Voice over IP Manageability

Architecture Matters

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Comparing the Various Approaches and How They Deliver System Administration

In 1996, Gartner Group rocked the IT world with the results of a study showing that the average PC cost an enterprise $10,000 per year – five times the purchase price of the hardware and software – to own. Since then, the steady commoditization of computer equipment has steadily increased this alarming ratio. Clearly, if you want to reduce the total cost of ownership (TCO) of a technology platform operated by end users, you must address the human resources required to operate, manage, maintain, and support it. Manageability is paramount.

Voice over IP is a case in point. The fundamental concept is to stop maintaining two parallel networks for voice and data and instead have a single, converged infrastructure that leverages sunk investments, exploits economies of scale, and streamlines administration and management. A VoIP solution that is truly easy to implement and manage can pay for itself very quickly – sometimes in a matter of months.

However, manageability depends to a great extent on the underlying architecture of the voice system. Some platforms are extensions of legacy voice switches, others are built on top of legacy data technologies, and still others were designed from the ground up for converged voice and data.

In this paper, we examine the impact these very different starting points have on management, consider what should be possible when voice is managed via IP, and look at what actually is achieved by the holistic management you can get only from a solution that is VoIP by design.

Historical Context: The Voice Silo Conundrum

Voice communication is something we take for granted in the workplace. The phones all have the same basic interface, and we can walk into a new office and pick up the phone and use it. However, this ease of use on the front end was achieved by putting all the complexity on the back end, in a proprietary "black box" that is very difficult to manage.

This problem is compounded if yours is a mid-size company with multiple locations, because each site typically has a separate voice system, supported by a separate service contract from a local teleconnect. Some sites have outsourced everything to Centrex service, which is very expensive.

You may have as many service and support contracts as sites to manage, which is highly inefficient. This patchwork of independent voice silos tends to be worse if your company has grown by acquisition, as so many have these days. The acquired sites came with established and often obsolete voice platforms. It can even be hard to find a local technician who understands these aging black boxes well enough to work on them.

Worst of all, you have no control over your own voice network. You are completely dependent on outside service companies for even the simplest of administrative tasks, such as moves, adds, and changes. Placing an order and then waiting for the truck roll can take weeks. In voice-intensive businesses, holding new or relocated employees somewhat incommunicado this long can have a significant impact on productivity, and hardly helps morale.

Migrating these disjointed voice silos to a single, unified VoIP system that runs over your existing data infrastructure can save a fortune in local service contracts—as long as it doesn't pack a big management wallop.

Consolidating Voice and Data Networks

Theoretically, running voice as packet traffic on an IP network means conveying some of benefits of IP to the voice world. These include easier administration and open systems. However, all IP telephony is not created equal, and fundamental architectural differences have big implications for the inherent manageability of particular VoIP platforms.

VoIP solutions from the major voice and data vendors are typically retrofits—adaptations of legacy platforms originally designed to handle one type of traffic or the other. Various new functions are often added by acquiring existing products and technologies and then cobbling them together.

This has serious implications for management, because the different components of the consolidated voice system—such as voicemail, network call routing, and client applications—are managed by different tools. Some have a Web-based interface, some are graphical but not Web-based, some are controlled by specialized applications, and some require command-line entries and scripting. There is no single, truly unified management.

Some of the retrofits attempt to provide the appearance of unified management by spray-painting a single Web-based interface over everything, but it's just a disguise. All the complexity and cryptic or confusing vendor-specific nomenclature has just been moved from one interface to another.

It hasn't reduced the level of expertise or amount of training administrators must have to run the voice network. With VoIP, simplicity is expected. You should be able to take in-house IT staff who have only a general knowledge of voice systems and data switches, give them minimal training, and have them manage your phone network as part of their daily tasks without needs for additional staffing.

This can be realized with a system designed from the outset to be a VoIP platform. You get a single-system image of the entire network, via a single, holistic management interface. Everything from the simplest administrative tasks to high-level network management and configuration is done through the same Web-based interface. This greatly reduces training requirements and the skill levels your network professionals must have.

Network managers and administrators get complete visibility into all of the physical equipment through the same intuitive interface that is used for adding new users or administering voicemail. And because the interface is Web-based, they can manage any part of the multi-site network from anywhere. Administrative tasks are further simplified with web-based forms that have your default values already entered.

Easy to Manage = Easy to Implement

The same factors that make a VoIP platform easy to manage should also make it easy to implement. This is very important if your company is going to pursue a policy of growth by acquisition, or you want to migrate existing sites gradually. When VoIP solutions are based on legacy voice or data platforms, rolling one out at a new site can require an entire team of specialists from the vendor and local reseller. And when this is the case, you will be forever dependent on these outside parties for management—count on it.

In contrast, a voice platform architected to be VoIP *by design* can have a management interface that pulls all the settings needed to add a new user or switch onto a single Web-based form, making it much easier to complete common tasks. In fact, such an interface can be so simple and intuitive that a basic network administrator with no voice background can be brought up to speed with very minimal training. Such an individual then has all the skills necessary to roll VoIP out to new sites without assistance from outside experts.

To add a new user, administrators simply type the user's name into one user-edit form, and then either accept the default values in the other fields or change some or all of them. New users are given extension and direct-dial numbers, assigned an analog port or designated an IP phone user, and set up with a voicemail box. Access privileges are defined by selecting the level of client software the new addition will be using and placing the individual into various user groups.

The software automatically generates an editable user ID by applying the company's naming convention to the user's actual name, and the new user will then receive an e-mail notification of the new voice system account. This is all done from a single web page.

The process for adding a new switch is similarly easy, and can be done from a switch-editing page within the same management interface. Once a new switch is plugged in, it is configured from a single switch-editing form. The software immediately recognizes the switch and fills in its IP and Ethernet addresses, along with any default parameters that have been previously defined. Often all the administrator has to do is name the switch and accept the values with which the form has been automatically populated.

In the event that a switch in service should need to be swapped, the administrator goes to this same switch-editing page for the old switch and changes the IP and Ethernet addresses to those for the new switch. All the configuration details for the old switch are then automatically picked up and burned into the flash memory on the new switch.

A rapidly growing regional bank with branches on both coasts provides a real-world example of just how powerful this type of management interface can be. The bank's CTO was able to take a Windows NT administrator and turn him into the voice network manager after one day of training.

This one individual replaced a whole series of local service contracts and Centrex service. He is now managing the entire 12-site network from one office, and can handle the cutover for newly acquired branches quickly and without assistance. When management is simple and straightforward, cutovers can be simple and straightforward.

The Converged World Is a World of Peers

One of basic concepts of the Internet from its very beginnings is the use of a distributed architecture that is inherently resilient. The network's servers are peers, so they can cover for each other. We address this reliability issue at length in a separate paper. Suffice it to say that when VoIP solutions impose a hierarchical management structure on what is fundamentally a peer environment, their systems fail to exploit one of the Internet's key strengths.

Nevertheless, many VoIP vendors centralize management at one location on the network, because the legacy platforms they are adapting have such hierarchies. Creating a peer-to-peer architecture requires greater initial design and engineering effort, but once it is delivered, it provides a level of reliability that can't be matched in any remotely cost-effective way by a centralized, hierarchical system. The hierarchical approach creates a single point of failure, and it also tends to result in a lot of truck rolls (or "foot rolls") to the location that hosts the management system.

The challenge is to deliver a peer solution that is inherently resilient and yet still easy to manage. The solution must make a collection of peers function as a single system, and harness them under a simple management tool. Such a peer architecture also makes it inherently easier for a VoIP platform to support the emerging class of distributed workgroup applications, so you can embrace them without incurring a lot of additional management overhead.

Does Voice Really Ride for Free?

Despite the dramatic drop in long-distance rates in recent years, VoIP converts report that there is still a lot of gold to be found in the toll-bypass hills. However, setting up the network call routing for toll bypass must be quite easy to do. Otherwise, the management overhead can cancel out a lot of the toll-bypass savings.

In VoIP networks based on legacy voice and data platforms, automatic peer-to-peer exchange of newly entered routing information doesn't happen. Routing information has to be entered into routing tables, possibly even in each individual PBX or physical router. Small, localized groupings of such devices may automatically update each other, but the information doesn't get pushed out to every switch in the network.

Given the capabilities of the IP environment, network managers shouldn't have to define specific routing behaviors for each location, and users shouldn't have to remember which area codes qualify for toll bypass. Long-distance calls to outside numbers that fall within the local dialing radius of one of your company's sites should automatically get routed over your IP backbone, in a process that is entirely transparent to the caller.

When the VoIP solution is one single, distributed system, an intuitive graphical interface replaces routing tables, and the toll-bypass routing can take place intelligently and automatically. Then, the system can do what you mean instead of what you say.

Easy to Use = Easy to Manage

In the traditional voice world, most companies barely scratch the surface of their PBX's capabilities, because the features are too hard to implement. Taking advantage of them requires too much management help, and too much user training.

In the new world of VoIP world, self-service is one of the key metrics for measuring alternative platforms. The best VoIP systems have a single intuitive user interface that enables non-technical users to help themselves to such functions as call management and setting up conference calls. With this same interface, they can "hotel" themselves at temporary locations. All voice-system features and user-profile information automatically get conveyed to the new desk, so there is true location transparency.

When IP phones are being used, self-provisioning is taken to an even higher level, and all administrators have to do is add new users to the system. Then the users can plug any IP phone into the network jack in a new or temporary location, log in, and start receiving calls. All the features the enterprise voice system has to offer are immediately available, assuming the necessary access privileges.

When users can take care of themselves, they make fewer support calls, and there is less of a burden on network professionals. Users are more productive, and the IT staff can be freed up for other functions.

Conclusion

If we want to lower the cost of technology, we must make it simpler and easier to use and manage. It has to require fewer managers per user, and lower skill sets per manager. When voice platforms have been created as VoIP solutions *by design*, you get a seamless system that is distributed, reliable, and easy to manage and administer. You don't have to be a Fortune 500 company to have an enterprise-class voice system you can afford to maintain and operate internally.

INTELLIGENT PHONE SYSTEMS ™

960 Stewart Drive

Sunnyvale, CA 94085

(408) 331-3300

1-800-425-9385

Fax: (408) 331-3333

Email: info@shoretel.com

www.shoretel.com