Communicating With Sam

Convergence at the System Level

Sam Johnston

Question:

We are currently looking at replacing several telephone systems that are no longer supported by the manufacturer. We have a large PBX at our head office and mostly basic key systems at our remote sites. We are evaluating traditional systems and IP-based solutions for both the head office and remote sites. One aspect of the IP solutions that we don't like is the re-introduction of Windows based servers. We have spent considerable time and effort to centralize and consolidate most our data servers onto a System i. Is there a way we can leverage this investment to support our voice needs?

Answer:

The short answer is yes, absolutely. Like many new technologies IP communications has evolved significantly from the time it became a part of the enterprise communications infrastructure in the mid 1990's. We started out by enabling systems to send data to each other using TCP/IP over the Internet. Application layer access to data was provided via Web Servers based on the open standard of http. This allowed any browser that supported the http standard to display information. It didn't matter if you were using Internet Explorer or FireFox. Initially the video and voice worlds consolidated onto IP networks using standards like H323 and MGCP. While this allowed communications over the network, many of the solutions still used proprietary standards like SCCP (Skinny) to work with end-points. This meant that you either had to buy the solution vendor's end-point or ensure compatibility independently, taking on the associated support challenges.

The past few years has seen a tremendous explosion in voice and mobile applications and devices over the Internet. VoIP has become a publicly recognized acronym thanks to endless repetition of TV commercials promoting Internet voice communications. With this growth, and the acceptance that VoIP will eventually be the standard for all voice communications, the importance of open standards has emerged to ensure that disparate systems can communication over a single network. As a consequence, Session Initiation Protocol (SIP) has emerged as the application layer protocol of choice for end-points, and now promises to do for voice applications what TCP/IP did for data applications by enabling any voice device anywhere to communication with any system.

So how will this transform voice communications? The traditional Public Swtiched Telephone Network (PSTN) has a very centralized intelligent network and dumb end-point devices (phones) that need to be directed by the centralized network to communicate with other end-points. SIP is a peer-to-peer protocol that puts the intelligence in the end-point, and allows devices to use a simple and highly scaleable Internet, or IP, network. SIP is similar to http and shares some of its design principles. It is human readable and request / response structured. SIP proponents also claim it is simpler than H.323. SIP shares many http status codes, such as the familiar "404 – not found", making it very user friendly. SIP is also not limited to voice communications. It can accommodate many types of communication sessions from voice to video and potentially many future applications.

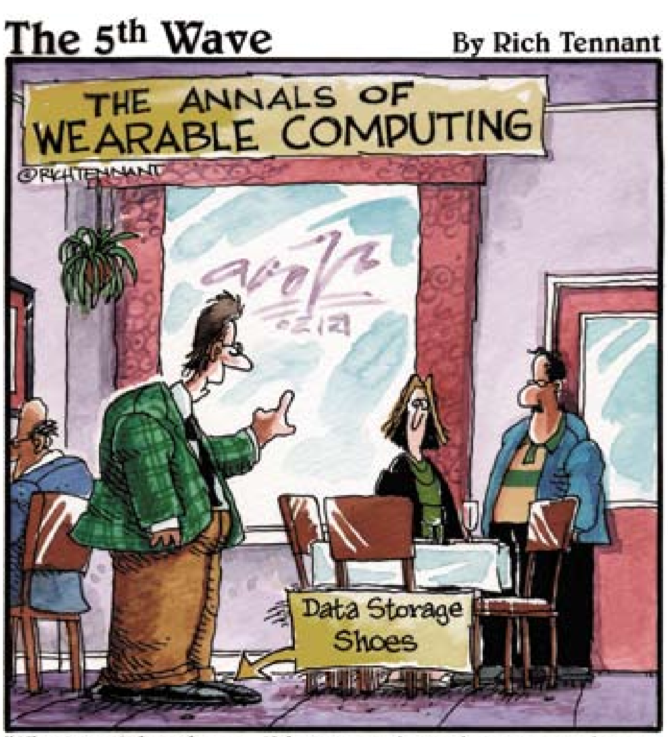
With the evolution to SIP as a protocol standard for end-points, the question becomes: How quickly will the applications evolve and adapt in order to fully leverage the robust service provider and enterprise IP networks? Most IP telephony applications have been around since the late 1990's and have evolved into robust and reliable communications systems with functionality that far exceeds that of traditional systems. Initially most IP telephony applications, like Cisco's CallManager, were based on some form of Windows OS and usually came baked onto an appliance. Recently, as IP telephony solutions have moved from bleeding edge to mainstream, there has been a significant transition to the Linux platform and open source for IP telephony applications. The dominant IP telephony product, Cisco's CallManager, in the most recent 5.0 release is now Linux based, and supports SIP. While Cisco is embracing some aspects of an open standards world, they are still using enterprise market share clout to control how open customers can become. On the other hand, many smaller niche application providers have emerged in the IP telephony space with a true open source approach, obviously hoping to use this flexibility to nibble at Cisco's dominant market share position.

It is clear that the tie to the appliance method of deployment is being broken. The open source, open standards approach opens doors, allowing customers to select their own server platform and integrate this into their environment as they choose. Since System i runs Linux natively, this means that there is the potential to consolidate our voice applications onto a server platform that is more reliable than Windows. The natural question is: Will Cisco compile CallManager for Linux on POWER for System i deployments? Not only is Cisco the dominant IP telephony product, but also number one in market share in the enterprise space, including traditional voice systems. As the dominant enterprise player, it would be ideal if Cisco was to embrace the System i — but this does not appear to be on the horizon today.

Despite Cisco's reluctance to embrace all versions of Linux, there are enterprise class options for System i advocates. Recently IBM and 3Com announced the 3Com IP Telephony Suite for IBM System i. The 3Com IP telephony product is a mature solution, and while 3Com may lack Cisco's clout to gain market share, it should not be discounted as a niche player that is incapable of playing in the enterprise space. All components in the 3Com telephony offering — including VCX call control, IP Messaging (which is unified messaging), IP Conferencing (audio or video), and IP Contact Center — are based on the SIP standard. This is what differentiates the 3Com offering from others currently available, which are based on multiple operating systems or proprietary protocols. 3Com VCX uses only SIP and runs it natively. All of the server side (traditionally done on a PBX) solution elements are software applications. Customers now have a suite of converged applications to mix and match to address their needs. Convergence has been defined as the ability to combine voice, data and advanced applications on a "single network". This announcement takes convergence to the next level by extending it to include running on a "Single System I".

At first thought, the notion of running your phone system on a System i that is the host to your enterprise applications seems a bit far fetched. While it is certainly not a mainstream strategy at this time, and perhaps not the compelling reason you may need to migrate to a System i, for loyal System i customers the benefits are strong and play on the overall strength of the platform:

- **Server Consolidation:** Your organization has already assessed the benefits of consolidating servers and has determined that there is ROI for your business, and by leveraging existing infrastructure you will eliminate the costs and complexity of managing additional servers. The fact that the System i is generally regarded as one of the easiest and least costly platforms to manage will make the decision easier. By leveraging the existing OS on the System i you will also take advantage of existing skills as opposed to introducing (or re-introducing) either Windows or an alternative Linux version that will require support, operating and maintenance.
- **System Availability:** High system availability is a major factor in most decisions to implement the System i. In addition to the highly reliable platform itself, as the core of your data centre, the System i will be in an environment that is likely to create a highly available voice system. This includes residing in a data centre with proper power protection, perhaps back-up generation and environmental controls. Back-up and recovery investments can be leverage, and system high availability can be easily extended to phone users and voice applications.
- **Centralization:** Due to the strong communications heritage of the AS/400, many System i servers reside in highly distributed organizations that leverage networks to service remote offices via a centralized data and application platform. This means that most organizations will have the appropriate network infrastructure in place to leverage a centralized voice communications strategy. Adding phones to the remote branches and processing calls from a centralized voice system can provide several cost savings areas: the elimination of remote site PBXs and the associated operating and support expenses; aggregation of voice circuits at a central bank of lines, and; the ability to place calls over the network to avoid long distance charges.



"I'll be right there. I just need to do a core dump into my shoe first."

Convergence at the system layer can be attractive in the right situation, especially given the history of the System i platform for stability and simplicity in terms of operations. However, you need to make sure that benefits truly will be derived by your organization versus the potential added complexity associated with administering voice users on a system that may also host complicated data applications. One of the biggest challenges will be the service model of finding a way to manage the need for rapid response to the relatively high volume of mundane changes associated with voice communications that will need to be administered by an IT group that normally deals with rigid and meticulous change management processes.

The potential to leverage the System i to deliver voice applications is yet another example of the platform's ability to adapt to the changing needs of enterprise customers to maintain relevance as a strategic platform. We can look forward to a day very soon where Web, IP telephony and Business applications will converge and work seamlessly together on a System i, allowing organizations to deploy these applications more cost effectively, with less organizational resources. TUG

__Sam Johnston__ is a partner and Chief Technology Officer of Intesys Network Communications Ltd., providing value-added networking and e-commerce solutions to the iSeries community. He can be reached at (416) 438-0002 or via e-mail at sjohnston@intesys-ncl.com. Any TUG member wishing to submit a question to Sam can forward their typewritten material to the TUG office, or to Intesys. The deadline for our next issue is Friday June 16, 2006.