IP Telephony in Branch Networks: The Case for Voice Boundary Routing

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Telephony in Branch Networks Today

Enterprises in banking, insurance, real estate, and retailing have a basic business topology of dozens, hundreds, or thousands of small branch offices that are chartered to service their local communities. Communications-wise, these branches need both local telephony services to connect with customers and business applications such as e-mail, workflow, command and control, or technology support services to support hierarchical corporate infrastructures.

Legacy voice solutions for PBX interworking or combinations of PBXs in regional centers and key systems in branches have typically required costly digital circuits between branches and the regional offices. Too often, because of the cost and complexity, the branch office becomes an island of communications, avoiding any special corporate connection, and offering no corporate-wide features. Legacy solutions from vendors such as Nortel or Avaya don't easily facilitate global dial plans, lack support for important regulatory compliance-required applications such as call recording, unified voice mail, or web conferencing for the corporation, and have no backup facility in the event of local key system failure. Even replacement handsets are dispatched from a special branch inventory since key systems and PBX handsets are not interchangeable, even if they're from the same vendor. Furthermore, the data network infrastructure operates independently. Since so much of the workflow is controlled by central applications, the possibility of integrating telephony and workflow-based data is further complicated.

The ability to deliver low cost and feature-rich performance has historically been a big challenge. Partly because of the limited availability of secure and reliable 'first mile' infrastructure and partly because of the limitations of PBX architectures, those enterprises with this basic business topology have been forced to settle for less than optimal cost structures, and feature-poor solutions that do little more than deliver dial tone.

Until now.

Convergence—the bringing together of voice and data networks for directories, authentication, call control, and the integration of business workflow and processes—is the cornerstone of a new set of requirements for the next generation branch office that will change business significantly. IP telephony and an emerging suite of convergence applications can uniquely deliver outstanding reliability, productivity, and lower costs. These technologies and capabilities are at the heart of the business case to justify a new approach to the branch network.

Convergence Applications

For many small and medium enterprises, IP telephony has made sense because of the economics of self-administered telephone moves, adds, and changes. However, for larger enterprises, IP telephony makes sense as the first in a new suite of applications that are possible from the convergence of voice and data networks.

These applications share more than common networks: they share common infrastructure for call control, authentication, privacy, location, and presence services. Integration with centralized corporate directories and authentication services enable single sign-on for both telephony and other business-critical applications. These so-called "convergence applications" are inexpensive and practical because of standards such as Session Initiation Protocol (SIP) that provide reliable and powerful mechanisms to ensure quality and choice in application design and deployment.

SIP, a key feature of convergence applications, can enable workflow integration, satisfy industry-specific regulatory compliance requirements, strengthen customer interactions, and improve personal productivity. Popular examples of these emerging convergence applications are call recording, centralized voice mail, find me/follow me service, and a business-class instant messaging and presence service.

Call Recording

Many regulated industries such as healthcare, financial services, and government services call for strict adherence to privacy and auditing requirements, that can be facilitated with automatic recording of customer interactions. Some customer service calls conducted in branch offices also need to be recorded for quality and audit purposes. SIP-based call control systems can be configured to duplicate packet streams and terminate duplicate sessions on a storage area network for the corporation, reducing the cost, risk, and ease of compliance management throughout the enterprise. Furthermore, parameters in user registration files can be set so that specific designations trigger recording that avoids casual users such as receptionists and managers.

Centralized Voice Mail

PBX and key systems classically have local services for message storage. However, industry experience in enterprise computing has shown that centralized storage is proving to be a reliable and inexpensive model for information control. Voicemail storage is no different. Centralization can reduce the cost of storage, strengthen corporate audit procedures, and facilitate productivity improvements through global access that enables practical uses such as division-wide (not necessarily PBX-wide) broadcasts. A regional sales vice president can easily deliver important voicemail notices to employees across all appropriate sales offices—if having more timely information can improve sales performance, she can directly improve sales productivity.

Find Me/Follow Me

Users make calls to reach the person that they are dialing. Rarely is their intent to leave a voice message. Convergence applications enable improved call completions by cycling through a calling logic, defined by the called party, that is more flexible and sophisticated than simply call forwarding. Improved call completions can only improve sales, customer service, and business efficiency.

Business-Class Instant Messaging

Business-class instant messaging and presence management provide instant text messaging without the advertising, pop-ups, and awkward screen names of public instant messaging services. Users' status (on a call, typing, or off-line) can be presented to other enterprise users who subscribe to that person's presence.

Many of these applications can be deployed within a PBX environment. Yet the cost of operations, engineering, and management of diverse networks, infrastructures, and technologies has made it impractical for widespread deployment and has severely limited the development of markets for these applications.

With IP telephony however, given the low cost, tight infrastructure integration, and business benefits, enterprises with networks of branch offices will look to deploy these advanced applications for real-time communications that are only possible through convergence and an open, scalable architecture. For a growing number of larger enterprises—especially those with extensive branch networks—convergence applications are the compelling reason for enterprise investments in IP telephony.

Session Initiation Protocol

Session Initiation Protocol, (SIP) is an Internet Engineering Task Force (IETF) draft first proposed in 1996. It is a text-oriented protocol modeled on HTTP that in a short time has been proven by 3Com and the world's largest telecommunications operators to be a reliable and quality approach to establishing real-time connections between network endpoints such as IP telephones.

3Com introduced the world's first carrier softswitch based on SIP in 1999 and was first to market with a SIP-based convergence applications suite for enterprises in 2003.

The SIP protocol is simply designed and easily deployed in enterprise settings. A growing community has created telephones, access devices, servers, gateways, and applications using the protocol to quickly establish a market. 3Com is a member of the SIP Forum which is dedicated to promoting SIP and its role in the communications industry.

Voice Boundary Routing

Boundary Routing Innovation

In the early 1990s, 3Com introduced boundary routing as a method for minimizing routing table size, which in those days carried a huge cost in convergence time, memory, and router processor performance. 3Com innovations in boundary routing and now voice boundary routing are similar examples of applying innovation for efficient and automatic network operations.

Highly-networked industries such as banking, retail, insurance, and healthcare need to balance the economics of local connectivity in the branch office with the trade-offs and benefits of services such as central provisioning, single sign-on, and the convergence-enabled capabilities already mentioned. 3Com invented voice boundary routing architecture to provide flexible alternatives that resolve this imbalance. The corporation's unique approach delivers a distributed communications solution for distributed businesses that combines low cost, high reliability, and an applications-rich enterprise communications suite.

Voice Boundary Routing Philosophy

3Com® voice boundary routing deploys call routing intelligence in multiple hierarchical levels as shown in Figure 1 below. Since the calling pattern in most branch networks usually favors local over global dialing, it is appropriate for local routing decisions to be adjudicated at the local level. In certain circumstances, however, it will be appropriate for call control to move up the hierarchy axis to a central call controller. This is a change in control that needs to happen quickly and automatically. A rapid shift might be necessitated by unusual call pattern requirements, including the opportunity for branch employees to act as call center agents during peak demand seasons such as tax season for financial planners, or disaster relief resulting from a cataclysmic failure of a corporate call center. When needed, call control requests can be forwarded up the hierarchy where local images of service parameters can be made available or where global services can be defined.

Voice boundary routing is implemented in 3Com VCX™ convergence software that can be organized into a variety of topologies to deliver the appropriate level of service, resiliency, and cost:

- The media processing and transport class includes devices such as SIP phones, Quality of Service-aware LAN switches, and media gateways for connectivity to the Public Switched Telephone Network or to legacy digital PBXs.
- The call control class provides the functionality for call setup and teardown using SIP signaling, and consists of the SIP call controller.
- The applications class delivers services such as authentication, authorization, global routing administration, call accounting, and other network-centric services, as well as user-centric services such as voice mail, find me/follow me application logic, and business instant messaging.

In some customer implementations it will be appropriate to combine these elements into a single server and present it as a logical IP-PBX for local service delivery. However, in most deployments, the call controller will be deployed as an overlay to the existing dial tone infrastructure. Digital gateways, and in some cases message lamp indicator gateways, will interface between PBXs and the convergence application suite to enable enterprise-wide applications, IP trunking, and IP telephony to new stations.

Within this "overlay" model, applications are usually implemented on multiple servers to deliver service performance, high reliability, and global coverage. As the useful life of the legacy PBX equipment degrades, IP telephony service to users can deployed, phasing the obsolete equipment out of service.

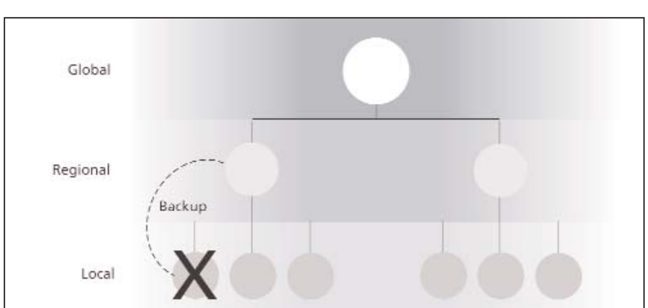


FIGURE 1: Local, regional, and global logic is backed up by the next higher level of the routing hierarchy.

Local Call Routing

For performance, business continuity, or economic reasons, there is a need for a local, self-contained IP telephony device in some networks. 3Com branch devices contain a call controller, authentication server, directory mapping server, web provisioning server, and network ports for 100 MB Ethernet service and both local PSTN and analog terminal connectivity for fax machines or 2500 series-compatible analog devices. The call controller may support the IP telephony requirements of up to 1,000 SIP devices. This flexible design provides a high degree of local service, centralized control, advanced applications, reliability, and the low cost required for distributed business.

In a branch network, the call controller is configured with a set of call routing rules relevant to the anticipated calling plan for a specific location. The call routing information does not need to reflect all of the call routing rules of the enterprise. For example, a controller located in Chicago might only be programmed with dial plans covering the 312, 630, 847, and 815 area codes, while a controller in Dallas might have dial plan information for the 214, 972, 469, and 817 area codes.

Every IP phone is registered with a local call controller that keeps track of location status and personalized configuration details such as speed dials, hunt groups, message routing, and other applications. As the user picks up the handset, the familiar dial tone is delivered to the earpiece and a call request message is initiated between the IP phone and the local call controller via SIP. The local branch call controller detects the number dialed, determines it is a local call known within its call routing rules, and routes it through integrated analog or digital gateway ports to the Public Switched Telephone Network (PSTN) for public network call handling. All the expected call progress messages (line busy or ringing tones) are interpreted and delivered to the display (if so configured) and or earpiece of the handset.

This process is what one would expect with any key system, PBX, or IP-PBX design. What makes a 3Com VCX deployment and voice boundary routing system particularly useful in a branch network is that the local branch call controller is designed to operate as a node in a network of call control servers. The user-specific configuration details and call routing rules are recorded in the database of the call controller and are periodically updated using an asynchronous mechanism with a call controller higher up in the network hierarchy, such as a regional or global call controller.

Central Call Routing

In a situation where an outbound call is initiated and the local call controller does not have the appropriate information at hand, the call request is forwarded to the regional call controller since it contains the image of each of the call routing rules within its domain and is likely to contain the routing table for this particular type of call.

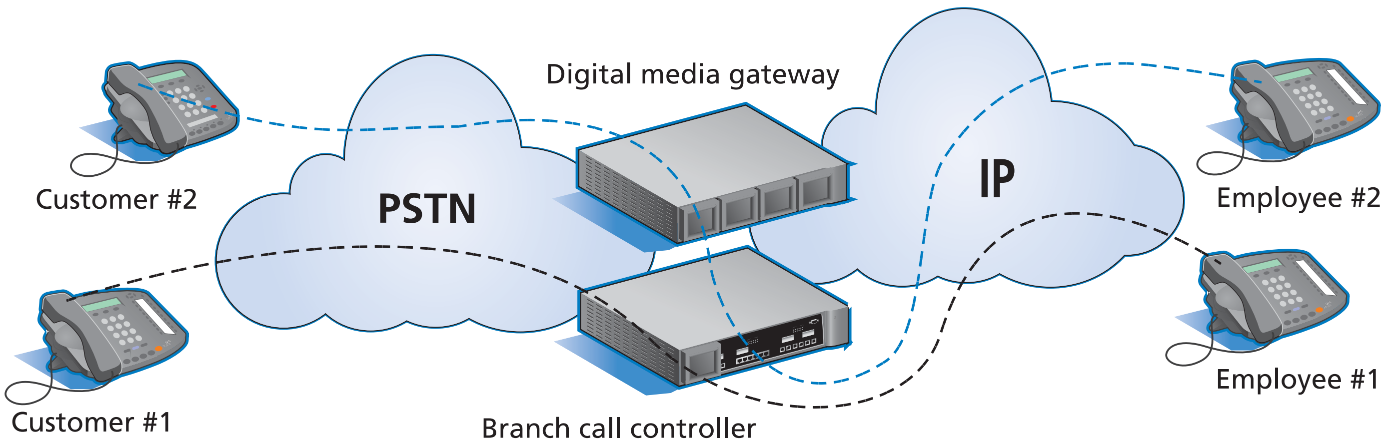
The regional call controller can search the complete compendium of routing tables to discover the optimal call route. If it determines that the appropriate routing is handled by another call controller at a different branch, it returns the IP address of that controller to the originating device and the call signaling passes across the wide area network.

The call is then routed to the remote controller where the local rule is invoked and that local call controller directs the call out to the PSTN. This procedure preserves the least cost routing and toll bypass advantage and utilizes the corporation’s gateway infrastructure to maximize its integrated business investment.

Because the 3Com VCX system utilizes SIP call control, the actual overhead for setting up a call over the IP network is insignificant. The use of local routing tables and their availability in a regional or global context ensures that least cost routing and strategies for coping with branch outages or network congestion are available.

With an enterprise-wide deployment, organizations can greatly improve the reliability of their telephone service for both customers and internal users because there are many available paths known by the network of call controllers, as compared to only those paths known by a single call controller. This constantly updating network-mode of call path determination is a major improvement in call routing services. At any point in time, the network of call controllers knows the greatest set of possible call paths. With knowledge of so many available alternatives, network outages can be easily bypassed.

FIGURE 2: Minimum cost branch design, highlighting the boundary routing benefits of completing the call of a second employee attempting to engage a single local call controller port that is already in use. With 3Com voice boundary routing, call controllers work together to identify and implement the most cost-effective call route.



Furthermore, the distributed nature of the hierarchy inherent in voice boundary routing ensures a high degree of fault tolerance since the call routing database can be routinely and easily backed up on the regional or central call controller.

In some network scenarios, it is important to minimize the number of PSTN connections at the local branch. Even with this limitation, the voice boundary routing system can work as an enterprise solution to deliver lower cost and non-blocking service.

Consider a network design where the objective is to minimize local PSTN connections, yet never present a busy signal should an inbound or outbound call occur at the same time. In this case, the cost of the local PSTN connection in each branch is more expensive than one channel on a digital T1 circuit plus regional toll charges (if any)—not more than one local dial port needs to be provisioned at each branch for inbound local customer calls. The PSTN is configured with a hunt group that forwards calls when busy to a digital gateway scaled to support all branches. Here, the digital-to-packet conversion occurs, and calls are forwarded to the appropriate branch hunt group logic existing on the local branch device. Outbound calls are supported on these digital gateways in regional centers or directly on the platform depending on the economics of long distance versus local PSTN calling.

Similarly, if an employee attempts to engage the single local port that is already in use, the local 3Com call controller can determine the next-lowest cost digital or analog port, completing the call with no hint to the user that service has been routed in the most cost-effective way. This scenario is shown in Figure 2 above.

With this network design, fax calls can be centralized also. Each local fax machine can be assigned a telephone number corresponding to the local branch analog terminal adapter port. Incoming fax calls are terminated on the gateway, packetized, and delivered to the local analog port for the branch fax machine—the fax is printed as if it had been received directly. Alternatively, the 3Com IP messaging solution can deposit these incoming faxes into a user's e-mail box for even greater flexibility and security.

Voice Boundary Routing Benefits

The distributed architecture of voice boundary routing offers incredible power and advantages for business continuity and resiliency in at least four scenarios:

1. In the event of a WAN failure, the 3Com call controller is equipped with all the capabilities to enable local and long distance PSTN calls.
2. In the event of a failure of the local 3Com call controller, local IP phones can automatically register with the secondary call controller higher up in the hierarchy, ensuring business continuity while maintenance procedures resolve the failure.
3. In the event of a service disruption in the company's call center, the disaster recovery plan can be quickly deployed through the central call controller, so that customer contact center calls can make telephones in branch offices ring as if branch employees were call center agents. This is also an important feature in seasonal businesses where a distributed model can reduce the cost of engineering and staffing for peak load periods.
4. In the event of a regional or central call controller failure, local 3Com call controllers continue to operate normally. Unresponsive call requests from the central call controller can be re-routed to redundant systems or over the PSTN.

Voice boundary routing ensures that local calls are processed locally wherever possible or economical, and therefore provides a high degree of WAN bandwidth efficiency. Network latency can be minimized since calls are not usually directed through the central call controller.

3Com voice boundary routing is extremely adaptive to modifications of branch configurations, since changes are captured and available at both the regional office and branch office virtually at the same time, allowing globally optimized routing decisions.

Summary

Branch networking is a challenging domain of many companies in finance, retail, and healthcare industries and agencies in education and government organizations. The large number of end points, the carrier's challenge in delivering inexpensive “first mile” network access, security issues, transition management, and the cost of supporting such a large, distributed operation seems daunting to most IT executives.

The business impact (revenue, service, support, and business control) of effective voice communications can grow with productivity enhancing features and applications. Convergence applications strenthen the backup, resiliency, control, and feature-rich commonality that are essential for truly effective remote working.

The 3Com voice boundary routing system provides the framework for balanced, effective service with reliable features and low cost to meet the most demanding branch office challenges. Network failures have automatic failovers. System outages point end users to higher controllers in the hierarchy, assuring business continuity. Features work uniformly across the company to facilitate employee productivity and positive business impact. Clearly, as the legacy of PBX and key system deployments exceed their useful life, the architecture, the applications, and the time for IP telephony in the branch office is now.

About the Author

Pat Rudolph currently oversees a team of 3Com network consultants following 16 years of experience in the networking and telecommunications industries and extensive experience in successfully creating large networks for Fortune 500 companies and several of the world's largest telecommunications service providers. His career with 3Com began as an ATM/WAN specialist in the company's Global Design Center where he was responsible for Fortune 500 network configurations. A frequent lecturer at telecom and networking conferences, Rudolph's professional career includes working in the former Soviet Union as part of a U.S.-Soviet joint venture tasked with installing networks for a national bank.

Rudolph earned a bachelor's degree from Hendrix College in Conway, Arkansas, and received a master's degree in business administration from Northern Illinois University in DeKalb, Illinois.

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