_Asterisk
- Tehrani, R. (2004). Open source telephony…the next big thing. *TMCnet.* Retrieved January 2, 2007, from http://tmcnet.com/it/0104/0104PO.htm
- Van Meggelen, J., Smith, J., & Madsen, L. (2005). *Asterisk: The future of telephony*. Sebastopol, CA: O'Reilly Media.
- Voice over IP. *Wikipedia, The Free Encyclopedia*. Retrieved January 2, 2007, from http://en.wikipedia.org/wiki/Voice_over_IP

References

- Garcia, A. (2006, December 1). Outlook 2007: VOIP. *eWeek.com*. Retrieved January 2, 2007, from http://www.eweek.com/article2/0%2C1895%2C2066822%2C00.asp
- Hochmuth, P. (2006, September 18). University dumps Cisco VoIP for open-source Asterisk. *LinuxWorld*. Retrieved January 2, 2007, from http://www.linuxworld.com/news/2006/091206-von-sam-houston.html
- Mockapetris, P. V. (2006, April). Telephony's next act: Moving from circuit-switched networks to the Internet will be the biggest challenge phone systems have ever faced. *IEEE Spectrum, 43*(4)*,* 28–32.

Endnote

1. The CSU software programmer coded the interface between the open source voicemail and legacy PBX and incorporated the message-waiting indicator function to light the “message waiting” lamp on feature telephones. The light is extinguished when the voicemail message is read via the Web interface or listened to on the telephone.

About the Author

Jose Joaquin Valdes, Jr. (jose.valdes@colostate.edu) is Associate Director of Telecommunications at Colorado State University.