IP Telephony from A to Z

Chapters One to Eleven

ebook

The Complete IP Telephony eBook

ShoreTel™

TABLE OF CONTENTS

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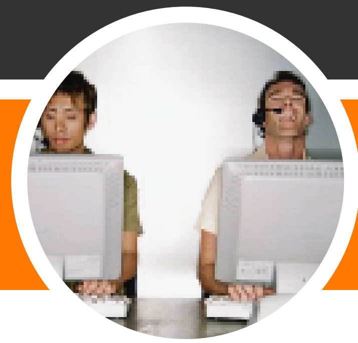

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TABLE OF CONTENTS continued

The Benefits of IP Telephony

Chapter 1

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TABLE OF CONTENTS

This chapter highlights the benefits of IP telephony and discusses the costs in detail so that you can make decisions about your deployment.

The Benefits of IP Telephony

Chapter 1

One of the key drivers of converging voice and data networks is cost savings. Money can be saved, with the right IP telephony solution, in almost all areas—from deployment and management time and costs to ongoing toll and lease charges. IP telephony can also help your organization gain a competitive advantage, boost employee productivity, and enhance customer service. However, there are important considerations to analyze when deciding on a solution, including: equipment costs, which include the cost of the infrastructure equipment (voice switches) and handsets (analog or IP telephones or a mix of both); operational startup costs, including the time and resources it takes to plan, install and troubleshoot the solution once it is deployed; and finally, maintenance costs, which includes the cost of labor to maintain the equipment plus whatever costs must be paid to the solution vendor for maintenance and upgrades. This chapter will highlight the benefits of IP telephony and go over the costs in details so that you can make decisions about your deployment.

The Savings

When you consider what most businesses pay for long-distance, you wouldn't see a huge need to move to IP telephony, necessarily. Large corporations can be paying pennies per minute for long-distance within the U.S. So while companies beyond North America may realize significant savings on toll charges, these savings are not usually enough to convince a North American company to switch to IP telephony.

Savings for most enterprise networks come from consolidating the voice and data network and using fewer circuits from the public switched telephone network (PSTN). In addition to circuit cost savings, as mentioned earlier, an IP infrastructure requires less time for moves, adds and changes (MACs) and often eliminates the need to hire an outside vendor or service provider to handle them. Moving an IP telephone station temporarily or permanently or adding a new user usually simply entails carrying out a quick and simple GUI-based command. With traditional PBX systems, moving an employee can cost hundreds of dollars in labor. In other words, with IP telephony, each user has their own IP phone profile and the network doesn't care where anybody is located at any particular time, so MACs are simply a matter of conducting a few commands and can often be easily handled by the user.

With IP telephony, management savings are usually immediate since the information technology team can support the voice network as well as the data network because they're now one in the same. There is no longer a need to have two teams of technical professionals to handle each entity, which adds up to tremendous savings. Further savings are seen right away when an enterprise needs to make a change, such as re-locating an office temporarily in the case of construction. The IT staff simply makes the changes from anywhere on the network (or remotely if need be) and a new temporary office is up and running without outside callers ever being the wiser.

Finally, infrastructure tools like physical ports are no longer needed for IP telephony because physical circuit-switched ports aren't necessary. An IP connected voice mail server is all that's needed.

All of these cost savings are tremendously appealing characteristics of IP telephony. When you add to them the features that are available for employees, call centers and receptionists, it quickly becomes obvious that IP telephony is going to continue winning converts.

The Added Capabilities

Call centers in many enterprises today are extremely expensive because dedicated buildings are often built to accommodate the many staff members. When a company needs to add additional call center staffers, traditional PBX-based phone systems must also grow in blocks because ports are bought in groups, rather than scaling seamlessly with each new hire. These factors make call centers very expensive to maintain and scale. However, with an IP telephony solution, call centers can grow one phone at a time and call centers can span several buildings across many states. There is no longer a need for one huge building to house all of the call center agents. In addition, enterprises are able to leverage expertise across entire organizations, rather than hoping to find a highly skilled team in one location to answer incoming inquiries. With an IP telephony solution, a user can sign in from wherever they are (even at home) and is instantly online and available as part of the call center team.

IP Telephony Savings

- Toll charges – least cost routing avoids toll charges.
- Management costs
  - System management labor – time and money saved.
  - Users' personal profile changes – handled by users, not IT staff.
  - MACs – quick and easy to handle from anywhere on the network.
- Physical circuit-switched ports no longer required.
- Fewer circuits from the PSTN needed.

Another customer service feature available in IP telephony solutions is the hunt group. This feature makes certain that all calls are answered by a live person rather than voice mail, which can be frustrating for callers. With various hunt groups enabled, a call into an organization rings extensions in a specified sequence or rings multiple extensions at once (depending on the company's preference), ensuring callers reach the person they need without navigating through menus or being forced to wait in a queue.

Remote sites are also easy to bring online. With traditional PBX systems, adding a remote site often requires adding a PBX extender, which can cost almost $1,000 per user for the equipment alone. With IP telephony, again, a user can log in from anywhere and have all the same capabilities as if they were working at headquarters or within the call center building. With IP telephony, to the outside world, it can seem as though you have call center locations scattered around the globe to be available 24/7, when really you are simply utilizing IP telephony features such as time-of-day routing and call forwarding to make sure calls are answered quickly by a live human being; these people can be working out of geographically-dispersed branch offices, at remote locations, or even at home. Callers always reach a qualified customer service representative, regardless of what time it is. You are also able to manage peak calling times by having the ability to add other employees, regardless of their location, to the call center to help meet the overflow demand.

With IP telephony, users can also easily re-route their calls so that they are reached wherever they will be working—they can make these changes themselves, without asking for IT assistance. This "find me" feature also enhances customer service as well as productivity by ensuring a caller reaches the right person, regardless of where he or she might be working. An employee can even program his or her extension to ring based on status—ring through when he or she is in the office, forward to a cell phone when there is no answer, or forward to a colleague when the line is busy.

The Customer Service Advantage

IP telephony offers organizations tremendous customer service value-add. First of all, IP telephony systems provide thorough information right at the time a call comes in by popping data onto an agent's screen. This information can include the most basic of information, such as caller ID information. By integrating specific business applications with the IP telephony system, more in-depth information can populate the screen, including the caller's buying patterns, address, current account status, and more. Many IP telephony systems also provide for operators significant background information on the current caller's experience, such as where the call originated, how many times he or she has been transferred, and whether or not the right person is available to take the call. When the person is again transferred, IP telephony systems eliminate the chance of a caller being asked the same question twice (which is frustrating for callers, and frankly, poor customer service) because the most current information, including notes taken during the present call, populates the next person's screen.

IP telephony systems also allow organizations to implement skills-based routing, whereby calls are routed via an automatic attendant (attendant prompts the caller to choose from a selection) to the most appropriate agent based on criteria like language, experience, technical expertise, and other details. Advanced features that most service providers charge for are also available "free" with IP telephony, including three-way calling and a built-in conference call bridge. This can further aid in customer service when more resources are required to fulfill a customer request or inquiry, and it also allows conference call access by international parties, a feature most expensive conference call services do not provide.

Finally, IP telephony enables self-service options. For instance, when a caller simply wants to find out information about their own account, interactive voice response (IVR) within IP telephony systems enable callers to securely access that information by providing specific information. This eliminates the need for a call center agent to take time to answer a call, and it also eliminates the frustration that can occur if a caller is put in queue on hold for the next available agent to find out information that is readily available.

The Productivity Boosts

IP telephony productivity programs can often transform a company's desktop application, such as Microsoft Outlook, into a multi-media communications center for integrated messaging, providing such features as directory dialing, contact screen pop, caller ID, call waiting, and calendar integration. Employees have more control over both voice and e-mail messages, in one centralized system, and can forward voice mails to colleagues for improved collaboration and customer issue resolution. IP telephony system reports also keep a history of calls made and received, which is helpful in meeting various compliance regulations. Sophisticated features include on-the-fly document sharing and dial-by-name capabilities. Workers are dialing one another, conferencing, transferring calls between locations, and changing their voice mail preferences all with the click of a mouse. There is no longer a need to call the help desk to make such changes. The bottom line is that employees spend less time navigating complex telephone systems and more time performing critical, revenue-producing tasks.

Soft phones further free people from their desks, delivering telephony capabilities to any PC. With calls directed to a laptop and a headset plugged into the USB port, employees can work from anywhere using their computer and its built-in microphone. Employees who travel a lot appreciate the power and simplicity of a soft phone and customers appreciate not having to dial different numbers to reach someone who is traveling.

The Growth Factor

IP telephony systems allow for quick and easy scalability to accommodate new locations or growth within existing locations, as well as the ability to add people one at a time as needed, rather than investing in equipment that will handle more than an organization needs at the time. Scalability benefits also work downward: when an organization reduces its staff count, it is simply a matter of removing those users' profiles from the IP telephony solution. Companies are no longer tied to long leases for equipment that remains underutilized.

The Management Ease

The best IP telephony systems have intuitive browser-based management interfaces, allowing companies to manage the entire system—from switches to voice mail, automated attendant, and desktop applications—from anywhere on the network. The best management interfaces make adding a new user a snap and automatically update every switch and directory feature, including the dial-by-name and number attendant and online directory. System updates are also quick and easy, taking an hour or two at the most when vendors release new code.

Some Features Available in IP Telephony Solutions (not comprehensive)

- Business application integration (for instance, tying IP telephony to CRM database)
- Calendar integration
- Call waiting
- Caller ID
- Click-of-a-mouse simplicity—employees make or transfer calls right on their computer
- Conference call capabilities with on-screen document sharing
- Contact screen pop and comprehensive information about each caller
- Desktop application (i.e., Microsoft Outlook) integration
- Dial-by-name capability
- Features easy to navigate for users
- Four or five-digit dialing to anyone, regardless of location
- Mobility—users can work from anywhere
- Three-way calling

In addition to managing the system itself, managing users and MACs is simplified tremendously. Employees can make most of the changes to their profiles without bothering the information technology professionals, and for changes that do require further expertise, IP telephony systems make it simple. There is no longer a need to spend time and money on having a service provider come in. These costs alone can save an organization thousands of dollars a month.

Nemertes Research, which is one of the few research firms that focuses specifically on IP telephony, suggests that you start the process by carefully assessing the size of your rollout. This consideration is not dependent on company revenue but how many stations you need the solution to support. You will analyze solutions for the time it takes to install these stations, and estimate your growth and how your particular solution's scalability will affect the deployment.

The Costs

Nemertes Research interviewed IT professionals from a wide variety of companies and analyzed four leading vendors in specific areas, including total hardware costs, network upgrades, IP handsets, management tools, and conferencing/collaborative applications. From these in-depth interview came a comprehensive report entitled, "Convergence & Next-Generation WAN Technologies" (February 2006). This section will look at some of the costs involved in an IP telephony solution deployment, as well as provide high-level results of the interviews conducted.

Capital costs are obviously the first line of investment for an IP telephony implementation. This is determined based on how many locations and users you have and a knowledgeable and experienced integrator can help you with this. How many switches and telephones will you need? If you need to, make sure you can phase the solution in over time and use your existing analog lines for some amount of time before switching to IP handsets. Nemertes Research calculated the cost of capital per user, by vendor solution (see Figure 1.1 below).

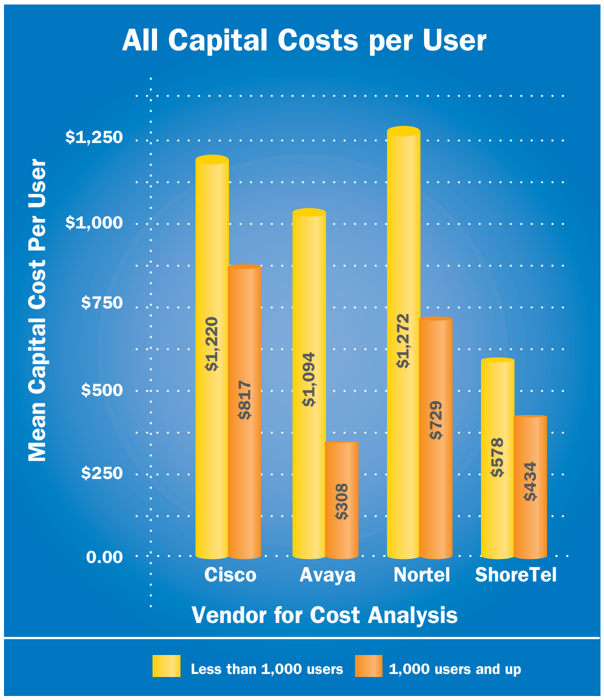

Figure 1.1 | All Capital Costs per User, by Vendor
Source: Nemertes Research

The planning and design phase of any rollout is one of the most important. Consider your team and think of how you will divide up responsibilities. Also, consider whether you will need to add to or reduce your team size. For the implementation, decide on a few team leaders who will commit to making themselves available in the off-hours when necessary until the deployment is complete. The best solutions are easy to implement rather quickly and seamlessly, but you will still want some key people available throughout the deployment.

Installation is the time it takes to physically deploy and configure the solution—it does not include training. Again, consider carefully who is available to help with the installation, taking all things into consideration such as work schedule flexibility, knowledge and expertise, and the ability to work under pressure. Consider your business and determine the best time to deploy the solution and when it will be easiest to switch over to the IP telephony solution.

Next up is troubleshooting—the time it takes to make changes immediately after the deployment until it works properly. Who is going to be available throughout the deployment right up until the minute you determine that everything is working perfectly? Consider the first few days and how you'll staff the help desk around the clock with people who are substantially knowledgeable about the infrastructure, the configuration, and the features of the handsets.

Next up are the costs for staffing to support the new implementation on a regular basis. How easy is it for your current staff to support the new IP telephony system? Generally, it is very easy for existing network staff to support IP telephony solutions because they work on the data infrastructure, which is what they already know well.

Management is the next cost consideration. What are your staff members doing each day to support the solution? Can things be handled in-house, without wasting time and money on an outside vendor or service provider to handle personnel MACs? According to Nemertes Research, MACs become very easy with IP telephony: Research participants estimate the time involved for an IP MAC at a mere 10 minutes or less, compared to the 30 to 90 minutes required for a TDM MAC. This means that total cost savings, depending on the average number of MACs at a given organization, can be significant.

Nemertes Research ultimately calculated the total cost of ownership (TCO) for IP telephony solutions from leading vendors (see Figure 1.2 below). These numbers were calculated considering all of the costs listed above. This gives you an overview of costs for each vendor's solution based on the implementation size.

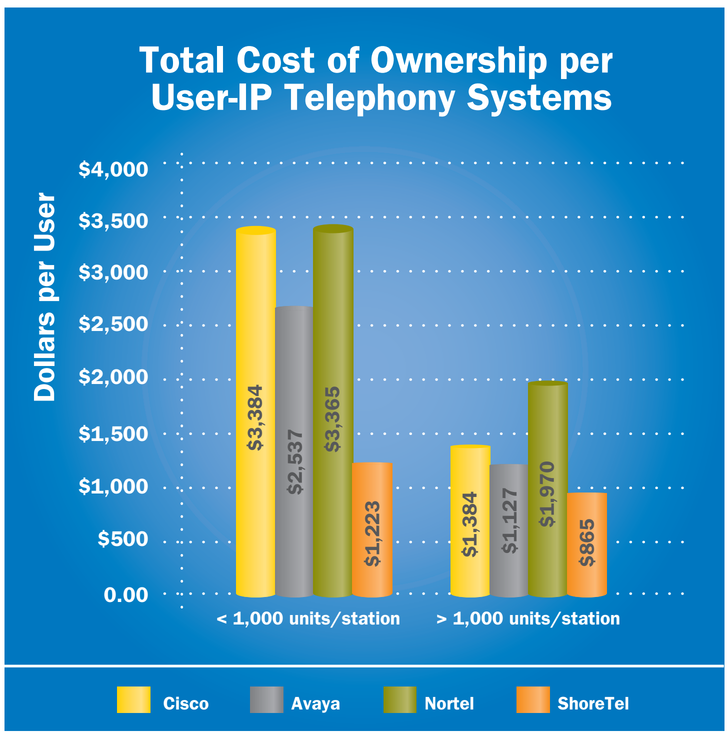

Figure 1.2 | TCO per User
Source: Nemertes Research

Ready to Make the Switch?

IP telephony is the way of the future, according to Nemertes Research, for a number of reasons. First, vendors are no longer investing research and development dollars into legacy TDM equipment. Second, IP telephony has simplified communications for numerous organizations and their positive results have been shouted from rooftops (or at least highlighted in well-respected trade journals). With TDM, there's no interoperability, transferring between offices is not an option, and employees are often on different voice mail systems so forwarding messages is not possible. With IP telephony, companies instantly improve productivity with robust feature sets such as built-in conference call capabilities, four-digit dialing across locations, call center capabilities, and integration with desktop applications. Because of robust features like the ability for an employee to log in from any phone, employees are not tied to a desk.

A Network World special report suggests that organizations should consider transitioning to IP telephony when:

- They are using IP Centrex lines that will support phone and Internet service on the same network. Moving to IP telephony will immediately reduce costs because these lines are so expensive.
- The organization is moving to a new building. Since the wiring does not yet exist, it's simple to create a consolidated data and voice network.
- They are coming to the end of a PBX lease agreement or the current phone system is outdated, obsolete or unsupported by a vendor or service provider.
- The company has offices in different area codes and employees dial a lot of long-distance numbers. The reduction in toll charges will be immediate and significant.

You will also want to consider IP telephony for your organization if:

- Your locations shift in size often
- Locations are added regularly
- You have a relatively small technology staff
- You use a great deal of outsourced telephony services that are beginning to add up
- Many of your employees frequently work remotely

Once you've evaluated your organization carefully, analyzing the costs of your current telephony solution along with your employee productivity and customer service needs, and decided that indeed, IP telephony is the way to go, the next chapter will help you with the vendor evaluation and selection process.

The Decision: Vendor Evaluation and Selection

Chapter 2

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TABLE OF CONTENTS

This chapter provides you with resources to help you evaluate and select IP telephony vendors.

The Decision: Vendor Evaluation and Selection

Chapter 2

You've made the decision to go with IP telephony after careful consideration, but if convergence is new to you personally and to your organization overall, the decision is likely accompanied by worry and concern about making the right choices. Your choice of technology vendor for this transition, as in any decision, is one of the most important. InfoTech, a recognized leader in project consulting and global research in over 90 countries worldwide, thoroughly researched how enterprise decision makers choose their vendor and reported on its findings in a report entitled, "Strategies for IP Telephony Evaluation and Migration" (April 2005). This information will be of great benefit to you since those interviewed by InfoTech have successfully deployed various new technologies, including IP telephony.

Expectations

InfoTech reports that enterprise decision makers generally have three main areas of expectation that help them choose the right vendor. These are areas you'll want to consider as you embark upon the vendor evaluation phase.

1. Convergence experience, expertise and vision
2. Expert, responsive support
3. Customer-focused approach to business

Convergence experience, expertise and vision

Look closely at vendors to determine whether or not they are committed to IP telephony. Have they built their solution as a true IP telephony system, or are they jury-rigging an old PBX-based solution to "look like" an IP telephony solution? Are their solutions built with flexibility, scalability, and longevity in mind? Will you have to completely rip out your old phone system and move to IP telephony in one fell swoop, or has the vendor built its solution with a phased approach in mind for those organizations that need to replace their phone system over time?

Expert, responsive support

When you're working with a vendor during the early stages of consideration, try reaching their technical support team during off-hours. Do you have easy access to technical support representatives and a full range of maintenance and support services? Have they committed to working closely with you during initial deployment as well as future and ongoing projects? While you'll almost certainly have quick and easy access to a sales representative and possibly a pre-sales engineer during the evaluation phase, you need to find out how you'll be treated once you've already deployed your system. Is vendor responsiveness just as good for customers as it is for prospects?

Customer-focused approach to business

This area focuses on the vendor's commitment to your success. Don't let vendors come into the proposal using a hard sell approach. If they do, they aren't demonstrating a commitment to your success but rather a commitment to their own success (meeting their quota). For real proof points, ask to see a list of the company's latest customer installations and ask if you can speak with those customers. If things have gone smoothly, they won't hesitate to let you talk to a customer in the early phase of their deployment. Don't settle for just a list of customers that have been using the vendor's system for years. Call early phase customers and ask them if the vendor is still in close contact with them, calls to proactively find out about the installation, and provides onsite support at a moment's notice during the deployment.

InfoTech found that while many companies vary in why they choose IP telephony, most enterprises have found the most common anticipated benefits as:

1. Lowering total operating costs
2. Enhancing end-user productivity
3. Improving IT organization efficiency
4. Reinforcing market differentiation and brand image

Choices, Choices

In their 2006 report, "Convergence & Next-Generation WAN Technologies," Nemertes Research provides a comprehensive and unbiased look at what organizations are doing specifically in terms of which vendors they choose. It is an independent and impartial report that translates mountains of data into succinct information organizations can use for convergence planning. The firm collected information by conducting in-depth interviews with IT professionals from a wide variety of companies of various sizes spanning many industries. While the industries varied greatly, all of the respondents had a similar interest in IP telephony and were committed to making technology investments that enhance productivity and the bottom line and prepare their organizations for the future. Nemertes Research presents an overview of how the respondents have assessed IP telephony solutions and how they eventually selected a system vendor. Included in the report are recommendations about which vendors to consider, including a complete IP telephony system vendor analysis, how to thoroughly evaluate all of the solutions available, how to plan for convergence, and how to actually conduct the rollout.

Vendors Analyzed

Organizations in the past have had few vendors to choose from. According to Nemertes Research, today there are more than 25 vendors and carriers out there to meet IP telephony needs. The increase in competition means more innovation and better products from a wider selection of companies. Nemertes Research analysts established that the most frequently evaluated IP telephony system vendors today are: Avaya, Cisco, Nortel and ShoreTel. The following section will highlight each of those vendors, but keep in mind that there are at least a dozen more to evaluate, depending on the size and particular needs of your organization.

Avaya

Avaya offers IP telephony solutions with its IP Office and MultiVantage solutions, which include IP telephones, as well as voice switches, media gateways, communication servers, wireless telephones, communication applications, and more. According to Nemertes Research, Avaya's key strengths are its product features, technology, and overall performance, while weaknesses, according to respondents, fall in the areas of customer service, ease of use (installation and troubleshooting), management tools, and VAR expertise.

Cisco

Cisco is a recognized network infrastructure equipment leader and offers IP telephony solutions under its Unified Communications family. Products include switches, telephones, communication applications, and more. Nemertes Research notes that Cisco's overall performance and technology areas have been rising steadily, according to respondents, while product features have left much to be desired. However, many networks are built on Cisco networking equipment and it would be hasty to overlook the company during an IP telephony vendor review.

Nortel

Nortel offers more than IP telephony solutions and has been around since its 1895 founding as Northern Electric and Manufacturing, supplying telecommunications equipment for Canada's telephone system. Nortel was the first networking vendor to provide an end-to-end IP telephony solution certified by the U.S. Defense Department Joint Interoperability Test Command (JITC) in 2004. For 2005, Nortel's top areas, as noted by Nemertes Research, were performance, product features, value, and customer service. Its weaknesses, according to respondents, were in the areas of ease of installation, VAR expertise and management tools.

ShoreTel

ShoreTel offers end-to-end IP telephony solutions including its ShoreGear voice switches and ShorePhone IP telephones, as well as communication applications, call center functionality, and more. ShoreTel scored highest in all categories studied by Nemertes Research. Four specific areas in which the company excels are value, technology, ease of installation and troubleshooting, and performance. The company's areas for improvement included management tools, solution experience and VAR expertise.

Issuing the RFP

If you work with a network integration partner or consultancy, you may want to call on them to help you with the Request for Proposal (RFP). You may also request a sample RFP from any of the vendors you'll be evaluating, but be careful to go through and make sure the one you use is comprehensive and not skewed toward any one vendor. If you decide to write the RFP yourself, here is an outline on how to go about it.

RFP: From Concept to Paper

Assemble your RFP team. Be sure and include an IT representative, a budget specialist, and any senior executives in charge of departments that will use the technology extensively (sales, telemarketing, etc.). This team should be briefed on the IP telephony project and should understand what new capabilities such a solution will offer so that they are informed enough to give you an extensive "wish list" for features that will make them more productive.

1. Select a project leader. This person should be experienced in networking and IP telephony, if possible, and should be able to answer basic technical questions related to the technology, if not the specific vendor solutions.
2. Assess what you need from the IP telephony solution.
   - Evaluate the current situation, including costs, etc.
   - Identify key goals.
   - Review most common product capabilities and decide on the importance of them.
   - Determine if there will be training required.
   - Estimate the cost of the project.
3. Record your requirements, goals, and recommendations in a tentative plan.
4. Present your plan to the appropriate organizational leaders (executive management, financial department, etc.). Get their input before writing the proposal.
5. Write the proposal. A typical proposal contains:

- A summary of the proposal.
- A statement of what you need – the reason you're looking for a new solution. Include every capability the RFP team has mentioned—be sure and get input from executives, managers, and staff level employees so that every need is met. Do not overlook the obvious and assume that every vendor provides one specific capability (you know the saying, "do not assume anything"). Conversely, what is missing from the current telephony solution should also be noted.
- A weighted ranking of all of the capabilities and features should be included (see figure 2.1 for a sample weighted ranking worksheet). Be specific in the features/capabilities list and avoid "buzz" words that each vendor could define differently. If necessary, describe any word that could be misconstrued, such as "availability," which vendors often define differently. Again, don't assume. Include every single capability that you need. The list should be exhaustive. In other words, don't omit "voice mail boxes for every employee" from the list because you assume all vendors provide them.
- A description of how the project will be implemented and evaluated.
- Provide information about your organization and its technology goals.
- Include a project schedule. Indicate when you want the new IP telephony solution in place. Provide details on how you want to implement: in phases, within three months from the date of selection, etc. Be sure to include how you want each phase to be implemented so that you get as much out of your old equipment as possible and extend the life of existing equipment and handsets.
- Provide an approximate budget.
- Conclude the RFP with specific open-ended questions for vendors, such as:
    - » What is your approach to training? Where is training held and how long does it take? Will the price of the solution cover travel time and expenses for your staff to attend if it is offsite?
    - » Is there a guaranteed response time for support calls? How will your system be updated? Is telephone support all that's covered in maintenance fees or are other things covered? Is there an option for hourly support? How many support staffers are on call 24 hours a day? Does the solution contract come with a support guarantee?
    - » What is your history? How long has your company been in business? How many customers do you have? How many new customers have you signed on in the past year? The past six months? Are there any current merger discussions?
    - » What about customer references—to whom can we speak? Beyond happy customers, ask to speak with the most recent customers. A reputable company should be able to give you references from the most recent three-month period.
    - » How are upgrades handled and what are the typical costs involved? Also, ask what the process is for a customer to make suggestions and specifically ask if they can name some features that were a result of suggestions from users.
    - » What kind of "bake-offs" and industry reports mention your company? Ask for references in the form of reputably published reports and articles.

6. Submit the proposal to the vendors you've selected in your long list. Your integration partner or consultant, if you have one, can help you with this process, or simply e-mail or fax it to your vendor list.

Seeing is Believing

The next step, after issuing the RFP, is to closely review the proposals from each vendor. It will be helpful to use a weighted ranking system to score each vendor based on your long list of requirements. First, rank each requirement based on the vendor's answer to your checklist items. See Figure 2.1 for a sample worksheet.

	Step 1	In Theory		
Ranking 0=unsatisfactory 10=excellent	This section is where you will simply note whether the vendor offers specific capabilities			
Criteria	**Answer**	**Score**	**Weight**	**Extended**
Cost of solution includes hardware, software and installation	No	0	10	0
Years in business	7	8	10	80
Number of customers	200	9	10	90
Solution can be implemented in phases	Yes	10	8	80
Support 24/7	No	0	9	0
Guarantees 4-hour replacement	Yes	10	8	80
Offers wireless solution	Yes	10	7	70
Features				
Voice mail for unlimited extensions	No	0	10	0
Centralized management of e-mail and voice mail	Yes	10	8	80
Intuitive GUI that simplifies MACs	Yes	10	10	100
Call forwarding	No	0	10	0
Caller ID	Yes	10	10	100
4 or 5-digit dialing across locations	No	0	10	0
Workgroup capabilities	Yes	10	9	90
Hunt capabilities	No	0	9	0
Call center capabilities	Yes	10	10	100
In-depth information about caller "pops" onto screen	Yes	10	8	80
Ability to integrate an IP telephony system with other business apps	No	0	10	0
Least-cost routing functionality	No	0	10	0
Score				950

Figure 2.1 Sample weighted worksheet for vendor evaluation—not an exhaustive list.

Next, ask to see a demo and request a sample set-up to test the solution in your office so you can revise the score based on actual experience. Once you have seen a demo or tested the solution, revise your weighted worksheet to reflect your actual experience. See Figure 2.2 for the revised worksheet and score.

	Step 1	In Theory			Step 2	In Reality	
Ranking 0=unsatisfactory 10=excellent	This section is where you will simply note whether the vendor offers specific capabilities				This section is where you will score each vendor's capabilities after testing		
Criteria	**Answer**	**Score**	**Weight**	**Extended**	**Score**	**Weight**	**Extended**
Cost of solution includes hardware, software and installation	No	0	10	0	0	10	0
Years in business	7	8	10	80	8	10	80
Number of customers	200	9	10	90	9	10	90
Solution can be implemented in phases	Yes	10	8	80	10	8	80
Support 24/7	No	0	9	0	0	9	0
Guarantees 4-hour replacement	Yes	10	8	80	10	8	80
Offers wireless solution	Yes	10	7	70	10	7	70
Features							
Voice mail for unlimited extensions	No	0	10	0	0	10	0
Centralized management of e-mail and voice mail	Yes	10	8	80	3	8	24
Intuitive GUI that simplifies MACs	Yes	10	10	100	4	10	40
Call forwarding	No	0	10	0	0	10	0
Caller ID	Yes	10	10	100	8	10	80
4 or 5-digit dialing across locations	No	0	10	0	0	10	0
Workgroup capabilities	Yes	10	9	90	8	9	72
Hunt capabilities	No	0	9	0	0	9	0
Call center capabilities	Yes	10	10	100	7	10	70
In-depth information about caller "pops" onto screen	Yes	10	8	80	4	8	32
Ability to integrate an IP telephony system with other business apps	No	0	10	0	0	10	0
Least-cost routing functionality	No	0	10	0	0	10	0
Score				950	Revised Score		718

These scores remain the same.

Figure 2.2 Revised sample weighted worksheet for vendor evaluation, with experiential scores—not an exhaustive list.

Decision Factors

Once you've collected all of the information and carefully evaluated your chosen vendors, including the four leaders, think carefully about your organization's priorities in general, and carefully consider the following qualities so you can clearly articulate your requirements in these areas as you approach your final decision. These are areas which InfoTech has determined enterprises consistently rank as top priorities.

High system reliability/availability

Do the vendor's products include redundant components in the case of a failure? Are there ways to re-route calls around a failed switch, for instance? Is there a threshold past which the system's performance will degrade? Ask for specific examples of each vendor's system maintaining availability under the harshest circumstances. Ask customer references specifically about how reliable the system is.

Equivalent voice quality to TDM

You don't want your own customers to call your organization and know right away that you're using something of lesser quality than a TDM system. Ask the vendor if it's possible for you to go to a customer site and listen to actual phone calls to evaluate the voice quality yourself. Or ask customer references specifically if anybody knows they are on an IP telephony system or if it is assumed that it is a traditional system. Customers are usually willing to share the downside of the solutions they've chosen, as well as the upside.

Easy scalability

Make sure that the vendor you choose knows exactly how you will need to scale the system for your specific needs. For instance, if your organization often grows and shrinks during different times of the year or in some other cyclical manner, ask how new users would be added to support your growth needs. Will new hardware need to be added and removed each time you grow and shrink? Or will the system support your needs up to a certain point, regardless of how many times you change size?

Multi-vendor interoperability

Some vendors are known for requiring a full infrastructure overhaul to accommodate the new IP telephony system. Be certain that you can use your existing network equipment with the new solution, and make sure that when you add new gear, you can do so without needing to consider the IP telephony system. IP telephony is only beneficial if it's truly part of the network and it doesn't bring you new headaches or worries further down the line.

Full suite of communications features & business-enabling applications

Cost savings, as discussed in chapter one, are not simply a result of toll charge avoidance. Most cost savings come from the additional features that you get with an IP telephony system. Will the system provide value-added services like call history logging, conference call capabilities, document sharing, follow-me features, etc.? Compare the checklist of capabilities of each system. This is not to say you should simply compare how many features, but rather decide on which ones are most important to your organization and come up with the vendors that meet the majority of your requirements. A simple ranking system for each system offering should work well (see Figures 2.1 and 2.2 for an example using a 1-10 rating system).

Ease of implementation/management/maintenance

IP telephony systems should make life easier for the IT team, not more difficult. Because the new system works on the existing network, everything is managed similarly. If management of the

IP telephony solution is not straightforward and intuitive, how long will it take your team to ramp up to the point that the system will be supported adequately? It's imperative that changes be made quickly and easily so that the addition of a new system doesn't add burden to busy IT personnel. Some of the most important factors of convergence are how it simplifies life and how it saves organizations in terms of management time and money. Does your staff need to train with the vendor every month, and can you afford their time out of the office? How difficult is it to train users on features of the system, and will they be calling for help more often than usual because of the IP telephony implementation? In reality, users should be calling your help desk less frequently with a new IP telephony system. Even employee moves, adds, and changes (MACs) should be simple for either the user or one IT staff member to make within a few minutes. You should also no longer need a service provider to make these alterations for you—this will save you money and time.

Efficient, integrated multi-site networking

You will want to make sure that architecturally, your solution is built around a distributed design. A centralized solution that distributes applications over the network to other sites is inefficient as far as consuming capacity on the WAN. If a vendor is proposing a centralized approach and suggests "simply adding bandwidth" as the way around reliability issues, remember that bandwidth costs are not insignificant and insist on a solution that is designed for optimal bandwidth utilization. Multi-site organizations inherently require a distributed, as opposed to centralized, solution.

Favorable overall cost and payback interval

You can use information from Nemertes Research "Convergence & Next-Generation WAN Technologies" report to compare total cost of ownership data for the four leading vendors (Avaya, Cisco, Nortel, and ShoreTel). Nemertes Research analyzed these vendors in specific areas, including total hardware costs, network upgrades, IP handsets, management tools, and conferencing/collaborative applications, calculating the total cost of ownership (TCO) by vendor solution (see Figure 2.3).

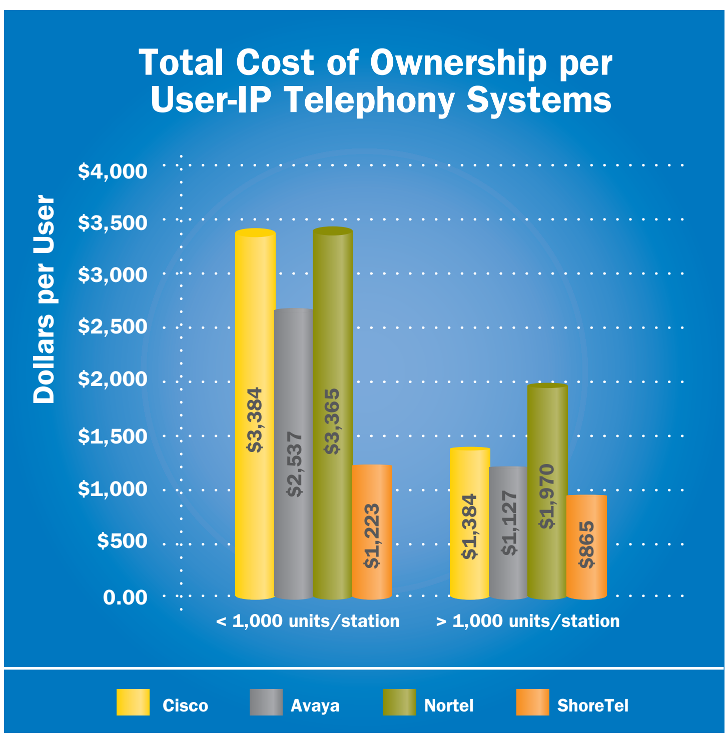


Figure 2.3 Total Cost per User Source: Nemertes Research

Last but Not Least: Ease of Use

Another factor you'll want to consider carefully is ease of use for end users. While you will undoubtedly need to familiarize employees with the system, training should not be cumbersome or lengthy. The IP phones and call control software should be intuitive and easier to use than the analog phones being replaced. Features like on-the-fly conference calling, drag-and-drop call transferring, and the forwarding of voice mail messages via e-mail should be simple for employees, even those who cannot attend training and have to learn the system on their own. You will likely have remote users logging in and using the system, and it will be difficult, if possible at all, to get those people to a training session. In these instances, you'll appreciate a solution that users can easily navigate so they come up to speed and begin capitalizing on features that enhance your company's employee productivity and customer service as soon as possible.

The Bottom Line

The most important things for you to remember during the evaluation process are the main business drivers of convergence. Make sure the vendor you choose is committed to making these perceived benefits a reality for your organization:

- Lowering total operating costs
- Enhancing end-user productivity
- Improving IT organization efficiency
- Reinforcing market differentiation and brand image

The next chapter will explore the IP telephony implementation from beginning to end, starting with research and vendor evaluation and ending with the actual deployment, and will include a helpful timeline for you to use.

Planning: The Implementation Calendar

Chapter 3

ShoreTel™

TABLE OF CONTENTS

This chapter provides you with a high-level timeline for the implementation, from research to actual deployment.

Planning: The Implementation Calendar

Chapter 3

Roadmap to IP Telephony*

12 Months	Read, Learn, and Ask (read about the technology, ask experts)
10 Months	Head for the Internet (scour vendor websites)
9 Months	Call in the Vendors
8 Months	Demonstration and Trial Period
7 Months	Do an Inventory
6 Months	Request Vendor Proposals
5 Months	Choose Vendor
4 Months	Gauge your Network's Readiness
1 Month	Pilot Installation and Testing
0 Days	Go Live

*This schedule can be accelerated to fit needs. For instance, if your organization decides to move locations and the timing is right to implement IP telephony, this schedule can be altered to fit a three month schedule.

The Road to IP Telephony

12 Months to Deployment: Read, Learn, and Ask

The first step is research. The fact that you've reached this chapter in the book indicates you are fairly certain about deploying IP telephony at least sometime in the future, if not the near future. It is best to be making that decision about 12 months before you want to deploy a new phone system, IP telephony or otherwise. At this time, you'll want to get your hands on as much unbiased research and as many reports from reputable consultancies as possible. Read the research with the goal being to decide if IP telephony is right for you. For now, pass up reports that talk about vendors, and get your hands instead on technology articles, technical papers, industry event presentations given by independent technologists or long-term experts, etc.

The following resources can be helpful in your search for IP telephony information.

- CIO Magazine (www.cio.com)
- Network Computing (http://www.networkcomputing.com)
- Network World (www.networkworld.com)
- VoIP Magazine (www.voip-magazine.com)
- ComputerWorld (www.computerworld.com)

Trade magazines and their online counterparts do cover vendors, of course, but you can find unbiased technology primers and overviews. It's also helpful to read customer case studies about deployments to learn about the experiences of those companies that have deployed IP telephony. Read case studies for technology tips first, vendor specifics second.

After you've searched on the Internet and leafed through your stack of technology publications, invest in some time with industry experts and analysts. For lengthy conversations, you may have to invest more than time—research analysts can be hired on a project basis to provide you with valuable information and insights. But be sure to keep a keen ear out for biases because often analysts are paid consultants for specific vendors and because they know the vendor, they'll tend to reference them more often than others. Keep your questions, at this point, in reference to the technology. Learn all you can from these experts about organizations like your own that have deployed IP telephony, what their specific challenges were, and what the results have been.

10 Months to Deployment: Head for the Internet

After you've completed your technology research, visit the web sites of the vendors you've heard about. Read about offerings from the industry leaders Avaya, Cisco, Nortel and ShoreTel. Learn about smaller companies and what the benefits and drawbacks to their systems are. It's recommended to take and keep good notes so that by the time you're looking at the eighth vendor and you've forgotten which solutions do what, you'll have detailed notes to refer back to. This is where you want to establish a long list and then whittle it down to a short list.

You'll read about each solution with your own organization in mind. Jot down questions as you click through vendor web pages. You may get the answer to the question quickly, or it may remain on your list until you eventually meet with the vendor. If your organization has many offices across the United States, for instance, look at solution descriptions with scalability, flexibility, and ease of deployment mentioned early. If your organization rarely changes in size and has a limited number of telephony requirements, look for solutions that offer the basics at a very affordable price point.

Next, create a checklist or table with some common features. For instance, most IP telephony solutions offer standard features like caller ID and three- or four-digit dialing. As you exhaust the common feature list, start adding unique features that matter to your organization. Learn (or try to learn) what differentiates each vendor you're considering. If you save the differentiation for the vendor presentation, you likely will get a skewed answer to the question, "What makes your solution different and superior?" This checklist is just the beginning and you won't do anything with it until the RFP phase.

Read articles about each vendor and mark items off your checklist as you determine what each offers. Start with articles that the vendor links to (usually found under headings like "press coverage," "news coverage," "case studies," "success stories," and "customer solutions" on the website). However, vendors obviously will only highlight their true success stories. Use an Internet search engine to do a little sleuthing yourself—you may find three or four stories about users' unhappiness with a certain vendor. Dig for the dirt. Use all of this information for your checklist and research notes.

9 Months to Deployment: Call in the Vendors

After you've looked at your checklist and decided three or four vendors probably offer the best solutions for your organization, invite each of them to come in and give you an overview of their solutions and a demonstration if possible. You will hear a sales pitch, of course, but you may also hear features you hadn't learned about, or you may hear the names of customer references that have organizational needs like yours. Whenever a sales person drops a customer name, ask for the contact person to speak with after the vendor presentation. If you are told the customer cannot be a reference, (which is understandable—many companies will not speak as a customer reference by policy), ask for a similar customer that you can speak with. If your organization is a bank with 23 branch offices, ask to speak with a similarly sized bank reference. If the vendor is not able to give you even one customer reference right away, take note and be cautious.

8 Months to Deployment: Demonstration and Trial Period

After you've seen each vendor's presentation (and possibly after you've spoken with customer references), inquire about an onsite demonstration and also a trial period. Some vendors, after they've shown you how their system works, are willing to deploy a sample set-up so you can test the solution in your office. Some vendors give you just a few days or a week. Often, as the trial period nears the end, you can easily get an extension just by asking. A reputable vendor does not put a deadline on your decision. They want you to be happy with your choice of their solution; an extended trial period is not a huge cost to them.

Crucial Tasks - Do Not Skip

- Talk to multiple customer references: insist on recent customers as well as success stories.
- Get each vendor to bring an RFP into your office, in person, to discuss details.
- Talk to colleagues at other organizations that have deployed VoIP (beyond vendor references).
- When you're close to choosing vendor, obtain equipment for a trial period.

7 Months to Deployment: Do an Inventory

Assessing your current network is crucial to a successful IP telephony deployment. There are a number of things to keep in mind and questions you'll want to answer about the organization's telephone usage. The following checklist will help ensure you think of everything.

1. **Determine your business requirements.** How will the system be used? How many calls per month (or day) are made out of your office? Are those calls to customers or internal employees? How many offices will you have on a system? Are there remote offices to consider?
2. **Look at your LAN.** What equipment are you using? Do you have an up-to-date network diagram? Is the equipment current or outdated? Are you using Virtual LANs (VLANs) for security or performance issues? VLANs improve voice quality by prioritizing voice traffic.
3. **Assess your WAN.** How much WAN bandwidth do you have between offices? How many home or remote offices do you have and will you need dedicated circuits or will DSL suffice? Consider whether managed IP services are a fit for your organization as an alternative to traditional dedicated circuits.

6 Months to Deployment: Request Vendor Proposals

If you work with a network integration partner or consultancy, you may want to call on them to help you with the Request for Proposal (RFP). You may also request a sample RFP from any of the vendors you'll be evaluating, but make sure the one you use is comprehensive and not skewed toward any one vendor. If you decide to write the RFP yourself, chapter 2 of this book includes an outline on how to go about it.

The next step, after issuing the RFP is to closely review the proposals from each vendor. It will be helpful to use a weighted ranking system to score each vendor based on your long list of requirements. Again, see chapter 2 for ideas about creating these checklists and spreadsheets. After you've narrowed down the vendors to a short list, ask to see a demo and request a sample set-up to test the solution in your office. Most vendors will give you a free trial period so you can get more comfortable with the system.

Once you've collected all of the information and carefully evaluated your short list of vendors, think carefully about your organization's priorities in general and start talking to customers. Be sure you get customer references that have similar networks and similar business requirements to your own organization. Again, ask to speak with recent customers: It's easy to give you a list of happy customers. Ask for a list of the most recent customers signed on—within the last three months, for instance—and call them about their experience.

5 Months to Deployment: Choose Vendor

After you've taken all these steps, created a feature checklist, and determined which vendor best meets your feature/functionality requirements, you should be ready to make the decision. Be sure and ask any remaining questions before you indicate that you are leaning towards that vendor. It is very important to review the vendor's website, including where they post press releases. If there have been any recent upgrades or new product announcements, ask how customers are responding and call customer references again. This will give you the freshest input, and you'll be able to make the most educated decision on the right vendor for you.

4 Months to Deployment: Gauge your Network's Readiness

By testing your data network's ability to successfully support IP telephony traffic and discovering potential performance problems before your system is installed, a network assessment helps you plan, design and implement a successful IP telephony solution. The assessment can be administered by the solutions partner or by the vendor you choose, since both have a wealth of experience with IP telephony that they apply to interpreting the test results. Regardless if you use the solutions partner or the chosen vendor, an expert voice readiness assessment is required prior to installing a new IP telephony system across multiple sites.

In order to achieve toll-quality voice, you need to deploy IP telephony over a properly architected network infrastructure - i.e., it has to provide sufficient throughput and meet latency, jitter and packet loss requirements.

Throughput: How much bandwidth you need depends on the how many simultaneous calls your organization has going on, the voice encoding scheme used in the IP handset or soft phone, and the signaling overhead.

Latency and Jitter: Latency is the time it takes for a caller's voice to be transported (packetized, sent over the network, de-packetized, replayed) to the other individual. Distance and lower-speed circuits can cause delay. Latency that's too high interrupts the natural conversation flow (you may have spoken with someone using VoIP - you think they have stopped talking but they haven't-that's latency). Latency cannot exceed 100 milliseconds one way for toll-quality voice. Acceptable quality voice can go up to 150 milliseconds and participants can still carry on a decent conversation.

Packet Loss: Packet loss results in a metallic sound or conversation dropouts. It's caused by congestion, distance and poor line quality. Because IP telephony is a real-time audio service using Real Time Protocol (RTP) running over User Datagram Protocol (UDP), there's no way to recover lost packets. A mere one or two percent packet drop degrades voice quality.

A thorough assessment uses active application traffic across the LAN and WAN in order to reveal what's going to happen when IP telephony is introduced into the mix. Test agents send a variety of network traffic packets - using different application protocols, packet size, packet spacing and quality of service (QoS) levels. The tests simulate the various types of IP telephony traffic that are likely to occur on a live network. In addition to measuring peer-to-peer traffic, the agents can also generate real-time client transactions against production servers, including communication with IP PBX servers. This comprehensive approach enables the test engineer to pinpoint the source of potential problems and make recommendations for resolution, thus avoiding unwelcome surprises following the implementation.

1 Month to Deployment: Pilot Installation and Testing

If you have an integration partner or the vendor you have selected works with regional resellers and consultants, call and schedule a time to determine your needs list. If your organization or the vendor does not have an integration partner, get an engineer from the vendor in to help you with this list. With this person (or people), look closely at the current design of your network and make a list of any equipment upgrades or new purchases you'll need to make in order to optimize the infrastructure for IP telephony.

Update any existing network diagrams you'll be using. Be sure to label it so you know it is the original (pre-IP telephony). Next, sketch your new network diagram with the gear included. Determine if there is any overlap and if perhaps you don't need as many switches as you thought. If you're not working with an integration partner, you may want to invest some money in having a technology expert take a look at your new proposed network diagram. It's better to make major changes in the planning stage as opposed to after you've taken delivery of your IP telephony equipment. An expert can also make sure you maximize your equipment purchase and may make modifications to your diagram that will save you money in the long run.

After you've come up with your new network diagram, begin deploying the gear onto a test network. This will not only help ensure the new system works optimally, it will help you get accustomed to the new equipment so other deployments (to other locations, for instance) go smoothly. At the beginning, the test network should not affect anybody's workday. During the second phase, transition some non-critical employees or departments to the test network. This will help you further test the system in a real-world scenario and also gets users familiar with it.

0 Days to Deployment: Go Live

After you have played with the system for a few weeks or months and made appropriate configuration changes to adapt to your entire organization, begin rolling out IP telephony company wide. An installation in phases tends to work best, even if the phases are over one week. The larger your enterprise, the longer it will take and the longer you may need between phases.

After the rollout, it's imperative that you schedule end user training. You may handle this by department or location, depending on your organization. Vendor representatives are often available to be onsite to provide expertise and demonstrations during end user training sessions. While your choice of solutions will likely be rich in features, these features should also be intuitive to the end user; therefore training should take just two or three hours, as opposed to all day.

Make sure that the team you've put together is available for the duration (right through user training), at least on some level. If you've chosen a project leader, this is the person who will know all the details, even if he or she is not working daily on all of them. Once you've made the switch, so to speak, sit back and start enjoying the benefits of IP telephony.

The Bottom Line

You want to take your time implement IP telephony. A year may seem like a long time, but the more time you invest up front, the less money you're likely to waste overall. However, if you do not have a full year, this schedule can absolutely be accelerated—but do not skip steps, just shorten each cycle to fit your needs. The next chapter will go into more detail about reliability and what's required in order to ensure maximum uptime. Topics to be covered include redundancy, mean time to repair (MTTR), mean time between failures (MTBF), and network and applications reliability.

Ensuring Reliability in IP Telephony

Chapter 4

TABLE OF CONTENTS

This chapter covers varying IP telephony solution architectures, mean time between failure, mean time to repair, network reliability, and application reliability.

Ensuring Reliability in IP Telephony

Chapter 4

The most crucial characteristic of a business phone system is reliability. You must pick up the phone to a dial tone, you must be able to successfully place outgoing calls, and calls must effectively reach your organization. This chapter covers varying IP telephony solution architectures, mean time between failure, mean time to repair, network reliability, and application reliability. It is meant to help you dig deeper into the solutions you've narrowed down to your short list so that you can choose the one that fits best into your organization and existing infrastructure and provide you with maximum uptime.

How is Reliability Different from Availability?

Usually when reliability is mentioned in terms of a voice system, the reference is generally about hardware. Without hardware reliability, the system cannot be reliable. Reliability is determined by calculating how often the system fails compared to the percentage of time the system is available. In the telephony world, "five-nines" reliability is the acceptable benchmark. This means the system is available at least 99.999 percent of the time.

Availability, on the other hand, is predicted based on the probability of a hardware component failure. It is predicted by taking into account the type and number of hardware components in a system and calculating the mean time between failure (MTBF). So, if an IP switch has a predicted MTBF of approximately 135,600 hours, and each failure requires one (1) hour of mean time to repair (MTTR), we would use this simple computation to estimate the availability:

$$\text{Availability} = \frac{\text{MTBF}}{\text{MTBF} + \text{MTTR}} = \frac{135{,}600}{135{,}600 + 1} = 99.9993\%$$

This demonstrates that this particular unit will achieve "five-nines of availability." Alternatively, this switch is predicted to be unavailable for one hour every 10 years.

Let's take a household example. Consider a toaster that works for a year (an average year is 365.2425 days = 8,765.82 hours or 8,766 hours), and then it breaks, so you have to replace it: MTBF = one year. You take it to the store for a replacement the next day: MTTR = 24 (one day).

$$\text{Availability} = \frac{\text{MTBF}}{\text{MTBF} + \text{MTTR}} = \frac{8.766 \text{ hours}}{8.766 \text{ hours} + 24 \text{ hours}} = 99.7\%$$

This indicates two-nines availability. However, if you keep an extra toaster on hand, MTTR could be as little as fifteen minutes (.25 hours). While this increases the cost of equipment, it also increases the availability fairly significantly.

$$\text{Availability} = \frac{8.766 \text{ hours}}{8.766 \text{ hours} + .25 \text{ hours}} = 99.997\% \text{ or four-nines of availability}$$

Back to industry terms, there is no ordinary telephone system that can achieve five-nines. Since state-of-the-art MTBF for systems is 100,000 hours and MTTR is 24 hours, you would need to deliver 2,400,000 hours between failures to achieve five-nines.

$$\text{Availability} = \frac{\text{MTBF}}{\text{MTBF} + 24} = \frac{2,400.000}{2,400,000 + 24}$$

Even repairing the problem in 4 hours doesn't make it much easier to accomplish:

$$99.9990\% = \frac{400,000}{400,000 + 4}$$

You would still need 400,000 hours between failures. These examples are far beyond state-of-the-art. The way to meet these demands is via redundancy. Read on for a section on redundancy and specifically n+1 redundancy.

Distributed vs. Centralized, Chassis vs. Modular

IP telephony systems differ in their architectures: Some are centralized while others are distributed. In a centralized setup, the centralized call control server provides dial tone for all phones, whereas a distributed model is one where end points are handled by multiple call control servers. In this solution, call control is provided by each switch in the system. See figure 4.1.

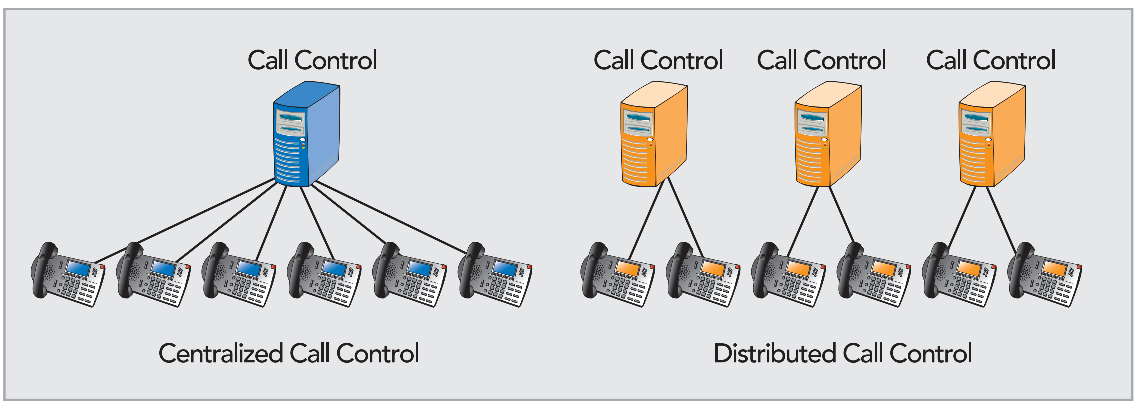


Figure 4.1. Centralized vs. Distributed Call Control

A classic chassis includes a number of circuit boards, with most of them providing telephony interfaces and one consisting of a specialized computer system, while some modular units contain a single board. The classic chassis can be compared to a string of holiday lights: If one bulb fails, the entire segment fails. The more lights on the string (number of circuit boards in the chassis), the more vulnerable it becomes to failures.

A typical chassis model, because you have to take into consideration the reliability of their components, typically has an MTBF in the 50,000 range, which is four (not five) nines availability. This can be raised to five-nines by adding switches for redundancy (costly but effective). More on this will be discussed in the n+1 redundancy section later.

In contrast, a modular architecture includes small, simple and reliable hardware. This modularity is more reliable and also offers more freedom in the design stages of an IP telephony implementation. Look at both modular and chassis-based systems, but keep in mind your specific reliability needs and remember that modular systems generally make configuration changes simpler and seamless.

The Bathtub Curve

Electronic product failures historically demonstrate a failure profile known as a "bathtub curve." See Figure 4.2 for a depiction of the bathtub curve. Because of a number of reasons, including stress, electronics tend to have a short life before they start failing. At the beginning of the lifecycle (the left side of the diagram), manufacturing defects, defective parts, contamination and other factors cause failures, before these settle to a much lower level (the middle of the diagram). The other end (on the right) signifies the end of life or wearing out of the product.

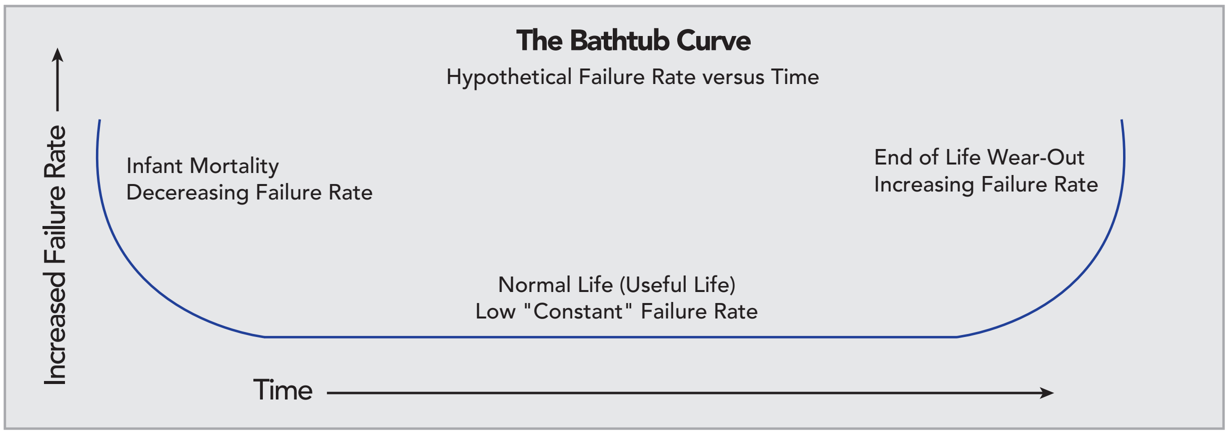


Figure 4.2 The Bathtub Curve

Be sure and ask vendors about their failure rates and how long a product lasts before end of life. If a vendor does not give you a concrete number based on scientific calculations (not marketing hype), ask more questions or talk to someone at the organization who can give you that information.

Mean Time To Repair (MTTR)

When a product is down, the entire system's availability percentage is dramatically affected. Consider the following example, where MTTR goes from 1 to 24 hours.

Long Repair Times Slash Availability Ratings

MTBF = 135,600 hours
Availability = MTBF/(MTBF+MTTR)

Availability with 1-hour MTTR = 135,600/(135,600+1) = 99.9993%

Availability with 24-hour MTTR = 135,600/(135,600+24) = 99.9789%

Figure 4.3 Comparing MTBF with varying MTTR

The more complex an IP telephone system, the longer it's going to take to identify what's going wrong during a failure. Only when you've identified what's wrong can you get a replacement for it, which can take even more time, and then there is the time it takes to get the system back up and running. Because of this, chassis systems described earlier in this chapter require personnel with significantly more expertise to ensure the system remains functional.

A 4-hour MTTR is industry standard, which creates a problem for IP vendors that want to maintain five-nines of availability with a 4-hour MTTR. Redundant systems are usually added to ensure this availability because a 4-hour MTTR requires a 400,000-hour MTBF to achieve 99.999% availability. (Availability = MTBF/(MTBF+MTTR) = 400,000/(400,000+4) = 99.999%.) Modular, distributed systems tend to make system repair easy, which results in a lower MTTR. These systems only require one power source and two or three cable connections.

Moving Parts and Complexity

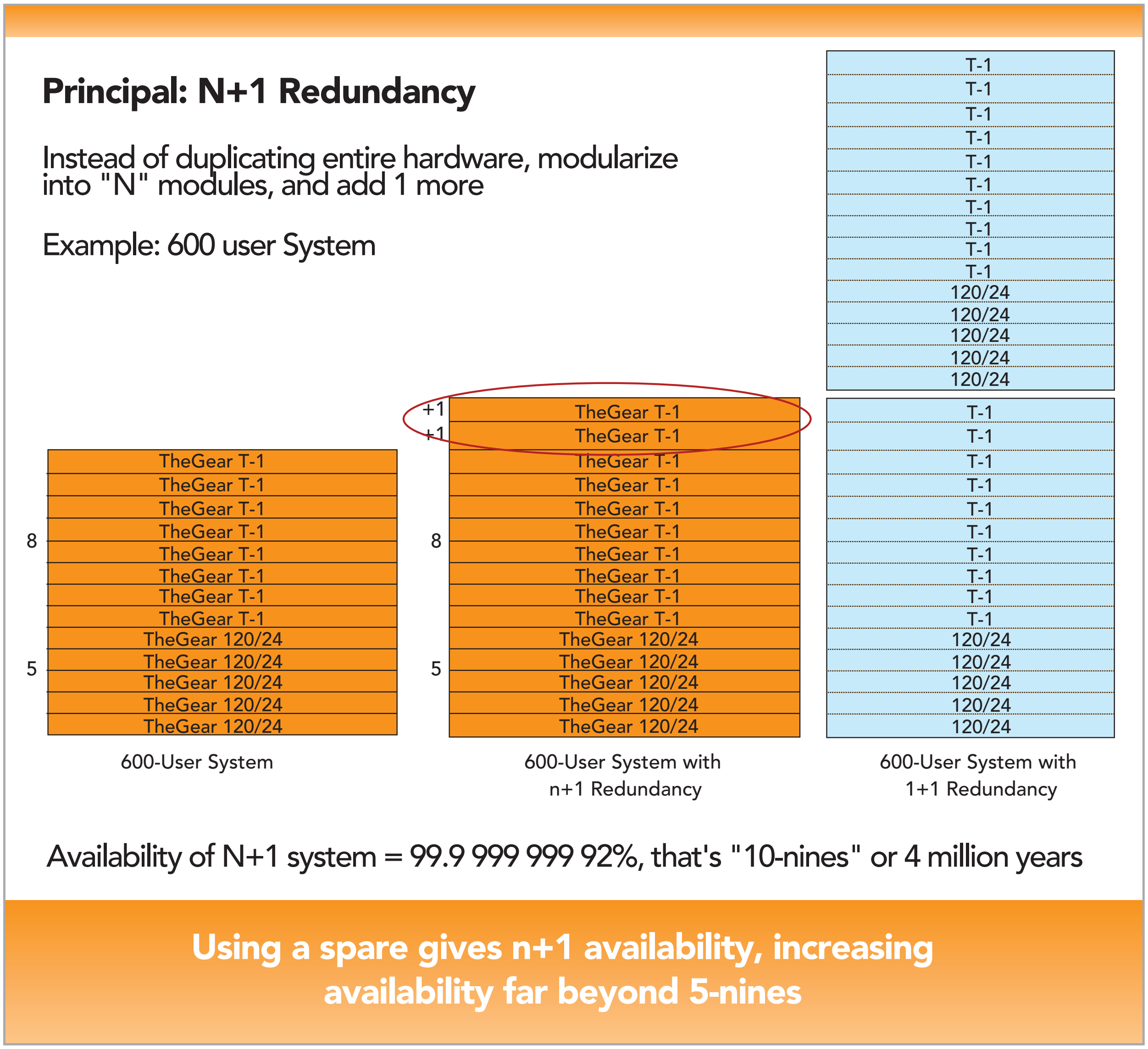
Another thing to keep in mind is the number of moving parts there are in a system. For instance, adding a disc drive (rather than flash memory) with a 500,000-hour MTBF cuts the system's overall MTBF in half. Moving parts are also likelier to wear out faster than non-moving parts. For instance, the bathtub curve for disc drives is steep and it's often recommended they be replaced well before end of life to avoid failure. In the case of an IP telephony solution, you'd be replacing a disc drive during the time you have it on your network, since most disc drives last five years. Ask each vendor how many moving parts there are in each system. Again, insist on getting the information from another company source if the sales team does not have this information readily available.

Redundancy also impacts the failure rate, ironically. While vendors often add redundant parts, such as disc drives and power supplies, to their systems, the very fact that the number of parts are being doubled in itself can increase the chance that the system will fail (increase the MTBF). When you are considering an IP PBX system for your organization, be sure to look at how complex each system is. The more complex, the longer it takes to repair because problem diagnosis, part replacement, and system restoration can be difficult. Look for modular systems that are easy to manage and troubleshoot, with specific built-in tools to ensure quick and easy diagnosis and repair.

N+1 Redundancy and No Single Point of Failure

Look for a solution with a distributed architecture that allows for the use of n+1 redundancy, which means that extra parts—as opposed to entire units—can be added to provide redundancy. Some vendors have 1:1 redundancy, which means twice the hardware is used to accomplish redundancy. Other systems use n+1 redundancy—which improves reliability since it is not doubling the hardware. For instance, the n+1 redundancy solution may need two extra units (where parts to the IP telephony system are duplicated within the two units), while a 1:1 redundancy solution needs five extra units because each unit is duplicated in its entirety. Essentially, using n+1 redundancy creates a multi-unit system with no single point of failure.

In addition to a distributed architecture that provides n+1 redundancy, look for a solution that interconnects each module using IP rather than cards in box slots. This design uses the Internet as a bus rather than having a proprietary backplane, which allows you to use a wide variety of chips and software and also reduces the costs and increases speed because of the use of IP and Ethernet. This design also allows you to seamlessly scale your system to meet organizational growth demands, just as the Internet allows for growth. Finally, look for a system that provides most of its feature upgrades via software so that there is minimal time between the release and your organization's use of these features.



The goal of five-nines reliability is impossible for most systems because redundancy requirements can be complex and expensive. Using n+1 redundancy is not only more cost-effective, but it is less complex, which in turn reduces the chance of failure.

Network Reliability

The biggest hurdle when implementing an IP telephony solution is ensuring it works properly with the existing underlying infrastructure. LANs and WANs have lower reliability than telecommunications systems and are prone to quality-of-service (QoS) issues that make IP telephony solutions unreliable. LANs have multiple serial components, which negatively affects the reliability (typical LANs achieve three to four nines of availability), but it is possible to achieve five-nines availability on a network by using a redundant aggregation switch with redundant paths. After all, four-nines reliability translates to two hours of downtime per year. Can your organization afford that? Most 24/7 operations cannot. Focus on solutions that allow these redundant paths to an aggregation switch.

WANs cause the biggest headache because WAN links are generally available only 99% to 99.9% of the time, and voice quality availability can be as low as 98%. If your employees depend on superior voice quality for their many conference calls, for example, this is going to be a problem. Some solutions exist that distribute call control to local switches, which means that if a WAN link goes down, a remote switch can handle the calls because call control, business logic and system database information are all available within that switch.

A system with centralized call control relies heavily on its WAN connection because when it goes down, remote sites have no call control, which means calls cannot be made unless a backup system is in place. Look for a distributed solution that provides full and seamless call control functionality even during a WAN failure.

Application Reliability

In addition to ensuring your system is reliable in terms of hardware, you must also ensure that IP telephony system applications, including auto-attendants, voice mail, and desktop integration, work all the time for your employees. Look at systems that offer one application server for a full range of applications. You can use more than one server depending on your organization size, but make sure that it is not one feature per server, like some solutions may force you to do. A truly reliable system, in terms of applications, uses a site hierarchy, which means the first application in a user's hierarchy is used, and each application server has access to the configuration database in a central server. This design is highly reliable because each application server caches the configuration database, making information and applications available even during network downtime. For example, in the case of a network outage, remote users with their own server are unaffected by a failure in another server so that individual sites can serve features like auto-attendant.

The Bottom Line

There is always the possibility that a system can be completely unreachable because of multiple LAN and WAN problems (remember the saying, "never say never"). Look for solutions that allow you to build into your system a backup plan, such as the ability to implement failover trunks, switch failover, and copper bypass for emergency service. There are lots of vendors out there offering piecemeal solutions that could leave you dealing with increased complexity and decreased reliability. A distributed architecture is a good fit for multi-site organizations, and n+1 redundancy designs will keep your costs—and your chances of failure—way down. The next chapter will go over system handsets, including analog and IP telephones as well as hard and "soft" phones.

Handsets and Interfaces

Chapter 5

TABLE OF CONTENTS

This chapter outlines the many benefits of todays' well-designed and highly functional telephones.

Handsets and Interfaces

Chapter 5

Think about your home telephone, your cell phone, or any one of the multiple electronic devices you use everyday. You expect—and appreciate—a well-designed product. You shop for these items with design and functionality in mind. With IP telephony, you can now bring the same high expectations into the office and into your search for handsets. Mediocre office telephones are a thing of the past because IP telephone handsets introduce so many more features and benefits.

The Need

Business workers rely on the telephone many hours out of the day, from collaborating with business partners and co-workers to interacting with and helping customers and suppliers. Call center professionals literally spend the entire day on their telephones. It's not enough to "make do" with a standard, feature-lacking desktop handset. To make employees more productive—and happier—you need to provide them with the tools they need to do their jobs optimally. You'll only do this when you present them with a handset that is ergonomically well-designed, has great sound quality, and features a multitude of capabilities at the touch of a button.

Ergonomics

Ergonomics is the science of designing products, machines and systems to maximize the safety, comfort and efficiency of the people who use them. Ergonomics takes into account psychology, physical measurement, environment, and more to ensure that products are adapted to suit workers and their specific needs. Keep ergonomics in mind as you look at the handsets and graphical user interfaces (GUIs) of each vendor's solutions. If your organization is a machine shop, the most important feature for your handsets may be a very loud ringer. If you have a call center staff, a bevy of features that help shorten the call cycle will be most beneficial. A law firm may require a system that logs incoming and outgoing calls and keeps this information on record for future reference. A recording studio may require ultra-clear sound quality to ensure recorded voices are pitch-perfect. Look at your organizational needs in terms of what you need a handset and GUI to do for your employees.

Sound

IP telephony, with its packet-based design, is able to deliver better than toll-quality sound with hi-fidelity audio and innovative design. Better sound translates into productivity gains – shorter calls with fewer errors, increased sales because of the clarity of conversation between a sales person and customer, and increased caller satisfaction. Wideband audio is preferable over narrowband, because it has an increased range on the low end (50-300 Hz) and makes conversations sound less tinny and reduces error in translation. Look for a solution that supports both wideband and narrowband.

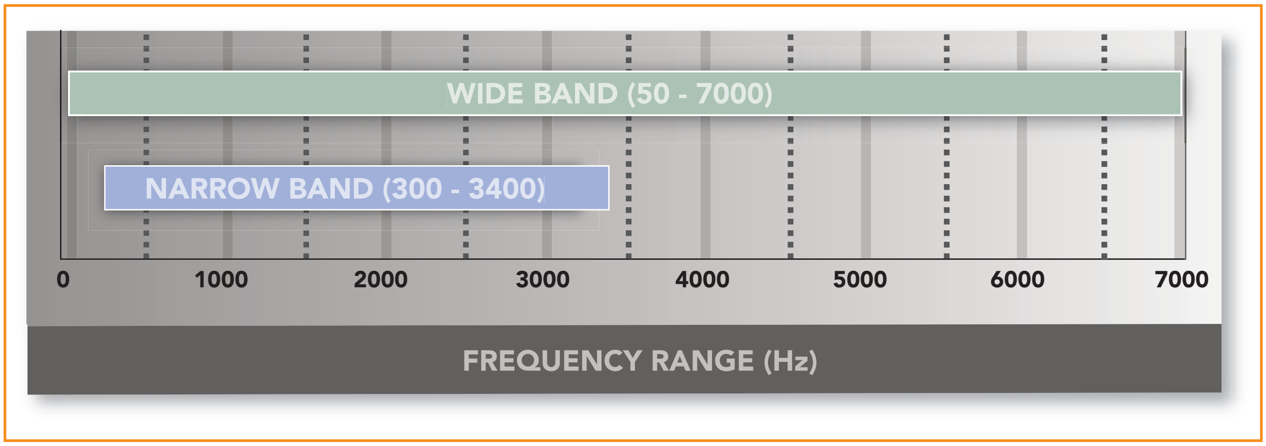


Figure 5.1 Wideband audio technology provides bandwidth from 50Hz to 7,000 Hz; narrowband provides 300 Hz to 3,400 Hz. Wideband delivers superior speech quality.

Speakerphone microphones are also an important part of sound quality consideration. Look for a solution that supports hi-fidelity sound and has a full-duplex operation speakerphone so audio flows freely on both ends (no delay if one speaker talks over another). Not all IP speakerphones are able to do this. In addition, ensure you choose a handset that meets the Americans with Disabilities Act (ADA) regulations for the hearing impaired, regardless of whether you have an immediate need or not. (More on ADA compliance will be covered later in this chapter.)

Screen Interface

IP telephones act more like computers than telephones—they have a bigger screen and more functionality attached to the screen. This screen also delivers more information about each call and prompts the user through the call with various options appearing on the screen. The user simply presses a corresponding key below the screen to accomplish any task while on the call (call forward, conference, etc.).

Make sure to consider carefully the size of the screen, with your users in mind. Is it big enough that after a long day of work, it's still pleasing to the eye? Is the display big and bright enough to see clearly after four hours on the phone? In addition, work with the phone and test what features are available and how easy those features are to access for a call center worker taking up to 50 calls an hour. Is there a message waiting light to ensure no message is missed?

User Considerations

Another characteristic to consider is the feel of the phone, since that is another source of fatigue for users. The phone should minimize shoulder and neck pain and fatigue, and it should essentially fit most users comfortably. The handset should not be too light or too heavy—try and get a phone with a balanced weight of about 170-190 grams. Also, consider a handset with a grip that is covered with a smooth rubber material, as opposed to the slippery plastic kind that can become uncomfortable during long telephone calls.

Keypad Functionality

Many systems will come with fixed-feature keys. Make sure those features are the one most pertinent to the needs of your employees. Fixed features usually include transfer, conference, intercom, voice mail dialup, directory, and redial. If a system relies mostly on soft keys, consider how difficult it may be for all your users to get those soft keys set up and working. Will you end up having to go around to every employee's phone to program two or three soft key functions? Soft keys are beneficial to have, but they should not be used for standard functions—these should be on hard keys. Some functions that you should look for on fixed-feature keys include:

- Directory: This key should be linked to a quick-dial program that allows a caller to dial by name using the telephone keypad (7 for S, 2 for A, 6 for M, which would bring up names that match beginning letters "SAM").
- Redial: This function key should do more than simply dial the last number dialed—it should allow you to press it and see an historical list of outbound, inbound and missed calls.
- Personal options: This feature key should allow for easy management of personal options, such as ring tone and call handling preferences.
- Voice mail: This key should provide quick and easy access to voice mail messages.

Soft Keys

Soft keys are multi-function keys that use part of the telephone display to identify their function at any moment. They are usually located directly underneath the display and their use changes depending on where the user is in the call process. You can set some soft keys for use by all of your employees, and you can choose to leave some to the discretion of each user. Make sure the setup of soft keys is straight-forward before allowing users to set up their own. If the IP telephony system you've chosen does not offer handset soft keys that are easy to set up or change, make sure the solution allows you to either set the soft keys for each user (or block users from trying to set up their own) or the ability to choose not to use the soft keys at all. This will minimize user confusion and frustration if the solution is difficult to edit.

Business vs. Basic Phones

Basic phones differ greatly from business phones in that they offer few or no additional functions beyond answering and hanging up. Business phones streamline tasks and offer users productivity enhancing features. You'll find that some vendors offer most functionality via soft keys, while others rely on numerous hard keys—one function per key. Your employees may fare best with fixed function keys, or classic business phones, which generally have a button per task. Some vendors do not offer this, however, relying mostly on soft keys. Still other phones offer a fixed number of hard keys and some extra hard keys that you can program to fit your organizational needs. These are optimal for organizations with workgroups that need specific functions to be programmed into keys.

Easy to Manage

You want to make sure the phones you are getting with the IP telephony system you choose are plug-and-play, particularly if you have a large organization with many locations, some of which have no technical staff on-hand for installation support. Non-technical employees should be able to plug in their phone and start working. When it's plugged in, the phone should automatically get its IP address, subnet mask, and gateway, as well as the accurate time from a time server. Handset updates should be equally as hands-off for employees—updates should be automatic as they are released by the vendor.

Aesthetics

While most businesses do not place emphasis on how a phone looks over the functionality, it is still an important consideration. A phone that is pleasing to the eye is as impressive as a beautiful desk or sleek-looking computer. Consider your options with your chosen vendor and ask about variety. What colors do their phones come in? Are there smaller versions for users who need minimal functionality? Are there ruggedized versions of the IP phones for public area usage? Look for a solution that will fit all your needs, with phones that are consistent in appearance and look classy throughout your organization.

Phone Choices

In an IP telephony solution, the IP-PBX manages telephones throughout the enterprise and acts as a gateway to both voice and data networks. Any kind of telephone, whether it be analog, IP or a soft phone, can connect to the IP-PBX via the network and calls are routed via the network instead of the public switched telephone network.

Analog phones

A regular analog telephone, the same ones you've been using throughout your organization until now, can be used in an IP telephony solution to input the caller's voice into the system. Once in the system, a series of analog-to-digital conversions and other processes change the voice signals into data, which is then transmitted over the LAN, WAN, or Internet. The voice data is then converted back into sound by the recipient's phone. Most IP telephony systems will allow you to use your existing analog telephones with the solution—forever or until you are able to afford and/or replace them with IP telephones. Be sure that your vendor will allow you to phase out older analog phones with their IP phones over time so you can maximize your existing equipment.

IP phones

IP telephones (or IP endpoints) actually perform the analog-to-digital and/or digital-to-analog conversions and can plug directly into the LAN or WAN. VoIP system vendors usually offer a variety of IP telephones so that you can choose different models based on various segments in your user population. Your legal department may need multi-line handsets with easy conference call capabilities. A manufacturing floor needs a phone with fewer bells and whistles but good, loud sound and a rugged exterior. Receptionists need handsets with many more fixed feature buttons so that they can handle calls quickly and accurately.

Soft phones

A soft phone is essentially software that is used to make calls over an IP telephony system using a personal desktop computer and either a headset connected to the computer's sound card, or a telephone connected to the computer using an adapter. It behaves like a traditional phone but usually offers much more information to the user, depending on the vendor's GUI. When a call comes into a station with a softphone, an icon appears on the computer screen, which allows the user to either answer it by clicking on an icon, or ignore the call by clicking on another icon, which in turn sends the caller to either voice mail or another employee.

Often, vendors offer an application that allows traveling employees to gain access to the robust feature set of their desktop computer from wherever they are working—at home or on the road. A user simply logs into the system from the local phone and has access to all of the same functions he or she would enjoy while in the office.

WiFi phones

WiFi phones use signals much like those used by cordless telephones. The WiFi phone receives signals which allow you to wirelessly connect to the network via wireless access points (APs). Unlike traditional cell phones, the technology of WiFi phones allows them to transmit data at really high speed, but areas of coverage are limited by the reach of the AP being utilized. There are also hot spots available in various locations (restaurants, Starbucks, libraries, etc.) that allow you to access the Internet using your own WiFi service (or a service utilized by your organization).

One drawback to WiFi phones is the fact that some things can impede on the quality of the calls, such as how many people are using the same hot spot, how close the WiFi phone user is to the access points, WiFi card capabilities, and possible obstructions to the AP (such as a wall). Another drawback is that WiFi technology does not offer the level of security offered with standard Internet access. More on security will be covered in the following chapter.

Want a SIP?

Session Initiation Protocol (SIP), a signaling protocol, is used for establishing a session in an IP network—from a simple two-way telephone call to a multi-media conference call session with many participants. The IP telephony industry has recently adopted SIP, an RFC standard (RFC 3261) from the Internet Engineering Task Force (IETF), as the protocol of choice for signaling because of its ability to facilitate Internet applications by working with other protocols. It is not the be-all and end-all of protocols—it was designed to be a facilitation mechanism, not an all-inclusive solution. Its flexibility is what makes it so powerful, and an all-inclusive approach does not offer this level of flexibility.

Essentially, SIP establishes, manipulates and tears down sessions, and its main purpose is to help session originators deliver invitations to potential session participants wherever they may be. It uses URLs to address participants and SDP to convey session information and it's easy to combine SIP with other applications, like Web browsers and messaging. The bottom line is that it's a modular approach to maximizing IP telephony protocols. SIP can find and invite call invitees wherever they are. It facilitates multi-media calls with many participants who may join and leave at will.

American Disabilities Act (ADA) Compliance

Your IP telephony system must comply with the American Disabilities Act (ADA) of 1990 and associated regulations issued by Federal agencies that define guidelines for accessibility by individuals with disabilities. These guidelines include requirements for telephones and telephone systems, and they include the "ADA Standard for Accessible Design" (Pt. 36, Appendix A, Section 4.31, Telephones) and the 508 provision for TDD/TTYs. A few of these requirements include:

- Volume Control: Telephones should have volume controls that provide a gain adjustable up to a minimum of 20 dB. The telephones should provide at least one intermediate step of 12 dB for incremental volume control.
- Automatic Volume Reset: The telephone should automatically reset the volume to the default level after every use.
- Hearing Aid Compatibility: The telephone must have a means for effective magnetic wireless coupling to hearing technologies.
- Minimized Interference: Interference to hearing technologies, including hearing aids, cochlear implants, and assistive listening devices, shall be reduced to the lowest possible level that allows a user of hearing technologies to use the telephone.

- Support for TDD/TTYs: Products that transmit or conduct information or communication shall pass through cross-manufacturer, non-proprietary, industry-standard codes, translation protocols, formats or other information necessary to provide the information or communication in a usable format. Technologies which use encoding, signal compression, format transformation, or similar techniques shall not remove information needed for access or shall restore it upon delivery.
- Controls and Keys: Controls and keys shall be tactilely discernible without activating the controls or keys. These controls and keys shall be operable with one hand and shall not require tight grasping, pinching or twisting of the wrist. The force required to activate controls and keys shall be 5 lbs. maximum. If key repeat is supported, the delay before repeat shall be adjustable to at least 2 seconds. The status of all controls or keys should be visually discernible, and discernible either through touch or sound.
- The cord from the telephone to the handset shall be at least 29 inches (735 mm) long.
- A wall-mounted object should not protrude into the walkway more than four inches to ensure visually impaired individuals do not run into them.

The Bottom Line

By now, you have either chosen your IP telephony vendor or at least narrowed it down to a short list. Take the telephone characteristics into account to help you finalize the decision. If you have already made your choice, look carefully at all of the models your vendor offers and choose the right phone for each user in your organization: Multi-function telephones for receptionists, soft phone licenses or WiFi phones for travelers, basic but ruggedized phones for warehouses and manufacturing floors. At this stage in the IP telephony game, you have more options than ever and you don't need to make one model work for everyone. What you do need to do is make sure your users are more productive because of the phones, and that your choice complies with the ADA. The next chapter will cover how you can secure your IP telephony communications.

Security

Chapter 6

TABLE OF CONTENTS

This chapter highlights the steps one should take to ensure IP telephony traffic is secure against outsiders and unauthorized individuals.

Security

Chapter 6

Anybody who's connected to the Internet or who owns a PDA/multi-function cell phone knows that they're at risk of getting viruses, worms, spam and other malicious threats. In addition to the potential damage these threats introduce in terms of lost data or corrupted files, there are now regulatory issues associated with ensuring protection. Healthcare has its own privacy regulations in the form of HIPAA (Health Insurance Portability and Accountability Act of 1996), and infringements can result in significant punishments and fines. The bottom line is that you have to protect your organization's devices and network. IP telephony is no different – the only difference is the form of the traffic: voice versus data. All traffic crossing a network can be stolen, manipulated or blocked if proper network security precautions are not put into place. This chapter will highlight the steps you should take to ensure your IP telephony traffic is secure against outsiders and unauthorized individuals.

Evaluate your Risks

The first step to determining the right network security strategy (VoIP or otherwise) is to determine the risks your particular organization faces. (Because of the increasingly complex network security threats and solutions out there, you may want to get a network security expert on board to help with the assessment.) For instance, a healthcare organization faces different regulatory requirements than a legal or accounting firm. An e-commerce organization has altogether different privacy and security requirements. Once you determine what your risks are, you'll be better able to determine the best multi-layer defense against attacks, eavesdropping, service theft and other evolving threats for the entire network, including the IP telephony system being utilized.

IP Telephony - Specific Considerations

Network-Based Attacks. IP telephony is susceptible to Denial of Service (DoS) attacks because these can cripple the network to the point that nothing, including voice calls, can get through. (It is generally recommended that every organization using IP telephony have backup telephone lines in the case of an out of control DoS attack or regional power failure.) Spam, spyware and phishing are other network attacks that are commonly used to commit identity theft and other fraud. Finally, viruses and bots can destroy data or devices or even hijack phones into a toll fraud scheme.

Phone Service Theft. A hacker could enter into an unprotected network and access the PBX to make endless international calls. There have been major cases cited in the news where toll fraud has cost companies millions of dollars. In many instances, the criminals have been caught and prosecuted, but not without major costs to the companies defrauded; and keep in mind, there are always those crimes that go undetected.

Eavesdropping. Without the proper security in place, a hacker could eavesdrop and possibly expose confidential information. A private conversation about financials could be recorded and played for anybody, which could lead to internal and external problems, including punishment from numerous regulatory agencies. Or a personal call from an employee to a florist with a credit card number could lead to credit card and even identity theft.

Power Failures. While outages affect data traffic, of course, there's a difference when it comes to telephony. People expect telephones to work even during an outage because homes often have a non-electronic phone that simply plugs into the telephone outlet. This expectation is generally brought into the workplace.

SPIT. Spam over Internet telephony is an alternative to telemarketing where one message can easily be sent to thousands of recipients with the click of a mouse. In other words, your employees' voice mail boxes can become as overloaded with spam as their e-mail would be without appropriate spam filters.

Other Threats. There are new threats created and discovered daily. One such attack is the spoofing of a phone number, which essentially allows a hacker to look like he or she is someone else, which is one of the easiest ways for this person to steal an unsuspecting person's identity. While individuals have learned not to trust e-mail, it is still generally believed that telephone communications can be trusted.

Network Security Basics

Network security will lead to a secure IP telephony system. Your organization has likely taken steps such as initiating the use of virtual private networks (VPNs) and installing firewall equipment, which protects the organization against intruders and threats mentioned earlier. Since voice is just another application on the network, the same precautions should be taken to secure the IP telephony equipment. Every form of security should be applied, including physical, human, network, and system security.

- Physical security: Buildings, equipment rooms, data servers, and wiring closets should be off-limits to anybody who is not authorized.
- Human security via security policies: Make sure your organization's informational assets are protected against inappropriate or unauthorized use by a renegade employee. Ensure hiring and system usage policies are in place to govern appropriate use. Establish and strictly enforce policies having to do with passwords and system usage.
- Network security: Again, create a multi-layered defense using firewalls, VPNs, and intrusion detection or prevention (IDS/IPS). Make sure wireless access points use the highest level of access control and encryption to prevent intruders from gaining access to your network and its resources.
- System security: Arm every desktop with anti-virus software to fight against spyware and other malware. Utilize host intrusion prevention systems to protect servers against attacks.

Another force to consider is segregating traffic via virtual LANs (VLANs). It is a method of logically grouping devices or departments onto their own LANs. Isolating LANs from one another provides an additional layer of security. It also reduces the impact of multicast or broadcast traffic since there are separate broadcast domains.

Finally, bandwidth management can be utilized to further guarantee bandwidth for business-critical, latency-sensitive traffic like VoIP traffic. Bandwidth management methods include assigning a certain priority to each type of traffic. VoIP packets should be assigned the highest priority to ensure voice traffic gets through.

IP Telephony Security Basics

When your network is secured, take it a step further and utilize best practices for deploying secure IP telephony.

- Firewalls: Make sure the firewalls you're using can handle the latency sensitive needs of IP telephony traffic.
- Switched environment: Use Ethernet switches (not hubs) to connect all your voice devices not only for better performance but also to limit the possibility of a hacker getting onto a call because in a

switched environment, the flow of traffic is between devices and nobody can tap in.

- VLAN assignment: Assign voice to a separate VLAN (or separate VLANs). This segregates traffic for improved performance and security.
- Priority: Prioritize voice traffic over data on these VLANs so that delay sensitive traffic gets through even during a network attack. Ensure your network switches can prioritize based on VLAN tags and support multiple queues.
- VPN: Use a VPN between sites, buildings, or departments to encrypt traffic. This is especially important when it comes to protecting confidential employee information, such as social security numbers. In addition, use software VPNs or VPN appliances for remote users to protect conversations from being tapped. Your system should also offer you the option of completely disallowing remote access for an even tighter security option.
- Port lockdown: Lock down IP telephony traffic on the physical switch ports so that only authorized MAC addresses can transmit over the port.
- Media encryption: Look for a solution that prevents eavesdropping by encrypting voice traffic. This way, even if someone taps a voice stream, they are unable to decode or understand the conversation. Not all IP telephony system vendors offer this but it is a necessity for IP telephony security.
- Voice mail storage: Make sure that your voice mail storage is itself secure to prevent unauthorized access of voice mail files.

IP Telephony System Security

Let's look now at the IP telephony system itself. While you can secure your network in all the right ways, you also need to choose a phone system that is secure itself. Consider moving away from a system that uses Microsoft Windows for call control because of the security considerations. With a constant stream of Windows security updates and patches, you're risking downtime and security breaches.

Another architectural consideration to keep in mind is ensuring your system is distributed, which will mean it has no single point of failure. A distributed system allows continued operation in the case of worms, viruses, or DoS attacks. An attack will not disable the entire system if intelligence is distributed amongst multiple devices.

Your chosen system should offer multiple levels for administrator permissions to limit control and ensure unauthorized individuals do not gain access. Once you've deployed, reserve full access for just a few key information technology employees. Ensure that a web-based management solution supports secure management using Secure Sockets Layer (SSL), which secures communications from the interface to the server.

According to the SANS (SysAdmin, Audit, Network, Security) Institute, a cooperative research and education organization, VoIP servers and phones are at significant security risk. The organization's 2006 annual update, SANS Top-20 Internet Security Attack Targets, indicates that there's been an increase in security scrutiny of IP telephony, especially on typical components such as the call proxy and media servers, as well as the phones themselves. Some products have been found to contain vulnerabilities that can either lead to a crash or a complete control over the server or device. "By gaining a control over the VoIP server and phones, an attacker could carry out phishing scams, eavesdropping, toll fraud or denial-of-service attacks."

How to Mitigate IP Telephony Vulnerabilities

SANS has determined and published a list of things enterprises must do to mitigate the IP telephony vulnerabilities mentioned in this chapter.

- Apply the vendor supplied patches for VoIP servers and phone software/firmware.
- Ensure that the operating system running the VoIP server is patched with the latest OS patch supplied by either the OS vendor or the VoIP product vendor.
- Scan VoIP servers and phones to detect open ports. Firewall all ports from the Internet that are not required for keeping up the VoIP infrastructure.
- Use a VoIP protocol aware firewall or Intrusion Prevention product to ensure that all UDP ports on VoIP phones are not open to the Internet for RTP/RTCP communications.
- Disable all the unnecessary services on phones and servers (telnet, HTTP etc.).
- Use VoIP "protocol fuzzing tools" such as OULU SIP PROTOS Suite against the VoIP components to ensure the VoIP protocol stack integrity.
- Additional caution should be taken at the product selection phase to ensure the VoIP product vendor supports OS patches as they are released. Many VoIP vendors will void support for unapproved patches and may take considerable time before approving them.
- Apply separate VLANs to your voice and data network as much as your converged network will allow. Ensure that VoIP DHCP and TFTP servers are separate from your data network.
- Change the default passwords on phones' and proxies' administrative login functions.

Source: SANS Top-20 Internet Security Attack Targets, 2006 Annual Update

The Bottom Line

IP telephony requires the same level of security as your data network requires. You need to ensure you're receiving calls from trusted sources, you're protecting your infrastructure from toll fraud, and you need to make sure your voice calls get through, even when parts of the network might be bogged down by DoS attacks, viruses, or worms. There are vendors that offer IP telephony solutions with additional layers of security. You don't have to rely solely on network security devices in place. You can take it a step further and protect your IP telephony equipment so that voice communications and resources are as safe as possible from hackers and other criminals. The next chapter will discuss wireless IP telephony, including more on security, as well as QoS, reliability, and coverage areas.

Mobility and Wireless

Chapter 7

TABLE OF CONTENTS

Mobility is an absolute necessity, as is the requirement for customers to reach anyone at anytime, anywhere. IP telephony is the ideal way to meet this need.

Mobility and Wireless

Chapter 7

In addition to cost savings, productivity improvements, and customer service enhancements, another driving force behind IP telephony is mobility. Workers are increasingly mobile—from traveling sales people to call center staffers who work from remote sites and even home offices around the globe to serve customers 24/7. Mobility is an absolute necessity, as is the requirement for customers to reach anyone at anytime, anywhere. IP telephony is the ideal way to meet this need. With it, organizations can use distributed hunt groups to ring employees around the globe with the right skill set to ensure a question is answered or an issue is solved immediately. As long as an agent with the skill set is logged in, even if on another continent, the issue will be resolved just as if he or she were at their own desk.

With IP telephony, calls are intelligently routed based on calendars, so agents logged as out of the office are reached via cell phone, etc. At the same time, the agent's cell phone acts as an extension of their desk phone with all of the integrated features, such as dial-by-name, transfer, conference call capabilities, etc. This wireless integration is crucial, especially since you don't necessarily need to purchase specific wireless handsets or specialized handsets for traveling employees.

This mobility is not even noticeable to the customer base. There are IP telephony vendors that allow employees to choose their device—for instance, a cell phone or home phone—and that device assumes the identity and capabilities of his or her regular office extension. For example, the caller-ID information provided when the employee makes a call can reflect their office number instead of the mobile or home-office phone actually being used. In other words, caller-ID will indicate that the call is coming from headquarters of their company. This is important to protect the employee's privacy and strengthen the corporate brand.

Going Mobile

With IP telephony, users are highly mobile, logging in from anywhere and gaining access to all the same capabilities as if they were working at headquarters, at their desks, or within a call center building. With IP telephony, to the outside world, it can seem as though your organization has call center locations scattered around the globe, making help available 24/7. In reality, you are simply utilizing IP telephony features such as time-of-day routing and call forwarding to make sure calls are answered quickly by a live human being. Your employees can be working out of branch offices, at remote locations, or even at home. Your workers are mobile and happy; your customers are being catered to and satisfied quickly. You are also able to manage peak calling times by having the ability to add other employees, regardless of their location, to the call center to help meet the overflow demand.

With IP telephony, users can also easily re-route their calls so that they are reached wherever they will be working—they can make these changes themselves, without asking for IT assistance. This "find me" feature also enhances customer service, as well as productivity, by ensuring every call reaches the right person, regardless of where he or she might be working. An employee can even program his or her extension to ring based on status—ring through when he or she is in the office, forward to a cell phone when there is no answer, or forward to a colleague when the line is busy.

Wireless Next

Once you've deployed IP telephony on your network, you'll almost certainly begin to consider how you might add wireless to the mix. With the broad adoption of Wi-Fi networks based on IEEE 802.11, your employees will also inevitably ask you when you'll be offering them mobile IP telephony since they'll quickly grow accustomed to the productivity-boosting and time-saving benefits. With wireless, employees can take these benefits beyond the wired network.

With wireless IP telephony, employees are not tied to their desks and delays are further reduced. Consider, for instance, the case of the sales representative meeting with the CEO. While in a meeting, urgent calls can follow him or her to a wireless handset. Take this example into a hospital, and it can mean the matter of life and death if a nurse is visiting a patient whose health suddenly degrades. The nurse need not waste time running to the nursing station to call the doctor or paging for help but rather, he or she can call the doctor directly from within the patient's room from a wireless IP handset, provide information and take steps the doctor is advising all in real time as a result of the phone consult.

On top of the savings offered by IP telephony, going wireless can also save your organization additional money. For example, when an employee is working in another location other than his or her office, calls can still find that person if they are free to talk, thereby eliminating any toll charges that would have been associated with returning a missed call, had a caller gone to voice mail.

Prepping for Wireless

Until now, many companies have used proprietary wireless voice systems for their warehouses and distribution facilities, for instance, but today, there are standards in place—namely, Session Initiation Protocol (SIP)—for call control over wireless LANs (WLANs). There are other requirements your network must meet, such as sufficient wireless coverage, network scalability, Quality of Service (QoS), and seamless roaming, and of course, security.

Sufficient coverage

You don't want your users hitting dead zones while they're in the middle of a conversation. It's poor customer service and costly to your business. Assess how many users you have in each location of your organization, and consider the bandwidth requirements of the applications they are each running to ensure enough bandwidth for voice traffic over the WLAN. You will need to maximize performance by adding a sufficient number of wireless access points (APs) to each location where many users work. Keep in mind that since a WLAN is a radio frequency (RF) network, the physical environment will affect the coverage capabilities of each AP. Walls, glass partitions, and cubicle separators can affect the coverage area because these materials absorb signals. Take into account the physical characteristics of your organization and buildings and design your WLAN plan to meet these challenges. A physical survey before deployment will help you determine how many APs and switches you'll need to meet coverage requirements. Keep in mind, however, that the more APs you add to a particular area will affect performance in terms of possible interference.

Scalability

As mentioned earlier, you can meet wireless traffic needs by adding APs to any given area. However, there is also the risk of interference when too many APs are working too closely together. Be sure to plan carefully and run tests to ensure smooth call delivery so that crucial voice traffic is delivered. Load balancing is another scalability tool, which means traffic is load-balanced, or shared across APs, to ensure users are sent through the most available AP at any given time. The IEEE is working on a standard to allow wireless IP phones to discover all nearby APs available for service in order to utilize the most appropriate AP. Before that is available, some vendors are offering their own similar proprietary capabilities.

Quality of Service

Delays for voice should not exceed 150 ms, and given that Wi-Fi is a contention protocol, when an access point is overloaded, voice quality will suffer. QoS is required for voice traffic whether it's traversing a wired or a wireless network. In other words, you want QoS for your voice traffic over the air or over land so look for gear that offers over-the-air quality of service. Guaranteeing voice over other applications minimizes packet loss, delay and jitter that results in poor voice quality. The IEEE is working on a standard to address QoS for wireless networks, but in the meantime, the Wi-Fi Alliance has released Wireless MultiMedia (WMM) as a subset of these capabilities. Vendors are currently bringing WMM implementations to market now. WMM defines four priority levels to support varying kinds of traffic, including voice, video, best effort for data, and background traffic, in that order.

Seamless roaming

As a user walks from one office or location to another, he or she counts on roaming capabilities of the WLAN to keep the call connected. The underlying wireless infrastructure must seamlessly hand off the user to the next location and perform the necessary re-association and re-authentication with APs, while keeping calls free of interruption (this will allow a call to continue seamlessly across zones without being mistakenly dropped between zones). A security standard is under way to allow users to be pre-authenticated to neighboring APs before roaming, which will reduce the time it takes for a user's call to move between APs, and in the meantime, some wireless equipment vendors are introducing their own versions of fast-roaming capabilities.

Solid security

IEEE 802.1X authentication should be used to verify a user's identity onto the network, which will ensure unauthorized guests are not allowed entrance to use the network or gain access to confidential corporate information. Laptops and handhelds can support 802.1X authentication, and you need to make sure your wireless IP phones, which have less computational capacity, are using less processor-intensive authentication methods like MAC address or username and password.

Selecting Handsets

As discussed in Chapter 5, Session Initiation Protocol (SIP), a signaling protocol, is used for establishing a session in an IP network—from a simple two-way telephone call to a multi-media conference call session with many participants. The VoIP industry has recently adopted SIP, a RFC standard (RFC 3261) from the Internet Engineering Task Force (IETF), as the protocol of choice for signaling because of its ability to facilitate Internet applications by working with other protocols. Essentially, SIP establishes, manipulates and tears down sessions, and its main purpose is to help session originators deliver invitations to potential session participants wherever they may be. It uses URLs to address participants and SDP to convey session information and it's easy to combine SIP with other applications, like Web browsers and messaging. SIP can find and invite call invitees wherever they are, and it facilitates multi-media calls with many participants who may join and leave at will.

It is possible to use traditional cell phones and they can become an extension of your IP telephony solution—this requires no new wireless network or SIP handsets. There are also many wireless IP telephony handset vendors out there, but they don't all offer the same features. Start your search by looking at your needs first. Are you looking at wireless handsets for a manufacturing facility and therefore need a rugged handset with dust covers so they don't get dirt inside the keys? Are you a healthcare facility and need to meet safety requirements so the handsets don't interfere with hospital equipment? After you determine your general needs, next move on to what you would like to see the handsets offer. Would you like the handsets to be able to transfer calls? Would you like your employees to be able to conduct conference calls from the wireless handsets? What's your wish list on top of your needs list? These two things will bring you to a number of vendors' solutions, and then the final question you need to ask is, will it work with your IP PBX vendor's solution? Your choice will be very easy at this point—you'll likely either have just one or two vendors left from your list.

The Bottom Line

You need to be prepared to establish wireless IP telephony because your users, customer service, and the bottom line will greatly benefit from mobile VoIP. You need to approach this the same way in which you approached the wired IP telephony system: First, look closely first at your basic wireless requirements in terms of handsets; next look at what features would be nice to have; and then look at what handsets will work with your infrastructure and your chosen IP telephony system. As you go through each of these steps, the number of solutions available to you will be reduced and you'll be left with just a few options from which to choose. The next chapter will go deeper into Quality of Service plus cover Virtual LANs and MPLS.

Quality of Service

Chapter 8

ShoreTel™

TABLE OF CONTENTS

This chapter covers Quality of Service (QoS) in detail, as well as your options in terms of circuit transports. Then, it delves into the internal infrastructure and the entire process of applying QoS.

Quality of Service

Chapter 8

Quality of Service

As we drill down deeper into the details about your converged network, going over topics like handsets, security, and mobility software, you are most likely growing more comfortable with IP telephony. However, you probably still have some concerns, most notably, “How do I know my boss isn’t going to experience poor audio quality during calls?” or “How can I be certain that all of our average 200 calls are always going to get through, even on our busiest day, and when accounting is doing its weekly check run?”

The answer is Quality of Service, or QoS. This chapter will cover QoS in detail, as well as your options in terms of circuit transports, and then delve into the internal infrastructure and the entire process of applying QoS. QoS can be boiled down to three major steps: Identify. Classify. Prioritize.

Voice Must Be Heard

Quality of Service is key when it comes to IP telephony implementations. Voice traffic must get prioritized so that it’s not delayed or discarded because of interference and congestion from other traffic. At the same time, you may also have other high priority applications, such as video or other business critical data, that also need higher than normal prioritization. But voice is always going to be high up on your priority scale.

You need to consider four things that can affect voice traffic:

1. Latency (or packet delivery delay)
2. Jitter (or the variation in time between packets)
3. Packet loss (which can occur when too much traffic overflows buffers within the network causing packets to be dropped), and
4. Burstiness (when your network undergoes bursts of packet drops due to jitter)

It’s bad business to have your voice traffic burdened by any of these effects. Distance alone on the WAN circuit can cause delay, as can lower-speed WAN circuits. Delays cause call participants to start interrupting each other because they believe the other person is finished speaking. Latency should not exceed 100 milliseconds (ms) one way for toll-quality voice and must not exceed 150 ms one way for acceptable quality voice. At 150 ms, delays are noticeable by the human ear, but callers can still carry on a normal, comfortable conversation.

Jitter can cause strange sound artifacts to contaminate the voice and users will complain of degraded voice quality. Jitter has many sources: network congestion, queuing methods used in routers and switches, or network routing policies such as traffic engineering or MPLS paths used by carriers.

If your phone conversations do not sound right and callers have to keep repeating themselves or have a less than satisfactory experience when they call, they’ll start looking for other ways to communicate with your employees, or worse, they’ll start looking to another company to serve their needs—one with which they can communicate more clearly. Your IP telephony system should sound better than your previous phone system—after all, that’s why you made the switch. It’s the only way you’ll ensure that you don’t lose business because of your technology change. IP telephony should increase—not decrease—your business and your bottom line.

WAN Circuit Transports

You've probably already chosen your circuit transport method, but the IP telephony exercise may have made you re-think everything. The following descriptions are not intended to list everything about your options but rather to ignite ideas about the differences so you'll start questioning your options, which will help you make the best choice for your organization.

Leased lines

Leased lines are the most private way to go. They are also the easiest type of WAN circuits to configure guaranteed QoS. These circuits are direct point-to-point lines connecting your locations together. They can be used for data, including packetized VoIP, or Internet services.

Frame Relay

Frame Relay circuits are more economical than private leased lines because the Telco providing the Frame service shares bandwidth among many subscribers. This can reduce your costs, especially for long distance lines, but commonly reduces your guaranteed bandwidth to less than your full circuit speed. Frame Relay can guarantee bandwidth and packet delivery only if you shape your outgoing traffic to match your committed information rate (CIR). Properly engineered, Frame Relay can provide a cost-effective means of transmitting IP telephony traffic and still guaranteeing QoS.

Asynchronous Transfer Mode (ATM)

All information sent over an ATM network is broken down into discrete packets. Unlike other packet technologies, ATM employs fixed-sized packets, each consistently at 53 bytes long. This means cell delay in ATM switches is predictable and manageable.

MPLS

MPLS, like Frame Relay, is a label-switched system that can carry multiple network layer protocols. Similar to Frame Relay, MPLS sends information over a wide area network (WAN) in frames or packets. Each frame/packet is labeled and the network uses the label to decide the destination of the frame.

Figure 8.1 | MPLS works by pre-pending packets with an MPLS header, containing one or more 'labels'. This is called a label stack. Source: Wikimedia Commons.

MPLS networks can use Frame Relay, ATM or leased lines for the link layer.

Speed Your WAN

There are a number of products out there to speed your WAN. Known as "WAN optimization" products, they accelerate applications by eliminating redundant transmissions, staging data in local caches, compressing and prioritizing data, and streamlining chatty protocols. Other tools perform rate limiting to control the rate of traffic being sent to your network while more critical traffic, such as voice, is being transmitted, for instance. Rate limiting is performed by policing (discarding excess packets), queuing (delaying packets in transit) and/or controlling congestion (manipulating the protocol's congestion mechanism).

Be sure you've covered all your bases with your service provider and created a service level agreement (SLA) that you are comfortable will guarantee you acceptable service delivery. If someone downloads a huge set of files from the Internet that bogs down the WAN circuit, is that going to cause a dropped call for your CEO or is your service provider going to have the right tools in place to make sure the call stays up and the download takes a back seat to the voice call?

Internet VPN

A virtual private network (VPN) is a private network used by an organization or in many cases by a company and its partners or associates, to communicate or coordinate confidentially over a non-private network. VPN traffic can be carried over a public networking infrastructure such as the Internet. Internet-based VPNs offer the least amount of administrative control to regulate and guarantee QoS.

The Needs of High-Quality Voice

Remember that you need to minimize latency, jitter and packet loss and ensure enough bandwidth so that you deliver high-quality voice. In order to do this, you must have complete administrative control over the equipment and the circuits, end-to-end, as well as all the tools necessary to ensure your system remains up and running smoothly one hundred percent of the time. This is often impossible based on budgets and equipment inventories. An alternative is to compromise in an area that allows you to save money while giving up only so much control as to still deliver high enough quality voice where degradation is barely recognizable or where it is entirely tolerable. In all LAN/WAN environments, there will be packet congestion—it's inevitable. The key is to guarantee that VoIP packets are prioritized so that they are able to get through during those times of congestion, otherwise your QoS plan has failed.

Your spectrum of options runs from leased lines combined with feature-rich switches and routers at one end of the spectrum, to Internet-based VPNs using consumer-grade WAN circuits (DSL, Cable-modems) from separate providers with no SLA.

With the first option, you have complete administrative control over all points of congestion and have the configuration tools and features to easily identify, classify and prioritize your VoIP traffic. This is the optimal choice if you have the budget for it. Managed routers with features that you cannot control and circuits that you do not have administration control over can be less effective for your network and often require more labor to ensure configurations are correct and guarantees are being followed by the managed service provider. You have less power in terms of making forwarding decisions and changes on the fly. At the other end of the spectrum, you have cast off all control over every components, circuit and congestion point and have thrown your VoIP packets into the Internet with simply the hope that they get there, but effectively powerless to help them arrive safely and on time.

Minimizing Latency

Latency (also known as delay) is the time that it takes a packet to make its way through the network to its destination (or the time it takes the speaker's voice to reach the listener's ear). Actually, some latency is inherent and constant due to distance and the number of devices in the path. As mentioned, large latency values can cause hesitations and, therefore, call participants interrupting one another. There can be a number of factors that contribute to latency, such as propagation delays (the time it takes an electrical signal to travel the length of a conductor), queuing delays, packet forwarding delays, etc. Again, end-to-end latency should be less than 150 ms for toll quality phone calls. Here are a few suggestions for mitigating the impact of latency contributors:

- The faster the media, the less time it takes to serialize the digital data onto the physical links, and the lower the overall latency. The impact on latency depends somewhat on the link technology used and its access method. For example, it takes 125 microseconds to place one byte on a 64Kb circuit. Placing the same byte on an OC-3/STM-1 circuit takes 0.05 microseconds.
- Although some delay is unavoidable regardless of the bandwidth used, keeping the number of intervening links small and using high bandwidth interfaces reduces the overall latency.
- The packet forwarding delay is determined by the time it takes a router, switch, firewall or other network device to buffer a packet and make the forwarding decision. Among the forwarding considerations are which interface to forward the packet to and whether to drop or forward the packet against an Access Control List (ACL) or security policy. Packet forwarding delay varies depending on the function and architecture of the networking device. If a packet must be further buffered as a part of its processing, greater latency is incurred. (Source: VoIP 101, Juniper Networks.)

Jitter's Impact on Voice Quality

Jitter, a variable delay, is the time difference between when a packet is expected to arrive to when it actually arrives. In other words, given a constant packet transmission rate of every 20 ms, new packets would be expected to arrive at the destination exactly every 20 ms. Unfortunately, as Figure 8.2 shows, this is not always the case. In Figure 8.2, packet one (P1) and packet three (P3) arrive when expected, but packet two (P2) arrives 12 ms later than expected and packet four (P4) arrives 5 ms late.

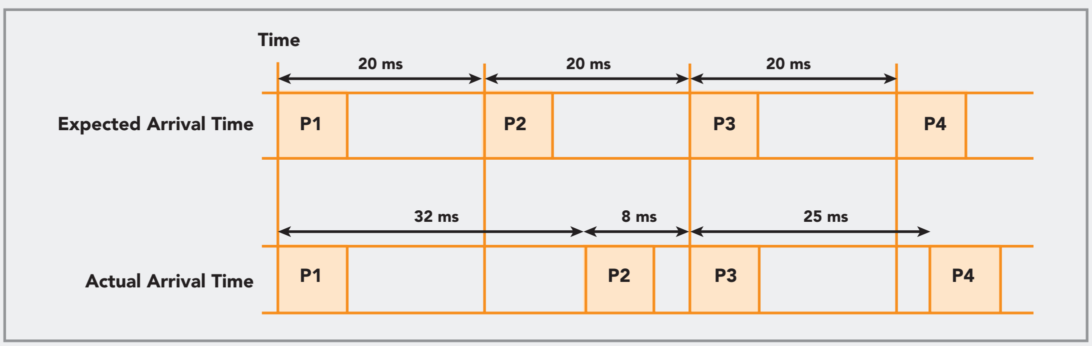


Figure 8.2 | Jitter. (Source: VoIP 101, Juniper Networks.)

Jitter is caused by congestion or other factors. Most media gateways have play-out buffers that buffer a packet stream, so that the reconstructed voice stream is not affected. Play-out buffers can minimize the effects of jitter, but cannot eliminate severe jitter. Although some amount of jitter is to be expected, severe jitter can cause voice quality issues because the media gateway might discard packets arriving out of order. In this condition, the media gateway could starve its play-out buffer and cause gaps in the reconstructed waveform. (Source: VoIP 101, Juniper Networks.)

Tolerating Packet Loss

Packet loss is often unavoidable and can occur for a number of reasons, such as the case of a router or switch overflowing, and in many instances, applications can tolerate packet loss (as in the case of a non-critical file transfer). However, most real-time applications are less tolerant of packet loss. Although packet loss is not desirable, some voice packet loss can be tolerated as long as the loss is spread out over a large amount of users. If the amount of packet loss is very small in comparison to the users and over a large amount of time, then it can be acceptable.

Applying QoS

When it comes to applying Quality of Service onto your enterprise network, it's a matter of identifying, classifying and then properly prioritizing data and voice packets. Your first step is determining the method by which you will identify high-priority packets. There are a number of options available within an enterprise network to identify and mark which packets are high priority. These methods include VLANs, Differentiated Services Code Points (DSCP), also called DiffServ, Type of Service (ToS) bits, IP Precedence, 802.1p markings, Layer-3 IP-address, Layer-4 source & destination ports, etc. Once you've determined which method to use throughout your corporate network, you will use this method to identify, and possibly re-mark, each high priority packet. (Keep in mind, your IP telephony vendor may mark them with one method, and you may choose to re-mark the packets with a different identification method.)

Once each high-priority packet has been marked with your corporate standard (for instance, DiffServ), then at egress (as the packet leaves a piece of networking gear such as an Ethernet switch or router), it needs to be prioritized above other packets. Keep in mind there are different levels of priority as well as different queuing methods, so if your organization is a hospital, you will likely have healthcare applications, such as patient records and networked images transfer applications, assigned a higher priority along with voice traffic.

Identification Methods

The following is a list, although not exhaustive, of identification methods you may choose to utilize for prioritizing voice traffic. Again, your IP telephony vendor may choose one method and you may choose another for your corporate standard, in which case you will be re-marking each packet with your method choice.

DiffServ or ToS

Layer 3 QoS using DiffServ or Type of Service (ToS) bits is a system of identifying IP packets by assigning values within the layer 3 IP header. Once identified, traffic can be classified into groups so that QoS policies can be applied. For example, maybe Web access needs to be reasonably responsive but acceptable e-mail response time can range from seconds to minutes. On the other hand, voice traffic (IP telephony) and IP videoconferencing require a much higher level of priority. The type of end-to-end QoS you choose to implement will depend on what type your routers and IP telephony solution support.

DiffServ and ToS add state information to each packet—allowing the network equipment to identify different service flows and direct queuing and forwarding treatment appropriate to the service requirements. This enables routers to identify voice packets and mark them for higher priority treatment over less sensitive packets. With DiffServ or ToS, each router on the network is configured to differentiate traffic based on its class and each traffic class can be managed differently, insuring preferential treatment for higher-priority traffic on the network.

802.1p

802.1p is a specification that gives Layer 2 switches the ability to identify and prioritize traffic. It works at the media access control (MAC) framing layer (Layer 2) of the OSI model. Eight classes are defined by 802.1p, which uses the priority fields within the packet's VLAN header to signal the switch of the priority-handling requirements.

VLANs

A virtual LAN, known commonly as a VLAN, is a method of creating logically independent networks within one physical network. A few or many VLANs can co-exist within such a network. For instance, a small 50-person grocery store can have 10 VLANs dedicated to different departments of the store and one VLAN for information technology. A hospital could literally have hundreds of VLANs to segregate different staff members, doctors' groups, departments, and labs. Administratively segregating and separating people and departments helps to reduce traffic on each VLAN so that each segment is performing optimally, aids in ensuring confidential information is accessed only by authorized personnel, and ensures that latency and bandwidth-sensitive traffic, like voice, is given priority. Often voice traffic is given its own VLAN (or multiple VLANs).

Prioritization Methods

Once you've decided how you're going to tag your high-priority packets, next you have to determine your prioritization method. Here are just a few.

Weighted Fair Queuing

Weighted Fair Queuing (WFQ) allows traffic flows to share link capacity but provides prioritization for small, time-sensitive traffic flows. The advantage is that a large flow will not clog the pipe or create lengthy delays for other smaller flows. WFQ is used in routers and switches that forward packets from a buffer that works as a queuing system where the packets are stored temporarily. Packets are essentially waiting in queues in buffer space, while WFQ estimates which packet flow will be "fastest" (the one with the minimum number of packets) and transmits those smaller, time sensitive packets ahead of the larger, delay-tolerant packets.

Priority Queuing

Priority Queuing supports multiple fixed-length queues from high to low, servicing the highest queue first, then the next-lowest priority and so on. If a lower-priority queue is being serviced and a packet enters a higher queue, that queue is serviced immediately. While good for important traffic, it can lead to queue starvation.

Custom Queuing

Custom Queuing is designed for environments that need to guarantee a minimal level of service to all protocols. It allows a customer to reserve a percentage of bandwidth for specified protocols. Customers can define multiple output queues for normal data and additional queues for system messages such as LAN keepalive messages. Custom Queuing can guarantee that mission-critical data is always assigned a certain percentage of the bandwidth, but also assures predictable throughput for other traffic. (Source: Custom Queuing and Priority Output Queuing, Cisco)

Class-Based Weight Fair Queuing (CBWFQ)

Weighted Fair Queuing classifies traffic into different flows based on layer 3 and layer 4 information, such as IP addresses and TCP ports. However, WFQ has some limitations—it's not scalable as traffic increases, and native WFQ is not available on all high-speed interfaces. WFQ also doesn't provide as much granular control as is often needed. CBWFQ provides a solution to these limitations. CBWFQ gives an administrator more control over what types of traffic classes are assigned to each queue and what unique prioritization methods each queue should be assigned including bandwidth, priority, queue size, reserved bandwidth, etc. The bandwidth you assign to a class is used to calculate the "weight" of that class. The weight of each packet that matches the class criteria is also calculated from this. WFQ is applied to the classes (which can include several flows) rather than the flows themselves. (Source: Understanding Class Based Weighted Fair Queuing on ATM, Cisco)

The Bottom Line

Quality of service is a must when it comes to real-time applications like IP telephony. Applying QoS is a matter of identifying, classifying (or marking), and then prioritizing voice packets. This chapter has outlined how to go from the outside (service provider) to the inside (infrastructure), from choosing the circuit transport to choosing prioritization methods. There are many options that can suit your needs and it will be a matter of discussing the choices with your service provider, integrator and/or colleagues. The next chapter will cover other options that are available to you from various service providers, including some new options being offered.

Resources:
http://www.wikipedia.com
http://www.networkworld.com/links/Encyclopedia/index.html
http://www.juniper.net/solutions/literature/white_papers/200126.pdf

Outsourced IP Telephony Options

Chapter 9

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TABLE OF CONTENTS

This chapter discusses outsourcing IP telephony options and compares the benefits of hosted IP telephony vs. managed IP telephony vs. IP telephony access.

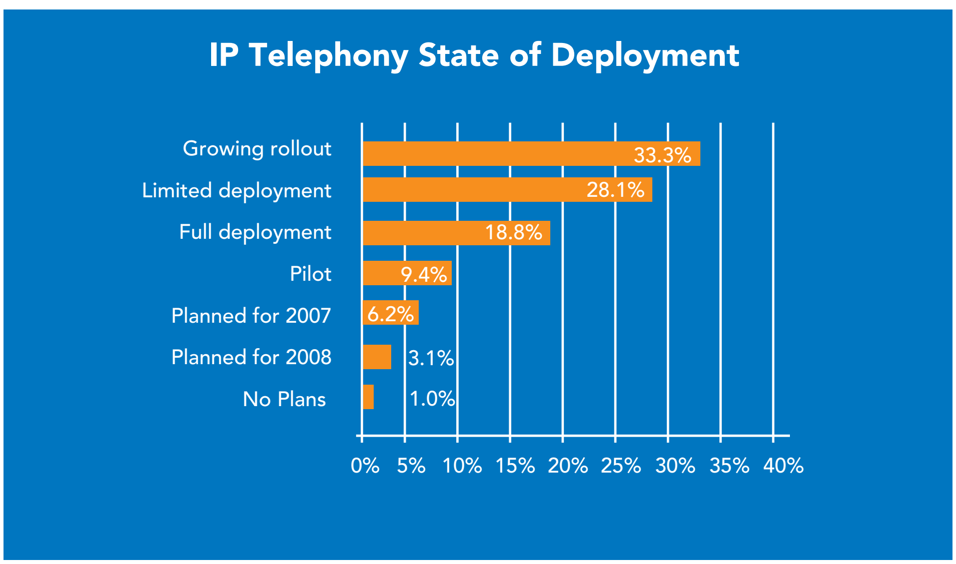
Outsourced IP Telephony Options

Chapter 9

Outsourced IP Telephony Options

Up to now, we've covered in-house IP telephony systems. Because you have full control over systems that you own outright, you will find they are the most flexible, reliable, and scalable. With fully-owned systems, you're also able to make the changes as soon as you need to make them—no waiting for a provider's schedule to free up, no fees to be paid, and no miscommunication risks. However, hosted IP telephony is starting to gain some popularity. Savings are attracting businesses to hosted IP telephony solutions, especially small and medium-size businesses (SMBs) and those with multiple sites and organizations with highly mobile workforces. As a matter of fact, according to research firm In-Stat, hosted IP telephony will continue to experience dynamic growth with projected revenues exceeding $2 billion by 2010. The firm's research also indicates that U.S. hosted IP telephony seats in service will continue to grow steadily to over 3 million in 2010, up from 373,000 in 2006.

While outsourced IP growth is impressive, it's not as impressive as the growth of IP telephony overall. For the past four years, Nemertes Research, one of the few research firms focused specifically on IP telephony, has been tracking IP telephony deployments. In its March 2007 report, "Building the Successful Virtual Workplace-VOIP Review: Products, Services, Architecture," the firm reports that year after year, a growing number of organizations have moved away from TDM technology to IP technology. The report draws from detailed interviews with 120 IT professionals from 99 companies. The benchmark determined that this year, only 1% of the benchmark participants have no plans whatsoever for IP telephony, down from 6% last year. While this doesn't mean every organization is using IP telephony, it does mean that nearly all organizations are doing at least something with the technology, whether evaluating vendors, running a pilot test, or deploying a IP telephony system. Figure 9.1 indicates participants' response to the Nemertes Research question, "What is your state of IP telephony deployment?"



Source: "Building the Successful Virtual Workplace–VOIP Review: Products, Services, Architecture," March 2007

Figure 9.1 | Nemertes Research Benchmark Question

Outsourced IP Benefits

In addition to the costs saved when organizations benefit from fixed rate bundles for local, national and international calls, outsourced IP offers improved effectiveness with features like voice mail and unified messaging. Outsourced IP telephony also offers the flexibility enjoyed by in-house IP telephony, in that employees can log in to any phone make and receive calls as if it were his or her desk phone. Voice mail can be retrieved remotely from any phone, including cell phones, and it can be retrieved via e-mail. Costs are more predictable because of price bundling, and quite often the system is more reliable because call re-routing for business continuity and disaster recovery is automatic. For a monthly fee, the hassle of managing your own telephony solution can be offloaded and a provider deals with the headaches plus the monitoring and management of your IP telephony system, in line with the service level agreements you've discussed.

Now SMBs have access to a cost-effective, feature-rich alternative to on-premises IP PBXs and traditional analog phone systems. Hosted IP offered by service providers specifically targeting SMBs offer modern features, scalability, and ease-of-use designed to give small and mid-sized businesses the competitive edge they've been looking for when it comes to their telephony systems.

You've Got Choices

Service providers understand the importance of being a one-stop shop for customers, providing data, voice and video services to customers, which is why they're building tremendous multi-service networks to support it all. Customers now have all kinds of choices—they're no longer locked into voice services only from their incumbent local exchange carrier (ILEC). The competitive landscape has changed and competing service providers (SPs) offer multiple services over a converged network. Companies can now look to a variety of sources for IP telephony services, including ILECs, competitive LECs (CLECs), Internet SPs (ISPs), and Value-Added Resellers (VARs). The competition has opened up the opportunity to all kinds of business, large and small, and provided competitive pricing and advanced capabilities to make IP telephony the right choice for many companies.

IP Telephony Features Offered by Service Providers

- 3 or 4-digit dialing
- 3-way calling
- Automatic call distribution (ACD)
- Authorization codes
- Auto attendant
- Call forwarding
- Call park/call pickup
- Call waiting
- Caller ID
- Conference bridge
- Direct inward dialing (DID)
- Distinctive ring tones
- E911 compliance
- Follow me
- Hunt Groups
- International calling
- Music on hold
- Operator console
- Voice mail
- Web-based management portal

Managed IP Telephony

In the managed IP telephony model, the company owns the IP PBX, either on-site or in the provider network, while the carrier provides oversight and maintenance on it and offers bundled services (caller ID, auto-attendant, call redirect, and other voice-related applications). The service provider also supplies the customer premise equipment (CPE) for packetizing voice before it enters the wide area, and also includes Service Level Agreements (SLAs) to cover Quality of Service (QoS) and support. Managed IP telephony is a great option for businesses that want to try out IP telephony but aren't ready to manage the system.

According to Laurie Shook, Verizon Business' director of managed IP telephony, when it comes to managed IP telephony, companies should:

- Evaluate service provider capabilities in terms of breadth of services and flexibility of offerings.
- Ensure the vendor is financially stable and committed to the business over the long haul.
- Determine whether resources are available when and where they're required.
- Look for a service provider that will build upon the existing hardware and software investment.
- Identify the scope and scale of service provider responsibility.
- Tour the company's network management facility and meet the people who will monitor the network.
- Ask about employee and site certifications.
- Select a service provider with built-in system redundancy.
- Obtain fully documented service resolution procedures.
- Consider vendors that are committed to continued investment in network operations and systems integration.

(Source: SearchVoIP.com, http://searchvoip.techtarget.com/originalContent/0,289142,sid66_gci1241254,00.html)

Hosted IP Telephony

Hosted IP PBX delivers IP telephony services to subscribers with cost reductions and improved business processes and customers do not need to be tied to the switch. The architecture is similar to a Centrex system or KTS (Key Telephone System), except a service provider rather than a local phone company provides switching along with the gateways to the PSTN. It is fully outsourced, and the customer utilizes the broadband IP network for voice and data without having to own or manage the switch. The only CPE (Customer Premise Equipment) necessary for IP telephony are the phones or converters if analog phones are used.

With hosted IP, all IP telephony components, from media gateways and switches to application servers, are located at the service provider location (or data center). This model is different from a managed IP PBX model, where the equipment can reside either on the customer premises or in the service provider data center. In the hosted solution, the equipment supports many customers, whereas in the managed IP scenario, the IP PBX is dedicated to the use of one customer (not shared across customers).

The hosted solution is based on flat fees usually costing approximately $60 to $70 per month per station or user. In addition to an access line and IP telephony dial tone, the fee usually includes an a la carte selection of advanced features and an unlimited number of local and long distance minutes. For a multi-site organization, hosted IP telephony allows the sites to have the same capabilities, an abbreviated dialing plan (3 or 4-digit dialing, for instance), hunt groups, and uniform voice mail and auto-attendant for outside callers. The service also usually provides a web-based service management interface for IT administrators to manage moves, adds and changes (MACs) or to perform other tasks.

IP Telephony Access Service

Because many businesses already have their own PBX/IP PBX equipment and are looking to capitalize on that investment and keep network ownership costs down, IP telephony access service can help them do that. IP telephony access also serves as an introduction to converged network services. The service provider offers its customers whatever IP telephony capabilities it has in the data center— typically an abbreviated dialing plan, IP telephony VPN, hosted voice mail, and possibly a few others—all bundled together. The customer owns the PBX system so the service provider offers IP telephony through the same loop it's already supplying to the customer (whether T-1 for large customers or broadband for smaller customers).

Managed IP Telephony	Hosted IP Telephony	IP Telephony Access
• Customer owns equipment but service provider maintains and oversees • Customer has full benefits of service provider's IP telephony investment • No need to train staff – customer benefits from service provider's extensive staff while having immediate access to CPE • Savings with bulk minutes • Features bundled or a la carte so customers buy what they need	• Fully outsourced solutions • Customer has full benefits of service provider's IP telephony investment • No need to train staff – attractive to SMB • Savings for inter-office communications because service providers typically offer bulk minutes • Features bundled or a la carte so customers buy what they need • Web-based interface makes administration easy	• For customers that manage CPE • Deal with one service provider for WAN, local and long distance, voice and data • Service provider supports connectivity to PSTN, primary rate interface (PRI) not needed at each location • Bulk minutes typically reduce costs • Service Provider has IP telephony infrastructure, which augments CPE capabilities with advanced features

Top 10 Questions To Ask [a Service Provider]

Before you sign a contract you should make sure you know the answers to these 10 questions.

Title, publication date coming shortly

1. What is the contract termination policy? Can I get out early and what penalty is there? Are there other termination costs?

   Unfortunately, providers and particularly their sales teams will often try to lock you into long-term contracts. This is common practice throughout the communications industry and you probably can't avoid it, but do your best to remove arbitrary and excessive early-termination penalties.

2. What startup costs are there beyond setup and equipment fees? Besides advertised and quoted basic equipment, what else will I REALLY need? Do I need to buy phones? Will extra servers/cards/add-ons be needed beyond the base cost to actually meet my usage requirements**?**

   It's easy to get sucked into a good-looking deal with switching and PBX equipment priced at attractive points like $1000 – but that is often for a minimal configuration and you find out later that you need add-on servers or cards or other extra equipment – and don't forget phones – some quotes will include them and some won't.

3. What day-to-day usage costs are NOT covered by my service plan? What are the rates for international calls for example?

   Make sure there are not add-on fees for important features – some providers charge more for conference calling, others for different forms of long-distance, still others for some of the advanced features. Make a quick model of your actual communications usage and ask about all the items on the list so you can forecast prices accurately.

4. Can the system, as it comes, handle outbound and inbound faxes easily? Can I just plug a fax machine in or do I need special equipment?

   Faxing is a hidden gotcha of IP telephony. Many older systems can't handle faxes. Others require a special faxing module. The bottom line is that just because you have a phone line with IP telephony it doesn't mean you can plug a fax into it. Make sure you're covered.

5. Do I need add-ons or extras to handle old-style analog phones I already have or that remote or branch offices already have installed?

   This is an issue that can save you some money. If you have offices that already have extensive modern but analog phones, some provider systems will work with them as transparently as with more advanced SIP and IP telephony phones. That can save you as much as a couple of hundred dollars per phone.

6. How does the system handle remote and mobile workers – whether temporary or permanent? Will the experience be the same for a telecommuter in a rural area as it is for someone at head office? How about when I'm on the road? Are there any services or features to handle that?

   Obviously, if you don't have mobile or remote requirements this isn't a concern right now, but in time it may be and if you DO have remote and mobile employees then you need to find out what the provider can and cannot do for you. Some providers can handle any mobile or remote phone almost as easily as an extension in the main office. But others cannot. However, creative solutions might save you money even going over to a system that can handle remote employees. For example, getting a single line for a remote employee on a different plan and then using your system's call-forwarding features might end up being more cost-effective.

7. If I estimate my requirements [incorrectly] and need a major upgrade, what are the additional costs for upgrading?

   There isn't too much you can do here – mis-estimating requirements are going to be a problem no matter what. But you can minimize the pain with a provider that is willing to work with you and provides good service.

8. How do you guarantee QoS? If I have issues how do I get support, who do I contact and how fast will it happen? When there is a complex issue and my network equipment supplier, my broadband supplier and you are all pointing the finger at each other, will you step up to solve my problem?

   These are all questions to ask the provider directly. If they cannot satisfy you in this area, then look elsewhere. Your communications system is vital to your success – so don't put yourself in a position of being held hostage by an incompetent provider.

9. What about emergency services – do you provide full 911 or E911 services? Will dispatchers know my location automatically? How about remote workers?

   This is a flat-out requirement that many IP telephony providers don't like to talk about. If they don't have a solution right now, they should have one in the pipeline. And that leads us to the last question...

10. How do I know you are going to be around in two years, let alone three?

    Communications in general is an industry of consolidation and aggressive competition. Find out if your provider is stable and make sure your contract is binding even if your provider gets acquired by another.

Source: VoIPNEWS.com http://www.voip-news.com/news/questions-for-voip-service-240406/

The Bottom Line

Outsourcing your IP telephony services may not be a fit if your organization is a mid-sized enterprise and you want full control of your IP telephony system, including control of your equipment and the ability to make changes to it on the fly, including moves, adds and changes (MACs). However, if your organization is a geographically scattered SMB struggling to look to the outside world like a larger unified entity, then a hosted IP solution may be the way to go until a time in the future when you're ready to bring an IP telephony system in-house. With a hosted IP telephony system, you can streamline communications, boost productivity and customer responsiveness, and reduce your communications costs. You can improve your bottom line and increase your customer satisfaction.

The next chapter will cover system management, from configuration and troubleshooting and MACs to sophisticated reporting tools that can help you improve your overall business planning.

Ease-of-Use and Management

Chapter 10

ShoreTel™

TABLE OF CONTENTS

This chapter covers the need for IP telephony solutions to be easy to manage, implement and administer.

Ease-of-Use and Management

Chapter 10

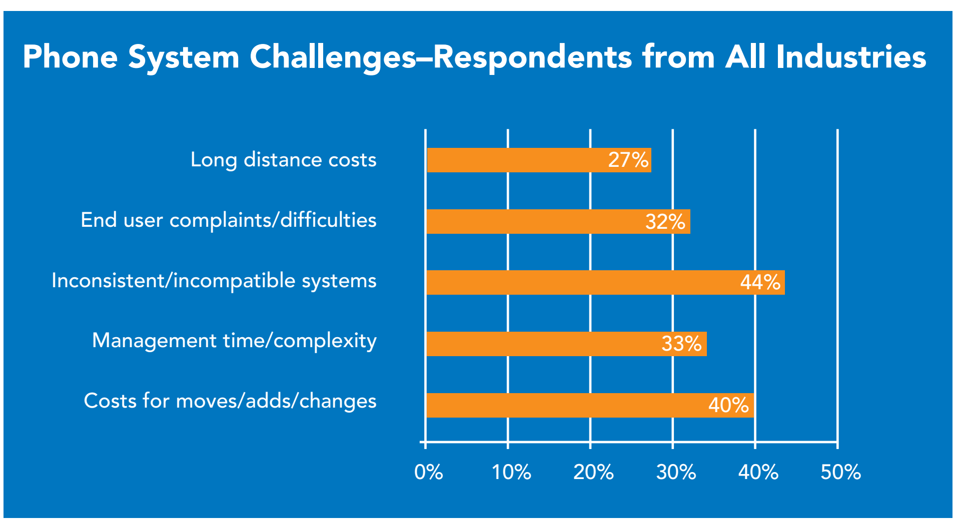
Ease-of-Use and Management

In order to lower the total cost of ownership (TCO) of anything, you must address the human resources required to operate, manage, maintain, and support it. Ease of management is especially vital to TCO when choosing an IP telephony system. The network convergence you've already conducted allows you to stop maintaining parallel networks for voice and data and instead have a single, converged infrastructure that leverages investments and streamlines administration and management. Your chosen IP telephony solution should take this even further, simplifying the end-to-end management of your IP telephony system, from device to network monitoring and analysis. An IP telephony solution that's truly easy to implement and manage can pay for itself very quickly–often in a few months.

What IT Managers Want

In July and August 2006, Computerworld invited online visitors to participate in a short survey on IP telephony, and the results were published in an October report. (Source: "The Financial Sector Rates Importance of IP Telephony Features, Management, and Applications," Research Conducted by Computerworld, October, 2006.) While the report focused on the financial sector, it also included information covering all industries with data collected in March 2006.

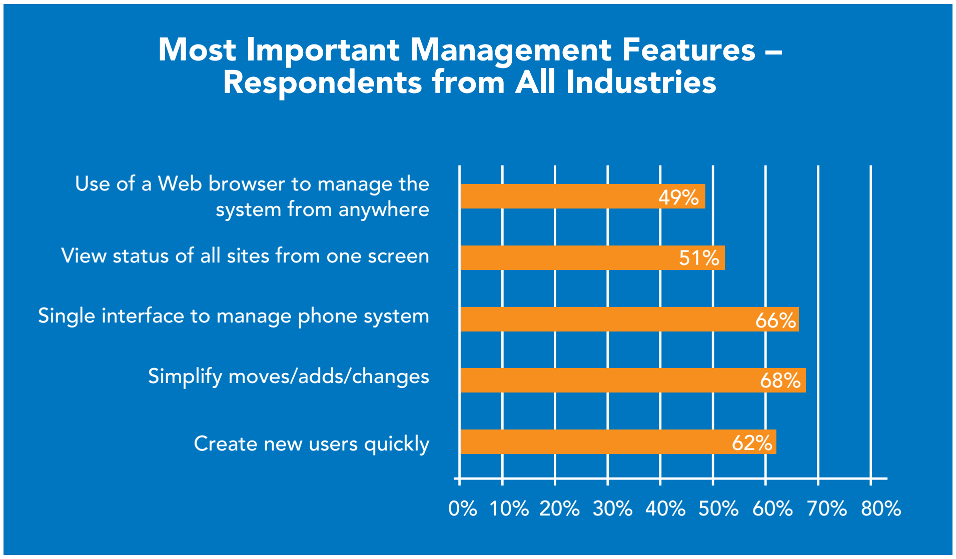
Respondents were qualified through a screening process conducted by Computerworld Research as being personally involved in evaluating IP telephony systems for their organization, and data was gathered and tabulated independently by Computerworld Research. Results indicated that managers of IP telephony systems identify a lot of challenges associated with phone system management. Figure 10.1 lists the top overall challenges indicated by respondents from all industries: Inconsistent/incompatible systems and costs for moves/adds/changes (MACs) were the two top challenges.



Source: "The Financial Sector Rates Importance of IP Telephony Features, Management, and Applications," Research Conducted by Computerworld, October, 2006.

Figure 10.1 | Phone system challenges according to respondents—all industries (data collected for all industries in March 2006)

When asked which management functions were most important, results indicated that IP telephony managers are basically looking for simplification. First on the list was the simplification of MACs, and close behind was the desire for a single management interface, followed by new user creation, status view from one screen, and web browser management capabilities from anywhere on the network (see Figure 10.2).



Source: "The Financial Sector Rates Importance of IP Telephony Features, Management, and Applications," Research Conducted by Computerworld, October, 2006.

Figure 10.2 | Most important management features according to respondents–all industries (data collected for all industries in March 2006)

Ease-of-Implementation and Administration

What this means is you probably have similar concerns about IP telephony management as many of your peers. Your IP telephony solution should be easy to implement as well as easy to manage. It's important that you are able to easily add new locations, branches, or remote offices if expansion is in your business plan through regular growth or acquisition. Generally, voice platforms that have been designed from the ground up to be IP telephony platforms have management interfaces that are easy to use and make MACs quick and easy. Generally, administrators with basic networking skills can easily be brought up to speed on purpose-built IP telephony solutions. It's often a different story with analog phone systems that have been jury-rigged to "look and feel like" IP telephony.

In general, it is very simple to add a new user to most IP telephony systems. It's usually just a matter of typing the user's name into an easy to use form and editing or accepting various default fields. With these simple steps, the user is given an extension, direct inward dial (DID) number, if the IP telephony vendor offers the feature, and a voice mail box. Depending on the IP telephony solution, various features are updated, such as automated attendant, dial-by-name and number and online directories.

Easy Toll Bypass Set-Up

Despite the fact that long distance rates have dropped dramatically in recent years, there are still savings to be had in toll bypass. However, the process of setting up the network call routing for toll bypass must be easy to follow and simple to carry out in order for the savings to outweigh the management overhead. In IP telephony implementations built on legacy voice and data platforms, automatic peer-to-peer exchange of newly entered routing information is not automatic. This routing information must be entered into routing tables—and possibly even into each individual PBX or physical router. Small, localized groups 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, a process that should occur transparently to the caller. With a distributed system, you should have access to an intuitive graphical interface to replace those routing tables, and toll bypass routing will take place intelligently and automatically.

Simplicity for End Users

In many cases, most companies that have already deployed IP telephony have not even scratched the surface of utilizing all of the capabilities, usually because the features are too hard to implement. If something takes time away from your job as IT manager to train end users and re-train them on phone features and other system features (such as logging in from another phone, or using the follow-me feature), these training costs (time and budget) outweigh the benefits. You want a system that's "self-service" for end users, and one that offers a very intuitive set of features. Setting up a conference call should be easy for anybody to handle—you should not be brought in to set up something every user should be able to utilize on the fly. Otherwise, the benefit is lost in the complexity and cost.

The best IP telephony systems have a single intuitive user interface that enables non-technical users to help themselves to such functions as call management, setting up conference calls, logging in to work from a remote location, and managing their own integrated desktop communications. When IP telephony is implemented properly, end users should be taking self-service to the highest level and IT managers or administrators should only need to step in when new users must be added or permanent changes have to be made. When this self-service model is followed, burden is lifted from the IT or network staff, who can then spend their time on other revenue-producing or business improvement tasks.

Benefits to the Contact Center

In addition to saving time and money when it comes to IT staff budgets, IP telephony should reduce contact center operating costs. The cost savings associated with an IP contact center can quicken the return on investment (ROI). First there are the savings associated with having the contact center built on IP telephony because you eliminate T1 charges between sites where contact center personnel reside. Management is also centralized for the call center, even though your agents may sit in multiple locations around the globe. You can manage the entire statewide, countrywide or global contact center from your own location.

When it comes to the contact center for overall reporting, your IP telephony system should offer as high-level and in-depth information on call handling as you need to maximize your organization's staff. Contact center reporting applications should enable managers and supervisors to generate historical statistical reports to assist in evaluating past activities and planning future actions. Some vendors offer predefined reports that can be generated as they're described, or you can alter the fields and create customized reports for various departments and/or executive scrutiny. The generation of these reports should be as easy as the entire system is to use via drag-and-drop commands from a centralized desktop.

Built-In System Monitoring

Your IP telephony system should provide you with a comprehensive monitoring application that collects network utilization and error information, as well as device performance statistics, to identify potential problems that may impact your IP telephony call quality. This can help you quickly identify issues and proactively make changes before the issues become problems. For instance, you should be able to view current network as well as historical utilization of any monitored link, and historical packet loss information so you can make infrastructure tweaks. Whenever new devices are added, the monitoring application should detect these changes and make automatic configuration updates, and it should scale easily to provide full functionality with minimal maintenance, no matter how large your network grows. Network analysis is a must-have for your IP telephony system so that you can proactively maintain an optimal network and reactively make changes so that call quality does not degrade.

Centralized vs. Distributed: Which is Right for You?

Should your IP telephony system be centralized or distributed? Centralized solutions have a few clusters of IP PBXs coupled with site-survivable gateways. A distributed model is one in which system intelligence is pushed out to all locations equally, which automatically provides redundancy. Each of the approaches has its pros and cons, and your choice will depend on your organization's needs and resources. Irwin Lazar, principal analyst and Collaboration and Convergence program director with Nemertes Research, describes each approach in a special report entitled "IP Telephony System Manageability: Architecture Matters."

A centralized IP telephony architecture includes telephony servers within data centers and large facilities, and branch gateways at remote sites. Call control functionality is centralized within the data centers and large sites. The branch gateways serve two purposes, according to Lazar: as access to the public switched telephone network (PSTN) for calls that do not ride on the network, and disaster recovery in the event of connectivity loss or failure of the centralized IP telephony servers.

A centralized model can support tens of thousands of users on a limited number of servers. In addition to reduced infrastructure costs, a centralized approach reduces management costs because the number of devices is limited. Servers are also centralized at data centers and locations where there are already IT staffers on-site available for administration tasks, further reducing management costs. Nemertes interviewed enterprise IT executives for a benchmark study and learned that the primary reasons for adopting a centralized IP telephony model were "the desire for centralized control of IT resources, as well as the desire to reduce the number of IP-PBXs." (Source: IP Telephony System Manageability: Architecture Matters, by Irwin Lazar, Nemertes Research, February 2007)

Lazar lists several characteristics of the centralized approach which suggest that it may be the best fit for very large enterprises:

- Centralized systems require a lot of up-front resources, to build the solution and to expand the infrastructure to support all sites.
- Centralized systems require separate servers for voice mail, unified messaging, and conferencing, which add to system cost and complexity.
- Centralized systems rely on the data center or large site call servers to control setup for all calls—even calls between users in the same physical location—which increases WAN traffic and resiliency requirements.
- Organizations using a centralized approach often require costly monitoring and management tools to keep the solution up and running.

A distributed IP telephony model has a call server (or servers) in each location, and application intelligence (voice mail, unified messaging, etc.) is distributed, as opposed to being contained in a centralized set of servers. This eliminates the single point of failure for equipment and applications, as well as reducing overall complexity. Distributed systems are also highly scalable, where IP PBXs can easily be added and configured to find peers.

Because each location has its own fully-functional PBX in a distributed architecture, WAN utilization is minimized. Only calls that leave the location generate WAN traffic; calls within the same facility are handled by the local PBX and do not need to utilize the WAN. Therefore, the distributed model often eliminates the need for more bandwidth and the need for a highly-available WAN architecture.

Lazar writes that "Nemertes Research has validated the cost savings potential of the distributed model. In research conducted for [the firm's] 2006 benchmark, 'Convergence and Next Generation WANs," [Nemertes] found significant cost savings in start-up time required to deploy IP telephony solutions for distributed solutions, such as ShoreTel's ShoreGear." (Source: IP Telephony System Manageability: Architecture Matters, by Irwin Lazar, Nemertes Research, February 2007)

Many smaller organizations interviewed by Nemertes Research appreciate the distributed model because it requires less of an upfront investment than the centralized approach. Respondents also cited ongoing cost reductions as a big factor in their decision, as well as the comfort factor because of the fact that there is less reliance on the WAN. (Approximately 40% of the enterprises Nemertes interviewed used a distributed approach.)

Lazar sums up the architecture decision by suggesting enterprises evaluate both approaches closely to determine which model best meets their goals, noting specific criteria that should be considered carefully:

- Operational start-up costs (planning, installation, troubleshooting), including outsourced and internal resources.
- Ongoing operational costs (software fees, maintenance plans, resources dedicated to management).
- Management and monitoring tools necessary for the solution.
- Business continuity plans.
- Expertise of resellers/integrators.

(Source: IP Telephony System Manageability: Architecture Matters, by Irwin Lazar, Nemertes Research, February 2007)

The Bottom Line

Lowering TCO requires simplicity. Your IP telephony system should be as easy for employees to use as possible, requiring fewer interventions by and support calls to the IT department. Initial training can be expected, of course, but beyond an initial training session, users should be able to easily master features on their own with very little effort. It should also be easy to manage and help you make the most of your organizational resources. When voice platforms have been built from the ground up, this is generally what you get—a seamless system that is reliable, easy for users to learn and navigate on their own, and easy to manage.

Determining the right architecture (centralized or distributed) comes down to your organizational needs, resources, and business requirements. Distributed systems are a fit for organizations that have a lot of branch and remote offices and companies that grow rapidly and need to scale easily. The centralized architecture is geared to large enterprises supporting tens of thousands of users with large IT staff centrally located to support the system.

The next (and final) chapter will highlight some IP telephony implementations as they've been covered in the press. Chances are, you'll find an example that closely resembles your own situation so you can make comparisons and learn from peers who have already implemented IP telephony.

Proof Points

Chapter 11

TABLE OF CONTENTS

This chapter highlights proof points of successful IP telephony implementations as covered in the trade press. These case studies focus on different size organizations within different vertical markets.

Proof Points

Chapter 11

The bottom line is that IP telephony can help your organization gain a competitive advantage, boost employee productivity, and enhance customer service. The real proof of these IP telephony benefits lies in the budget, resource and time savings, as well as the actual end user experience.

This chapter will highlight successful IP telephony implementations as covered in the trade press. The case studies will highlight different size organizations as well as companies within different vertical markets. Chances are you'll find an organization similar to yours.

- Education and Non-profit: Tiger Woods Foundation & Tiger Woods Learning Center
- Financial: Community Bank & Trust
- Government: City of Sioux Falls, South Dakota
- Healthcare: Visiting Nurse Association of Boston
- Professional Services: National Travel Systems
- Retail: AND 1

Education and Non-Profit: Tiger Woods Foundation & Tiger Woods Learning Center

The Tiger Woods Foundation (TWF) was moving to a new building and needed to replace an outdated PBX-based telephone system in order to improve internal and external communications and enhance employee productivity. A year earlier, the Foundation had installed a ShoreTel IP telephony system in its new Tiger Woods Learning Center (TWLC), so the choice became clear: ShoreTel was chosen for the Foundation and both organizations are enjoying tremendous benefits.

- The TWF and TWLC use ShoreTel's ShoreWare Auto-Attendant, which provides 24-hour automated call answering and routing to improve service and enhance the organization's image for inbound callers.
- The new system allows employees and volunteers to work from anywhere and still remain in touch via ShoreTel's Personal Call Manager capabilities, including the Find Me feature.
- ShoreTel's ShoreWare Director, a browser-based management interface, allows IT staff to easily manage every site from anywhere on the network, including moves, adds, and changes (MACs), as well as voice mail, automated attendant and desktop applications.

The following is a description of the Tiger Woods Foundation and Tiger Woods Learning Center's experience as described in *Internet Telephony*

In 2005, the Tiger Woods Foundation was moving to a new building and needed to replace an outdated PBX-based telephone system in order to improve internal and external communications and enjoy productivity boosting benefits. A year earlier, the Foundation had installed a ShoreTel IP telephony system in its new Tiger Woods Learning Center, so the choice became clear. ShoreTel was chosen and the Foundation had an IP telephony system similar to the Learning Center.

According to *Internet Telephony*, not only do non-profits "have requirements similar to other businesses, but they also must contend with very small budgets and cannot afford to waste time or money on less than optimal solutions."

“ShoreTel is one communications solutions provider that has made a name for itself providing easy to use enterprise class solutions to the SMB market at an affordable price point — precisely what the Tiger Woods Foundation (TWF) and Tiger Woods Learning Center (TWLC) were looking for when they chose ShoreTel for their new telephony solution. With ShoreTel, TWF and TWC have the reliability and functionality to effectively manage their communications and create a more efficient working environment.”

“The new system, according to TWLC’s executive director Kathy Bihr, is both flexible and easy to administer, allowing the organization to focus on its work rather than system administration and training. The endpoints, themselves, are easy to install and understand — intuitive is how Bihr described them — so that new users are quickly and easily able to integrate the benefits of IP technology into their everyday lives, including mobile staff.”

“In addition to field workers, on-site teachers also win, as they are able to move across campus, yet remain close to their extensions using ShoreTel’s Extension Manager. Additional customization features further enhance the communications capabilities of the organization, allowing them to spend less time worrying about how to communicate and more time actually communicating, leading to less time communicating and more time focused on the task at hand.”

Source: “Tiger Woods Foundation Looks to Break Par with ShoreTel IP Telephony,” by Erik Linask, Associate Editor; Internet Telephony, January 23, 2007

For the full story, visit http://www.tmcnet.com/enews/e-newsletters/Internet-Telephony/20070123/4679-tiger-woods-foundation-looks-break-par-with-shoretel.htm.

For the ShoreTel case study on Tiger Woods Foundation/Tiger Woods Learning Center and additional ShoreTel education case studies, visit http://www.shoretel.com/solutions/industry/education.html.

Financial: Community Bank & Trust

Community Bank & Trust was growing rapidly and was faced with different telephone systems in each of its 11 branches. The bank was unable to centralize communications and there was no consistency across the organization, so an evaluation was conducted and the bank chose ShoreTel for its IP telephony solution. The following is a list of the many benefits Community Bank & Trust enjoys with its ShoreTel IP telephony system.

- With ShoreTel, management is simplified and now handled in-house, eliminating the time and cost associated with an outside service provider.
- With features like Workgroups, Hunt groups, and Find Me, a live person is always within reach of every caller who needs one, improving customer service and satisfaction.
- Community Bank & Trust can bring on a new branch in just one day with ShoreTel.

The following is a description of Community Bank & Trust's experience as described in *FinanceTech*

According to a FinanceTech article, "There are few things that frustrate customers more than complicated phone systems. And following growth through acquisitions, Community Bank & Trust had different phone systems -- none of which shared the same feature set -- for each of its branches. Instead of transferring calls from one branch to another, employees had to tell customers to hang up and dial another branch, notes Sheila Genske, IT trainer and telecom specialist for the Sheboygan, Wis.-based bank."

"Further, Community Bank & Trust ($515 million in assets) needed the assistance of its local telephone providers to move extensions. 'It would cost us $100 or more to move a couple of extensions within the building,' Genske says."

"So the bank began looking for a telephone system that provided sophisticated call-routing abilities and was easy to manage internally. According to Genske, Community Bank & Trust initially focused on IP-based offerings from 3Com, NEC and Nortel, but eventually selected Sunnyvale, Calif.-based ShoreTel's IP phone system. ShoreTel's features hands down set the solution apart from the rest, Genske says, and in February 2004 the bank signed with the vendor."

The bank first implemented the system at its Appleton, Wisconsin location in March 2004. Genske attended a training session onsite at ShoreTel in November 2005, and after that, she was able to implement the ShoreTel system in seven branches in less than three months.

"All the bank's branches were migrated over to the ShoreTel system by January 2005. Genske notes that each branch required its own voice switch, which helps route calls, at a cost of about $2,500 each."

With the new ShoreTel system, according to the article, "calls can be transferred to and from any branch, and tolls for interoffice, intrastate calls are avoided. Call-logging and reporting capabilities generate detailed reports that provide a good picture of call volumes on the main switchboard and among the various departments."

A feature that stands out in Genske's mind, she says, "is the after-hours emergency mailbox, which sends a message to a designated representative's mobile phone. This is critical in the case of lost or stolen debit or credit cards, Genske points out."

Source: "Making the Call: Choosing a New Telephone System; Community Bank & Trust chooses ShoreTel's IP phone system for customer service," by Nancy Feig, FinanceTech, February 1, 2007

For full story, visit http://www.financetech.com/featured/showArticle.jhtml?articleID=197001367

For the ShoreTel case study on Community Bank & Trust and additional ShoreTel financial case studies, visit http://www.shoretel.com/solutions/industry/financial.html.

Government: City of Sioux Falls, South Dakota

Sioux Falls had a Fujitsu 9600 PBX phone system that it was quickly outgrowing, had limited functionality, and proved to be too costly to upgrade. After carefully considering IP telephony systems from leading vendors, Sioux Falls chose ShoreTel for its IP telephony needs. Since deployment, the City has realized numerous benefits.

- In terms of savings, the City, which handles over a million calls a year, saves $30,000 annually in line charge costs.
- The rich feature set of the ShoreTel system helps employees improve productivity and enhances customer service and satisfaction.
- With the ShoreTel system, emergency calls to 911 provide crucial location information that helps emergency response personnel quickly and easily locate the caller.
- The City of Sioux Falls has also integrated its call accounting software with the ShoreTel system, which allows the IT department to track and charge back T1 costs and accurately bill appropriate departments for toll charges.

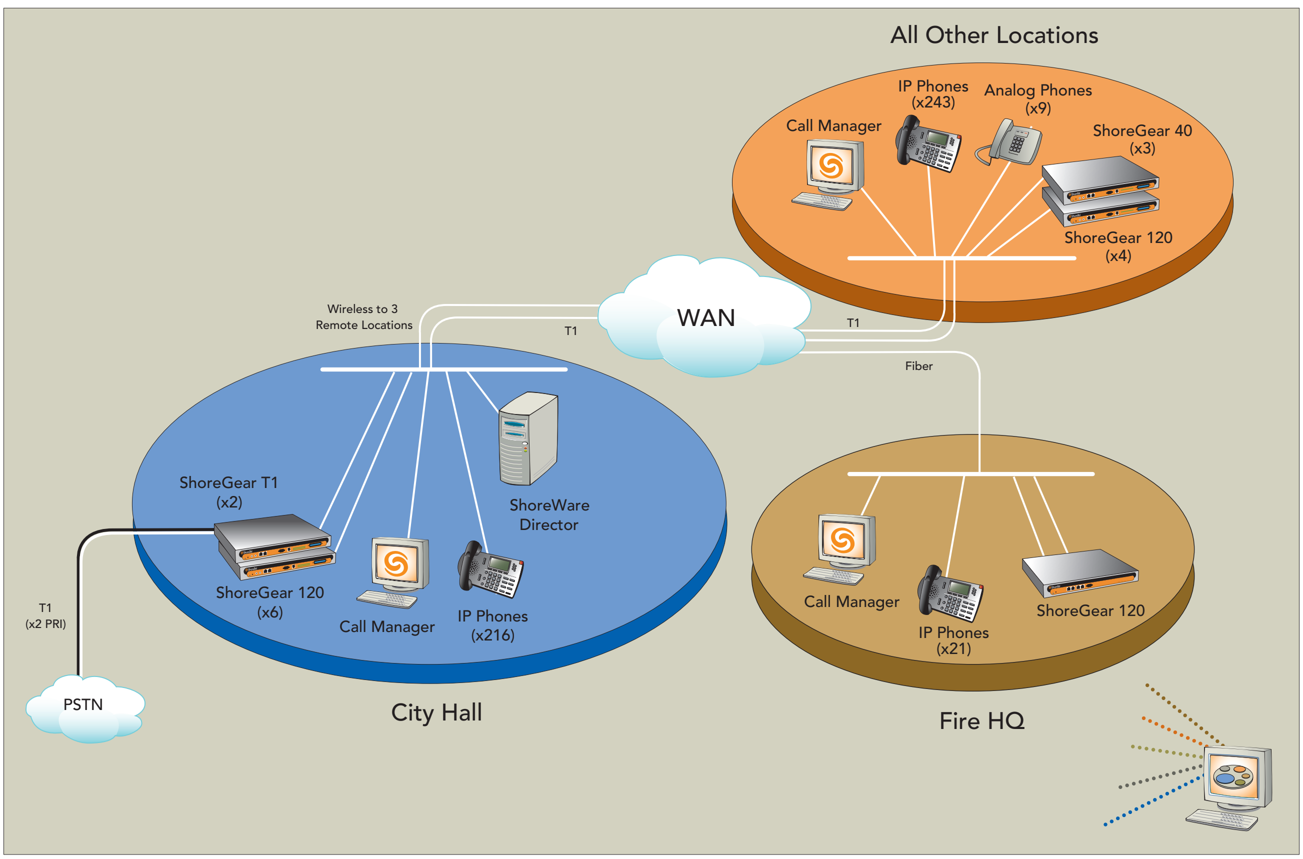


Figure 1: City of Sioux Falls Deployed ShoreTel to 14 Locations

The following is a description of the City's experience as described in a *Network World* article

In 2003, Ed Castle and Monte Watembach, IT Manager and Network Administrator for the City of Sioux Falls, started an aggressive IP telephony rollout to the city's 1,100 full-time employees. The city was replacing its Fujitsu 9600 PBX phone system with a solution from ShoreTel. Castle noted that the city wanted a system that would provide integrated and centralized voice mail so that everyone in every office would be on the same system. After considering solutions from Cisco, InterTel, Nortel and ShoreTel, the city chose ShoreTel for its features, functionality, and simplicity.

According to the Network World article, the focus for the City was on "making the switch over from PBX-based phones to IP telephony painless. 'While [users] are in a two-hour [training] session, we deploy the phones at their desks,' Watembach says. The process is so smooth that most users are comfortable with the new system in two to three days, he says."

"The biggest benefit [the city has] seen is the ability for employees to share workloads. For instance, before the IP telephony system, the community health clinic was overrun by customer calls. 'They only had two lines so callers would often get a busy signal,' Castle says. Now, calls can be queued, and if the operator has too many stacked up, other clinic workers will receive an alert to help handle the load. 'We've raised morale for workers and improved customer service,' Castle says."

Source: "Better customer service: City employees share workloads via IP telephony; Sioux Falls, S.D., making the transition," by Sandra Gittlen, Network World, May 29, 2006

For the full story, visit http://www.networkworld.com/research/2006/052906-voip-voices-case2.html?page=1

For ShoreTel's case study on the City of Sioux Falls and other government case studies, visit http://www.shoretel.com/solutions/industry/government.html.

Healthcare: Visiting Nurse Association of Boston

In early 2005, the Visiting Nurse Association was faced with an unreliable, inflexible, and poorly performing telephone system. In addition, a service provider was needed to manage the system and perform moves, adds, and changes, which was costly and inefficient. After issuing a formal Request for Proposal and carrying out a thorough evaluation of IP telephony leaders, Visiting Nurse Association chose ShoreTel. The benefits since then have been numerous.

- ShoreTel integration with Visiting Nurse Association's SQL patient information database allows receptionists and operators to immediately get a great deal of information on each caller and link calls to the database for a running history, dramatically enhancing customer service for patients.
- Because of ShoreTel's ease of management, Visiting Nurse Association is now able to manage the system in-house, saving money, resources, and time.
- ShoreTel's ShoreWare Auto-Attendant provides 24-hour automated call answering and routing for Visiting Nurse Association to improve customer service for inbound callers, whether they be patients or physicians.

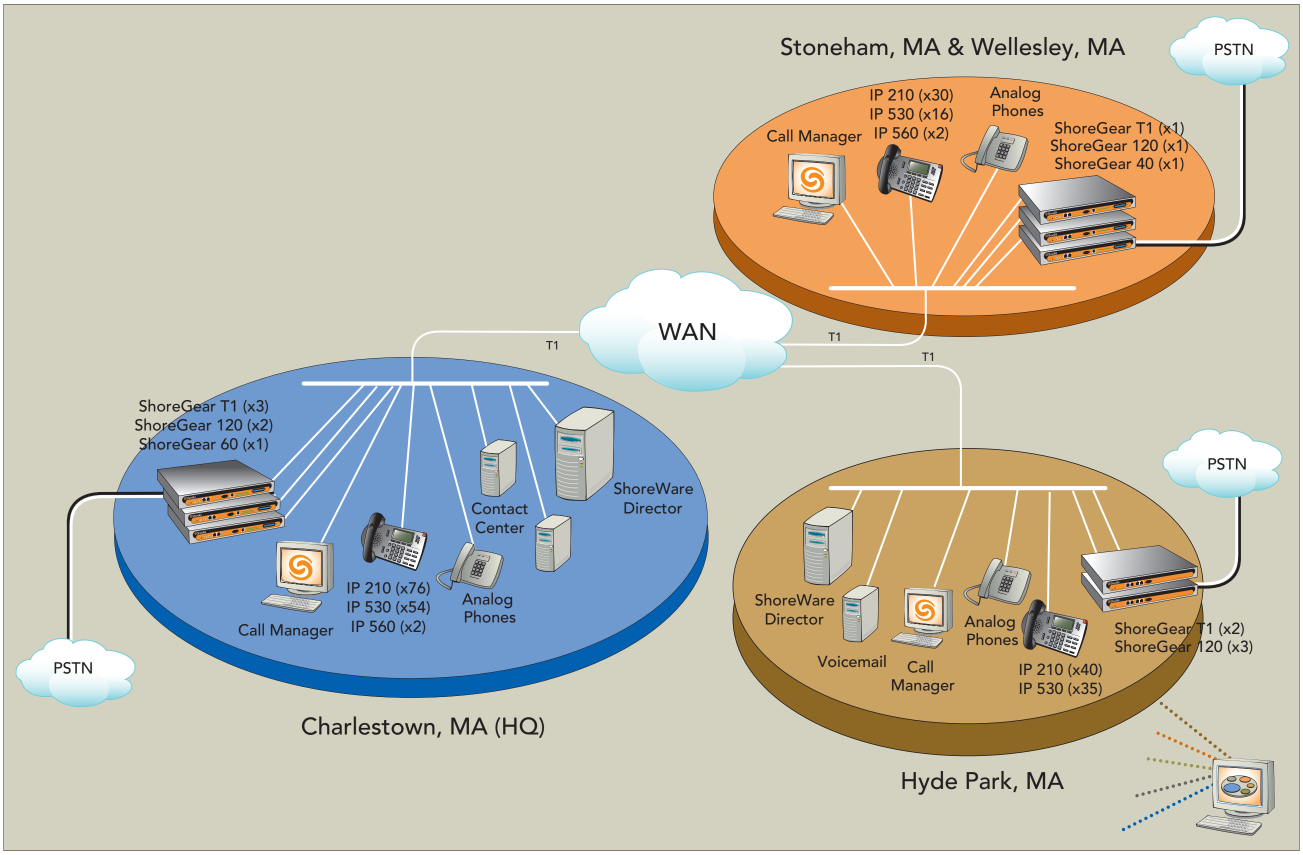


Figure 2: ShoreTel Deployment at Visiting Nurse Association of Boston

The following is a description of the Visiting Nurse Association of Boston's experience as described in *Network World*

For the Visiting Nurse Association of Boston, CIO Fran Lorion, a key test for IP telephony was how the front desk receptionist would react. "She was two feet off the floor for the first few days we had the new system," says Lorion of Lovey Pulsifer, the organization's receptionist, who handles 350 calls per day, roughly 200 from patients. "Believe me, she wouldn't have pulled punches if it wasn't working."

According to the article, "the nonprofit agency, which employs about 500 people and provides service to some 15,000 patients per year, ditched a burdensome Centrex contract and an out-dated call center system and switched to a $300,000 in-house IP PBX system this past summer that Lorion says should pay for itself before long."

"The reason that the system makes such a difference to Pulsifer, one of several receptionists in the organization, is that it automatically identifies callers by their phone numbers and presents their data on her computer screen as she answers the phone. With patients speaking many languages or having accents, the system cuts down on communication problems."

"The application supporting Pulsifer involves the ShoreTel system, through its Call Manager software, delivering caller ID information to an application from Traxi Technologies that sits on the receptionist's desktop. The application then grabs the phone number, queries a Microsoft Access database containing a subset of patient data and delivers pertinent information to the receptionist's desktop."

With the ShoreTel system in place, according to the article, "Lorion says he is looking forward to rolling out other applications. 'Based on what we've already been able to do, we can go in any number of directions with this.'"

Source: "VoIP cures agency's telecom ills," by Bob Brown, Network World, December 19, 2005

For the full story, visit http://www.networkworld.com/news/2005/121905-voip.html.

For the ShoreTel case study on the Visiting Nurse Association of Boston and additional ShoreTel healthcare case studies, visit http://www.shoretel.com/solutions/industry/healthcare.html.

Professional Services: National Travel Systems

In 2003, National Travel Systems was faced with a mix of disparate telephone systems, carriers, local vendors, and toll-free numbers. This often led to busy signals or dropped calls. NTS evaluated IP telephony solutions and chose ShoreTel.

- NTS now enjoys unified messaging so users can retrieve voice mail through their e-mail boxes.
- NTS employees can work from home and easily access the ShoreTel IP telephony system using a broadband connection using ShoreTel's VPN software.
- The ShoreTel system provides reporting capabilities that enable supervisors to monitor call volume, assess productivity, and make necessary changes to staffing schedules.

The following is a description of National Travel Systems' experience as described in an article in *eWeek*

Based in Lubbock, Texas, National Travel Systems (NTS) has the influence and global purchasing clout of nationwide travel firms through its partnerships, while maintaining the feel of a neighborhood travel agency where all management decisions are made locally. While the company may rank among the top five largest privately-held travel management companies in the United States, its communications system was "practically un-navigable," according to the eWeek magazine article. Chris Allen, a major account representative at integrator AMA TechTel Communications, had the difficult task of consolidating the mix of disparate telephone systems, long-distance carriers, local vendors and toll-free numbers into a comprehensive communications system.

At the time, NTS was using various telcos, including Southwestern Bell, ESI, Nortel and SBC Communications. At the same time (in spring 2003), according to the article, "NTS was granted the opportunity to bid on an exclusive, five-year contract with the state of Texas' executive branch agencies to handle all the government bodies' travel reservations. Although eager to land the account, NTS knew that its existing telephone reservation system wouldn't be able to accommodate the state's annual volume of 50,000 travel reservations. As it stood, NTS had fewer than half-a-dozen agents scattered across Texas."

The hodgepodge of disparate telephone systems at NTS "meant that customers would often encounter a busy signal, voice mail or—worse—be forced to dial a string of 800-numbers before reaching an available agent. Further, there's the high cost of managing service contracts with a wide variety of local phone companies."

With the help of AMA TechTel, NTS then spent six months evaluating solutions from four vendors and wound up choosing ShoreTel. Within four months, AMA TechTel completed the installation of the ShoreTel network and replaced NTS agents' traditional phones with ShoreTel IP phones.

"As for NTS' traveling employees, the ShoreTel system provides unified messaging so users can receive voice mail through their laptops' e-mail in-boxes as stored WAV audio for Windows files, Allen said. Similarly, NTS employees wishing to work from home need only establish a broadband Internet connection through a DSL line or cable modem and install ShoreTel's VPN software to access NTS' IP-based phone system. But it's the ability to increase workload rapidly that has truly sold NTS on the ShoreTel system. The system's built-in reporting capabilities allow NTS to monitor call volume to gauge employee productivity as well as flag high-traffic regions in need of greater agent support—key features that helped NTS successfully bid on a contract with the state of Texas executive branch agencies."

Source: "National Travel Systems Unravels Telecom 'Quilt'," by Cindy Waxer, eWeek, October 4, 2006

For the full story, visit http://www.eweek.com/article2/0,1759,2024574,00.asp.

For more ShoreTel professional services case studies, visit http://www.shoretel.com/solutions/industry/professional_services.html.

Retail: AND 1

AND 1's outdated analog phone system was costly to repair, unreliable, and difficult to manage. Additionally, in order to upgrade the system to support Microsoft Exchange 2000, it would have cost over $80,000. After evaluating solutions from IP telephony leaders, AND 1 deployed a ShoreTel IP telephony system, including analog and IP phones, as well as voice switches and software, at the company's Paoli, Pennsylvania and Portland, Oregon locations. The end result is a cost-effective and reliable system which is crucial to the retail industry.

- The new ShoreTel solution saves AND 1, which is running IP telephony over virtual private network (VPN) between its Oregon and Pennsylvania locations, at least $12,000 a year on dedicated point-to-point connections.
- The ShoreTel solution simplifies management and administrative tasks, improves reliability with its distributed architecture, and gives employees powerful communication features tightly integrated with Microsoft Outlook.

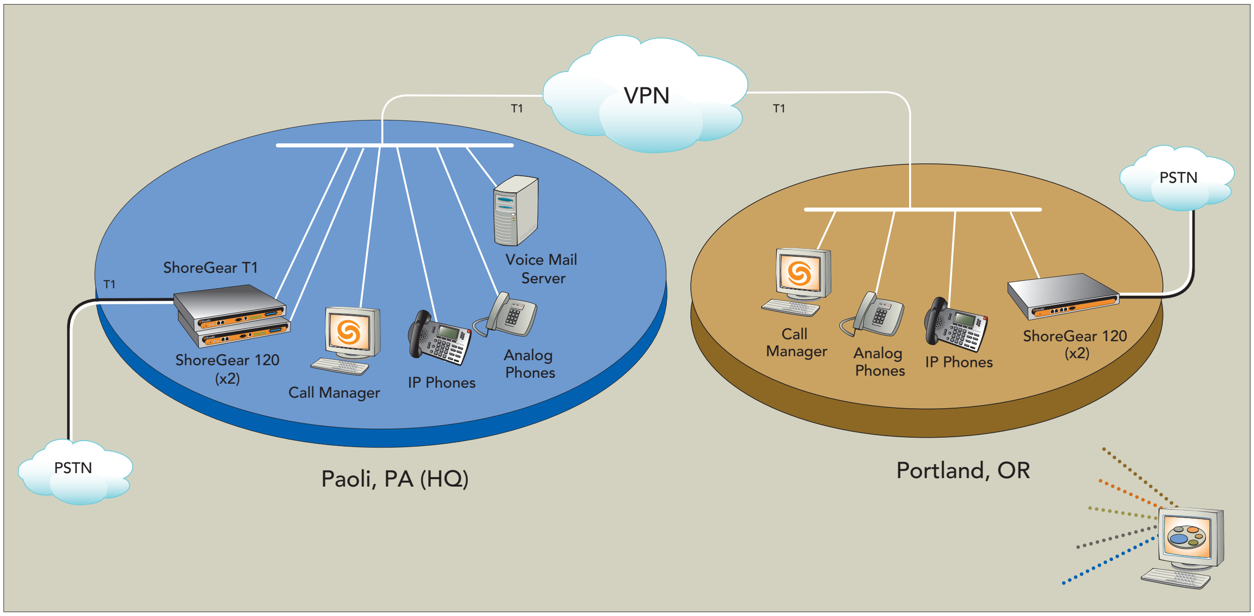


Figure 3: ShoreTel Deployment at AND 1

The following is a description of AND 1's experience as described in Communications News

According to a Communications News article, Adam Resnick, manager of network operations for AND 1, chose a voice-over-IP solution rather than an expensive upgrade of the company's existing analog phone system. Until 2004, the basketball footwear and apparel company was using an analog phone system that was costly to repair, unreliable and difficult to manage. In addition, the legacy system was not compatible with Microsoft Exchange 2000, to which AND 1 was upgrading, so the company decided it was time to make a change.

According to the article, "When we heard that the cost of upgrading the system was that high, we were shocked," says Adam Resnick, manager of network operations for AND 1. "We called on ITI to help us investigate IP telephony solutions."
Resnick chose ShoreTel over three of the other leading vendors. AND 1's technology integrator, Interchange Technologies (ITI), provided the company with seven ShoreGear switches and more than 170 ShoreTel SharePhone IP phones, as well as analog phones to capitalize on AND 1's existing copper infrastructure.

According to the article, "a ShoreTel ShoreGear-T1 provides high-density trunking to AND 1's central office, and ShoreTel SoftPhones give users the ability to turn their laptops into IP telephony phones from home or from the road. AND 1 offers Plantronics headsets to those employees who utilize SoftPhones frequently and prefer a headset over their laptop's microphone and speakers."

"With ShoreTel Personal Call Manager, our employees can make calls right from Microsoft Outlook with a simple mouse click, which is far quicker than the traditional dialing method," Resnick explains. "They can also send a voice mail through e-mail, change their phone settings and retrieve voice mail messages right from Call Manager. Our receptionists can see who's on the phone before they transfer a call, so nobody gets dumped into a voice mail unnecessarily, and if a live person is available, that's where the call gets routed."

"AND 1 runs IP telephony over a virtual private network between its Oregon and Pennsylvania locations, saving itself about $12,000 a year on dedicated point-to-point connections. In addition, Resnick's team no longer needs to dedicate as much time to supporting the telephone needs of AND 1 employees as it had to in the past."

Source: "VoIP+AND1=slam dunk; Switch from analog phone system saves sportswear apparel maker time and money," by Samantha Leggat, Communications News, October 2005

For the full story, visit http://www.comnews.com/stories/articles/1005/1005voip+.htm.

For the ShoreTel case study on AND 1 and additional ShoreTel retail case studies, visit http://www.shoretel.com/solutions/industry/retail.html.

For more ShoreTel success stories, visit http://www.shoretel.com/solutions/industry/.

Resources:
http://www.wikipedia.com
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