Strategy Paper

Five Steps to Successfully Implementing VoIP in Your Contact Center (and Everything Else You Wanted to Know but Were Afraid to Ask)

Executive Summary

You've been thinking long and hard about whether or not it is time for your contact center to deploy Voice over Internet Protocol (VoIP), either on a small scale or across the board. You know that it can provide your organization with a number of benefits now, while setting the stage for the future deployment of systems and applications that can offer new and innovative functionality. But, you're concerned. You know your traditional telephony system inside and out and you know its capabilities and limitations; you don't have this same comfort level with VoIP. You've heard horror stories about other companies implementing VoIP and having difficulties getting it up and running. Worse yet, you know of a situation where a VoIP deployment gone awry resulted in service interruptions, leading to decreased productivity and lost business.

The truth is that the majority of VoIP deployments that fail do so because of implementation errors that result from poor planning. In fact, a recent Gartner study indicated that 70 percent of businesses deploying VoIP fail to do a predeployment baseline study, then find themselves dealing with call quality issues[1]. By doing your homework and following five simple steps, you can avoid potential pitfalls and successfully deploy VoIP in your contact center – either all at once or incrementally, as dictated by your budget and needs.

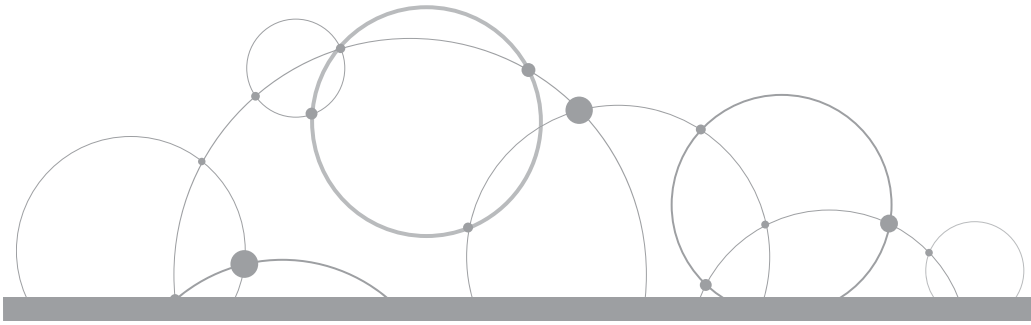
Migrations that Make Sense

According to a recent study conducted by industry analyst firm Yankee Group, VoIP adoption rates in contact centers are expected to grow from 17 percent in 2005, to greater than 47 percent by the end of 2007. In addition, the VoIP penetration rate of agent seats is anticipated to rise from 16 percent in 2005 to more than 60 percent in the next three years[2].

VoIP is being strongly embraced today largely because of the maturity and interoperability of Session Initiation Protocol (SIP), a standard protocol that initiates, modifies and terminates interactive communications sessions. One of the key advantages of SIP is that it allows for the exchange of information in a standard format without requiring specialized equipment such as telephony cards and gateways. Because it promotes interoperability between equipment and applications from various vendors, SIP gives companies more choices, enabling them to eliminate vendor lock-ins.

[1] *Enterprise Networking Planet. "Network Physics Intros VoIP Quality Monitors" December 28, 2006 http://www.enterpriseitplanet.com/networking/news/article. php/3651191*

[2] *Migration Costs, Vendor Loyalty and Need for "Agents Anywhere" Define Contact Center VoIP Adoption Plans, Yankee Group, June 2006*



The maturation of SIP is also allowing companies to easily and gradually migrate to VoIP. While it is entirely possible to immediately cutover to VoIP, slow, controlled migrations are generally recommended for existing contact centers to minimize errors and ensure that customer service continues uninterrupted. Companies that use a hybrid time-division multiplexing (TDM)/VoIP system during this transition period can begin to realize the benefits of VoIP while maximizing their previous investments in legacy technology. They can also select the best systems and protocols to meet the needs of their evolving networks.

However, when a company is opening a new contact center, or is adding another contact center site, immediate deployment of a pure VoIP environment may be the most cost-effective and efficient long-term option, providing an avenue for easy scalability or for adding incremental functionality. VoIP requires only one network – a data network; but significant network capacity is needed to accommodate voice packets, which consume bandwidth on a more constant basis than other network traffic. If a company's communication plan includes a future VoIP migration, rather than deploy separate voice and data networks in a new facility, it may make better sense to build an IP network that can accommodate VoIP, as well as other future high-bandwidth technologies.

What is VoIP?

Simply put, VoIP is the ability to transmit voice over data or Internet Protocol (IP) networks. It works in a manner similar to sending email, but instead of only transmitting packets of data, VoIP can send and receive both voice and data packets.

VoIP enables the simultaneous transmission of voice, data, and multimedia over one "pipe," without requiring special wiring associated with more traditional telephony solutions. It allows companies to share bandwidth between applications and to use network resources in an efficient manner.

Five Keys to a Successful Deployment

Migrating from traditional telephony to VoIP doesn't have to be an arduous process, but it does take some up front planning and consideration. The fact is, vast numbers of VoIP deployments fail because of poor planning that results in implementation errors. By following these five steps, companies can be sure to realize the benefits of VoIP as soon as implementation is complete and their new system is up and running:

1. ***Complete a Network Assessment***

   Before beginning the migration from traditional telephony to VoIP, companies should allocate time and budget, as well as work with their vendors to complete network assessments. This simple step could save companies a lot of money and aggravation by helping them fully understand what type of infrastructure investments are needed to ensure a successful transition and cutover.

   Network assessments are conducted by simulating the load of the maximum number of SIP calls on the company's network and monitoring delay, packet loss, and other characteristics of the network. The assessment should examine the amount of bandwidth a particular network can accommodate in aggregate and by segregating VoIP traffic. This is particularly important because VoIP requires voice and data packets to traverse the same network and compete for bandwidth. As a result, delays will be heard over phone lines if there is a lack of bandwidth. Free, easy to access tools, such as those found at www.bandcalc.com, can be used to help companies determine the optimal amount of bandwidth necessary based on their network or contact center requirements.

   In addition to confirming adequate bandwidth availability, the network assessment should evaluate Quality of Service (QoS) to ensure that voice packets are taking precedence over data packets. QoS allows companies to maintain the integrity of voice transmissions – even if large files are sent. Lack of QoS could result in a number of items that could denigrate the service, including: latency, which lengthens the amount of time it takes for spoken word to travel between speaker and listener; jitter, which causes conversations to be "broken"; and voice packet loss, which results in dropped conversations.

   By conducting thorough network assessments, companies can identify and correct network issues before they negatively impact their VoIP implementations.

Case Study: Telerex

Telerx is a Top 50 customer care service bureau that provides outsourced contact center services. The company implemented a hybrid contact center solution so that it could deploy TDM and IP-based agents on a single platform while maintaining its legacy investments in a PSTN infrastructure, taking advantage of VoIP capabilities while building a solid foundation for future applications and processes.

"We worked with Aspect to validate our network infrastructures. We brought in quality of- service monitoring tools to incorporate on our networks. We isolated voice networks we had in place and brought in some additional high-end routers. We also looked at our network layout to make sure we had multiple carriers going in so we had dual pathing. If the voice couldn't get out one of the routes, it could deliver itself on another route. We also took advantage of data circuits that we already had, bonding them together to increase the bandwidth out of the contact center. So from a data network standpoint, we had a really solid infrastructure in place, and we had very good voice quality as well."

Carolyn Chamoun, Senior Vice President of Technology Solutions, Telerx

2. ***Provide Agents with the Right Phones***

Once companies have selected their service providers and addressed the basic network issues, they need to determine what types of phones they will use and how they will ensure the security of all transactions and conversations transmitted over IP. Certain phones offer simple "plug & play" capabilities that eliminate configuration headaches, and enable companies to connect their phones to their networks and immediately begin to interact with customers.

There are two different types of phones that can be used in conjunction with VoIP. A "soft" phone is a software program that is installed on a PC and allows users to make telephone calls using their computers. The computer must include speakers or a headset so that the user can hear the information that is being transmitted. It must also have a microphone to enable the user to speak to the other party. While the idea of using a computer as a phone, and for email, IM, and Internet access may be appealing, it poses a similar problem as network congestion. If the PC is too busy handling other programs, it may not provide good audio quality.

"Hard" phones, also known as IP phones, are separate from computers. Unlike soft phones, hard phones contain the necessary hardware and software to handle VoIP calls. They sit on desks and look and act like traditional phones, however rather than plug into TDM networks, they plug into Ethernet ports on routers. There are a number of hard phones on the market today, ranging from basic to complex, and many come with a plethora of features, as well as corresponding price tags. Some of these phones incorporate standard protocols, such as SIP, that allow for easy interoperability with other applications and systems, while others are proprietary.

3. ***Implement Proper Security Measures***

Regardless of the type of phones that companies select, it is imperative that they ensure the appropriate authentication, security and encryption. In many ways, VoIP is actually more secure than typical PSTN calls because authentication mechanisms are actually built into SIP and other VoIP protocols. Contact centers can employ existing technologies, such as virtual private networks (VPNs) to provide agents with data access via secure VoIP connections. Implementing SIP over a VPN allows centers to take advantage of secure encrypted audio channels. And, deploying session border controllers over the signaling and the media streams involved in call handling enables contact centers to meet any additional encryption needs that have been identified.

4. ***Prepare for Redundancy to Prevent Data Connection Failure***

When deploying any technology, it is wise for companies to consider, evaluate, and plan for potential failures. Unlike traditional telephony, VoIP offers point-to-point connectivity, linking two parties using only a data network, and making the data network the most critical point of failure. Like VoIP, traditional telephony uses a network, but it's a voice network that requires additional equipment in order to transmit

conversations. This additional equipment means that the traditional telephony network has many more potential points of failure than a data network, making it difficult at times for companies to determine the origin of their failure and resulting in prolonged outages.

The use of IP phones can help companies shorten the down time associated with remote locations experiencing data connection failure. When this type of failure occurs with VoIP, as long as the company's network is still functioning, the remote office connections can be moved to a different server, and the location can be up-and-running in a short period of time. The external availability of this feature with VoIP makes it easier to access and use than with traditional PBXs, where the feature is built into the PBX. The external availability of this feature also means that a company can easily and cost-effectively achieve redundancy by simply buying another computer or gateway.

Whether contact centers are using traditional telephony, VoIP, or a combination of the two, they can safeguard their operations against failures by providing redundancy whenever possible. They should use multiple carriers, have spare (N+1) redundancy for critical components, and employ backup network power supplies to keep the data network viable in case of power failures.

5. ***Connect to the IP Network; Get Started***

Companies that are considering deploying VoIP in their contact center may also be thinking about implementing an Internet Protocol Private Branch Exchange (IP PBX). An IP PBX enables companies to move to IP at their own pace – either immediately or via a gradual transition. A company can either select a proprietary IP PBX – one that is built and maintained by a vendor; or an open source IP PBX.

Open source IP PBX, which is based on the open source software model, can be downloaded for free, and the source code is available to all users for modification and redistribution. Open source IP PBXs create a unique way for businesses to purchase and deploy software, and free contact centers to realize their full value by interoperating unified or multichannel contact center applications with any underlying transport, rather than focusing on the PBX infrastructure. Cost savings, coupled with a rich and robust feature set and customization capabilities make open source IP PBX a viable option for organizations, especially for contact centers, which handle customer-facing interactions.

Aspect – Helping You Stay Ahead of Your Competition

Aspect is a valuable resource for companies that are considering, planning, or implementing VoIP in their contact center. As a leader in the contact center space for more than 30 years, we have a significant amount of experience with both traditional telephony and VoIP architectures, giving us the necessary knowledge to help our customers seamlessly bridge earlier technology investments with new technology and capabilities.

We fully stand behind the use of open standards – a key component of VoIP – in the contact center for a number of reasons: they are vendor neutral and therefore provide customers with more options, they make contact centers more productive, and they give us a solid platform upon which we can develop new technologies that easily interoperate with other contact center solutions. Because we believe that SIP and other open standards are good for our customers, we are committed to continuing to invest in open standards, and we are working hard to incorporate them into the majority of our solutions.

For example, all Aspect Signature and Unified IP products have SIP capabilities. The neutrality provided by SIP makes it even easier for our customers to interoperate all of their Aspect solutions with systems and applications from any other SIP-enabled product. And, the successful interoperability of our portfolio of contact center products with SIP Trunking solutions means that our customers can use SIP Trunking to route calls directly to their contact center products. This eliminates the need for costly load balancing products. As a result, customers are able to reduce international and interstate call costs, increase call quality, and reduce points of failure.

However, SIP is not the only standard that the company is embracing. As an enthusiastic proponent for standards-based technology, we are also incorporating other established standards, such as VoiceXML, Web Services, and Service-Oriented Architecture (SOA), into our products. And, our subject matter experts are contributing to the development of some emerging standards, such as State Chart XML (SCXML) and VoiceXML 3 (VXML3), which we believe will greatly benefit our customers. Aspect is coupling open

standards with our belief in being infrastructure-neutral so that we can focus our attention on what we do best – develop the applications that ride on the transport. Through our partnership with Digium, we are able to offer Asterisk Business Edition and provide customers with the support needed to successfully implement and maintain an open source IP PBX. Companies that take advantage of this technology are able to enjoy all of the features and functionality available in a proprietary IP PBX at one-half to one-third of the price, "test drive" an IP PBX for little or no cost, and easily integrate an IP PBX with their existing telephony infrastructure, as well as standard hardware and applications from a variety of vendors.

Finally, the Aspect Unified Command and Control™ offering helps customers tie all their traditional and VoIP-enabled Aspect products together to consolidate administration, routing, reporting and workflow rules within a single site or across the entire enterprise. Through the use of centrally created and distributed routing rules, contact centers can use this solution to ensure consistent management regardless of which contact center a customer is connected.

While some people think that VoIP is less secure than traditional telephony, the truth is, there is no inherent security difference between the two. In fact, by using certain tools such as virtual private network (VPN) and audio encryption, which cannot be utilized in conjunction with traditional telephony, VoIP can actually be made more secure than the alternative. At Aspect, we are committed to keeping our customers safe from unwanted intrusions. We can help a customer deploy SIP over a VPN so that they can take advantage of encrypted audio channels. We also work with our customers to deploy session border controllers to exert control over the signaling and the media streams involved in call handling to meet their additional encryption needs.

Our award-winning products and services combined with our three decades of industry-leading experience allow us to help our customers make a seamless transition from traditional telephony environments to VoIP. We have extensive product offerings that support the VoIP needs of most companies – whether they are in need of a hybrid solution or a pure IP option. And, we help companies make the migration at the time and pace that best works for them.

The Last Word

Many contact centers know their proprietary traditional telephony systems inside and out, and therefore have a high level of comfort that they may not have with VoIP. However, for companies that haven't already started investigating the possibility of incorporating VoIP into their communications plan, it may be time to do so.

VoIP is an effective technology that provides contact centers with a number of new and exciting opportunities. It enables increased interoperability and flexibility that is not available with traditional telephone systems. As the industry develops new technologies and applications, many of them will be SIP-enabled and designed specifically for use in a VoIP environment. These applications will deliver new capabilities that will help contact centers provide their customers with better service. As a result, companies that become proficient with VoIP may have a competitive advantage in the future.

Companies don't have to implement VoIP all at once. Aspect helps contact centers deploy VoIP at the pace that makes the most sense for their organizations and their budgets. That means that there is no need to entirely replace existing systems in order to begin realizing the benefits of VoIP – contact centers can continue to use what they have while they incrementally add VoIP capabilities.

Deciding whether or not to migrate to VoIP is a complex decision that is unique to each organization. VoIP has the potential to open up a whole new world of possibilities, but following pre-determined procedures and selecting the right partners can make all the difference.

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About Aspect
Aspect provides software and consulting services that turn the potential of unified communications into real business results across the enterprise and in the contact center. Applying 35 years of insight and experience, Aspect helps more than two-thirds of the FORTUNE Global 100, as well as small and medium enterprises, power their business processes with communications. For more information, **visit www.aspect.com.**