# Communicating With Sam

## Convergence Final Frontier:

# Consolidated Voice Systems

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### Question:

We have recently consolidated servers from our regional offices; including several iSeries, and Exchange servers to a completely central model. This has resulted in a significant decrease in the number of severs. In addition we have centralized our Internet access for all our locations. We have scaled our IP WAN to support the increased traffic. The support and operations savings of supporting all systems in a single data centre and the economy of scale benefits associated with fewer larger systems has significantly outweighed the added network cost. Currently each branch has a PBX and we are looking for ways to reduce the administration and support costs of these systems with external suppliers. Can we do the same with our voice services without risking availability?

### Answer:

It is really quite incredible how voice and data have cyclically diverged and converged over the past twenty or so years. In the early 1980's when the first generation WAN networks started to emerge often the small amounts of data traffic were transported over dedicated voice trunks that linked remote offices – what was known as tie-lines. High long distance charges were the incentive to implement dedicated trunks to enable toll bypass while efficient X25 traffic from centralized mainframes were easily handled by the robust voice networks of dedicated trunks. Growth in IP GUI applications, and in particular the advent of e-mail, flooded networks in the 1990's leading to dedicated networks for data and a proliferation of new servers at the edge of the network, while increasingly low long distance rates all but eliminated the need for tie-lines.

Well here we are in 2004 and what is old is new again, with a renaissance in voice and data convergence, albeit for different business justification.

The easy answer to your questions is that absolutely you can centralize your voice processing similar to your data architecture and it's easily done now that you have a fully centralized IP WAN. The core technology that will enable this strategy is IP telephony, which uses the communications foundation of the Internet to transport voice conversations alongside corporate data. Unlike traditional circuit-based analog phone service, IP telephony segments voice conversations into separate digital packets allowing them to be transported over the IP WAN, eliminating the need for dedicated trunks and ports which make centralizing traditional or TDM calls all but cost prohibitive. In this way you can eliminate the costly long distance charges associated with inter-branch communications using traditional PBX and PSTN technology. This feature also allows you to minimize the number and cost of voice trunks required for voice communications by moving inter-office voice traffic to the WAN and eliminating the need for a voice trunk at each end to support an inter-office call. Eliminating multiple PBX hardware at each remote will also reduce the annual cost of maintenance and simplify the support for voice services by centralizing activity such a user administration.

The question we always get in terms of centralizing voice processing, and adopting a fully converged model is. "Will I be bleeding edge?" The answer is rapidly becoming, "You won't even be leading edge." According to Cisco, the market leader in IP telephony and VoIP, more than 14,500 organizations now use Cisco IP Telephony products to eliminate costly, inflexible, and redundant proprietary circuit-switched "PBX" office phone systems, while most traditional manufacturers such as Nortel, Avaya and Alcatel rarely ship a traditional system that is not IP-enabled for VoIP.

In the Cisco world, Call Manager is the heart of the IP Telephony solution. It is software running on a Windows 2000 server that extends enterprise telephony features and capabilities to devices such as IP phones, media processing devices, voice-over-IP (VoIP) gateways, and multimedia applications. A Call Manager can handle the call processing functions for your central location and remote branches. This will eliminate the multiple PBX support and administrative issues. Call Managers can be grouped into clusters to provide additional levels of redundancy, given that each remote will now rely on a single processing heart for all voice calls. The same availability principles that you likely have applied to Exchange would apply here.

Once you have centralized call processing, a byproduct will be that calls between offices will now be placed over the WAN without long-distance tolls. Call admission control (CAC) will now become crucial as it ensures that voice quality of service (QoS) is maintained across WAN links, and automatically diverts calls to alternate public switched telephone network (PSTN) routes when WAN bandwidth is not available.

While CAC and QoS have dramatically improved the call quality to eliminate the first generation weakness of VoIP – echo – it still did not alleviate the large concern with availability. Concern has continued over the potential for mission critical timing sensitive voice transactions being impacted by even small outages in the WAN and users being unable to reach the centralized call processing. Manufacturers such as Cisco have addressed this issue with "skinny" versions of the call processing software for the voice gateways (the WAN router) at each remote to act as temporary back-up systems. In the Cisco world, when a central Cisco Call Manager cluster also handles call processing for users at distributed sites, administrators can help ensure continuous phone service using Cisco Survivable Remote Site Telephony (SRST), Cisco IOS software for Cisco routers. (See **Figure 1**.) If the IP WAN link fails, SRST in the router provides the basic telephone capabilities until the link is restored. The Cisco SRST Telephony Software operates by taking advantage of the keepalive packets sent from both the centralized Cisco CallManager cluster and local Cisco IP Phones. If the WAN link fails, the Cisco IP Phones detect that they are no longer receiving keepalive packets from the Cisco CallManager. The call processing functions are performed by a SRST router until the link is restored and the phone starts to receive keepalive packets again.

Manufacturers such as Cisco have further enhanced the business case for centralized telephony by developing a wide array of applications that work in conjunction with IP Telephony and IP Communications products. Cisco Unified Communications, for example, offers such personal productivity tools as Cisco Unity, a platform for unifying voice and e-mails into a single message store for retrieval of either message formats from a multimedia PC or a telephone using text-to speech technology, and Cisco Personal Assistant. Both are aimed at managing the large volume of messages employees receive from an assortment of telephone, fax, email, pager and other devices. As you have already centralized your Exchange environment, it will be a natural and simple extension to leverage this platform as a single message store for all users, local and remote, to achieve centralized voice mail in addition to consolidated e-mail services.

Another opportunity to leverage Exchange is the Cisco IP Phone Address Book Synchronizer that is included in Call Manager V4.0. It allows users to synchronize Microsoft Outlook or Outlook Express address books with Cisco Personal Address Book. The Synchronizer provides two-way synchronization between the Microsoft and Cisco products. After the user installs and configures Cisco Personal

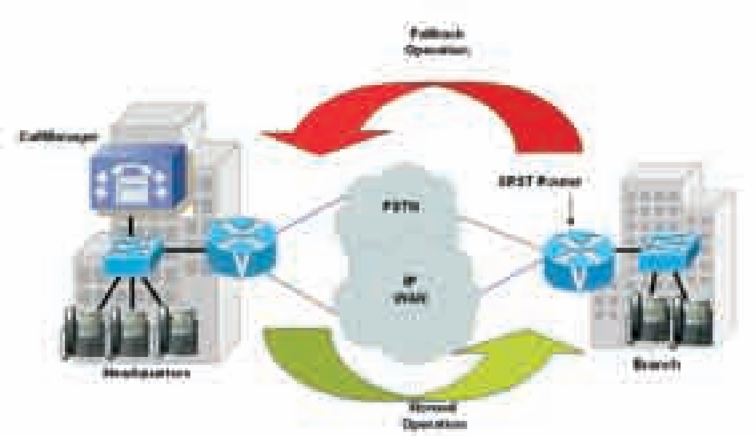


*Figure 1: SRST Operation*

Address Book, users access this feature from the Cisco IP Phone Configuration web page. Cisco also offers technology for contact centers as well as audio and video conferencing. By combining wireless networking equipment and the Cisco Wireless IP Phone, your employees that are mobile within your enterprise can always have their office phones by their sides.

Aside from the obvious technical support and operations benefits associated with centralizing your voice processing, there are many business benefits that increase the ROI associated with any capital investments.

Quantifiable benefits, some of which have been touched on, include the elimination of long-distance between offices and a reduced number of trunks needed to support internal calls. Should you implement off-net dialing for the ability of a remote site to access a trunk over the WAN to dial out in another city to avoid long-distance, then additional long-distance may be avoided. Further trunk optimization may be possible by allowing remotes that are at or near capacity to share trunks at other sites that are accessible over the network in order to handle bursting requirements rather than adding more trunks at the remote. Of course, a single cluster of servers for call processing is simpler to manage and will eliminate costly servicing, operation and maintenance of a PBX at each remote.

While the hard benefits are important, the soft benefits are likely the means to gaining support from the business in terms of business transformation. Centralizing voice services will make it easier to secure the voice environment (often securing voice mail, and sensitive content that may be contained in these messages is often overlooked), and it also ensures the same consistent level of service is available to headquarters, branches, and remote workers to optimize productivity and morale. A centralized solution also enables the implementation of a single universal extension plan to facilitate better internal communication by making it simpler to reach out to another employee. It may even mean that single receptionist is possible to service the whole organization, which can assist by ensuring that calls are transferred to the right resource regardless of location, while locations that currently do not have a receptionist may be able to add this resource for a more personal touch. By unifying your voice and message stores should you implement unified messaging, users will also have a single sign-on for voice and data, and new users set-up will be converged for voice and data for added efficiency.

In the end, we have come full circle and what was old is new again, but like any retro trend there is always a new twist. While there is a mad rush to converge voice and data onto a single network, unlike the 1980's it is voice traffic that is now dwarfed by data traffic and can ride for "free" on robust IP networks. While this obvious point is what most observers focus on, the key twist that has important impact on IT strategies is that after years of using network enabling technologies to distribute data applications and servers, IT managers have now used low cost bandwidth to repatriate these services circa 1980 centralized mainframes albeit with an IP flavour. By adding an IP twist to telephony, this path of centralization is now simple and irresistible for telephony just as it is in the date world. TUG


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