Five Steps to Ensuring a Successful VoIP Migration

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Voice over IP (VoIP) has arrived and is here to stay. Many contact centers are realizing the increased flexibility and cost advantages of placing voice traffic on existing data networks. This migration allows increased efficiencies by managing only a single data network instead of separate voice and data networks. At the same time, VoIP reduces overhead associated with right-sizing two separate networks for peak utilization during heavy call volume and data consumption periods.

Additionally, network convergence paves the way to new technologies in the contact center that operate under a similar premise and infrastructure to create a unified messaging platform. These options include: instant messaging/chat, PC to PC, presence/availability indicators, PC to phone, phone to phone calling, click to dial, find me/follow me calling, web conferencing, collaboration and many more.

While many clear advantages of a VoIP migration exist for contact centers worldwide, there are inherent differences in setting up a VoIP network from an existing telephone or data network. If executed correctly, this migration option should not pose many problems. Special attention, however, must be paid to the unique needs of a VoIP network to avoid any degradation of service that could impact call quality.

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The following key steps will ensure a successful migration from legacy voice technology to VoIP in a contact center or enterprise environment:

STEP 1 – LEARN VOIP BASICS

Any organization considering a VoIP environment needs to first learn the basics. Research the similarities and differences between VoIP and traditional telephony, equipment requirements, and so on.

Many sources exist to gain a better understanding of the VoIP environment. From provider resources, white papers, Internet articles and books, it is important to know what to expect before beginning a migration strategy in order to reduce surprises and frustration.

While VoIP from the user experience is likely to be very similar to traditional telephony once the migration is complete, the back office and network infrastructure differences must be understood in advance.

STEP 2 – ASSESS YOUR CURRENT NETWORK SITUATION

Initiate an in-depth analysis of the current environment and support this with complete documentation. Determine a baseline before beginning any VoIP migration to properly forecast additional capacity requirements and to avoid common disruptions such as latency and jitter.

Industry studies indicate that the majority of VoIP pilot programs fail. This is due mainly to the enterprise underestimating the network's ability to handle the volume of VoIP traffic. Typical enterprise telephone users will be on the telephone 20% of the time, while contact center agents are on the telephone 80% of the time. By using current call volumes and published benchmarks as estimates, the enterprise can assess how many simultaneous calls to expect, and can then drill down to specifics on typical usage.

Other, more accurate sources of data may include reports obtained from the PBX or other call reporting services. These reports will help determine current utilization, maximum number of simultaneous calls during peak hours, and average call duration. inContact recommend using both snapshots in time and historical reports.

A company's existing environment should be documented in terms of bandwidth by business category including: voice, email, web services or web browsing, FTP sites, access to network shared drives by location, etc. By knowing the network level or bandwidth resources being used, this will determine a baseline of capacity, overall equipment utilization, trunking and human capital.

A variety of generally available tools exist to help an organization collect this data from the telephone carrier. Some of these options are: ISP trunk utilization reports, packet sniffer software tools, log files and even a telephone system's busy hour call reports.

Realize that latency sensitive VoIP packets can be more adversely affected by poor quality network equipment, un-optimized configuration or bandwidth constraints than any legacy data traffic on the existing network.

STEP 3 – CREATE A SOLUTION AND DEPLOYMENT STRATEGY ALIGNED WITH YOUR GOALS

The overall goals of a solution will be unique to the organization. But typically the goals of the migration are: increased flexibility, mobility, reduced costs and integration with other IP devices.

Because each solution can be unique to the company's objectives, budgetary constraints, network availability and individual resources, there is not necessarily just one right solution. Overall, the desired outcome should be accomplished using available resources and realistic timeframes. A solution that takes several months or years to implement will likely not succeed due to changing technologies, shifting budgets and lack of momentum.

Once the proper solution has been created and documented, the next step is to outline a thorough deployment strategy. Often project management resources are critical to keeping tasks in line with timeframes and budgets and communicating successful benchmarks internally.

Deployment strategies consists of more than just how and when to cut over services to the data network, but also in obtaining the needed equipment (IP phones, gateways, and traffic shaping devices). Obtaining adequate levels of bandwidth to the Internet or private network provider carrying the VoIP calls to and from the outside world is also a critical component.

inContact recommends allocating 85 – 100 kilobits per second (kbps) for each simultaneous call using an uncompressed (highest call quality) configuration. For a compressed configuration, we recommend allocating 25 – 35 kbps per simultaneous call. Utilizing a typical data T1 with a usable capacity of 1,536 kbps, one should expect to provision 15 to 18 simultaneous users in an uncompressed configuration and 44 to 61 concurrent calls over a single T1 in a compressed environment.

Differing types of phones are available for contact center agents in a VoIP deployment. One option is a VoIP 'hard' phone, a dedicated VoIP phone set. Another is a VoIP 'soft' phone meaning a software-based phone that resides on the agent's desktop and is used in conjunction with a headset/microphone. The final option is a typical analog phone with an analog telephone adapter (ATA) unit that converts the signal back and forth from VoIP to analog as the call enters and leaves the agent handset.

Like the solution itself, there is not a single 'best' answer as it relates to VoIP agent equipment. Business needs and agent comfort level should be discussed to determine the best option.

STEP 4 – INSTALLATION AND TESTING

Once the solution and deployment strategy are in place with proper validation and approval, it is time to put the plan into action by implementing the strategy. Using the equipment and additional bandwidth likely identified and procured during the planning phase, installing the new solution can be handled in either a parallel migration or a hot cut.

In a parallel migration, the contact center may install new VoIP phones on agent desks next to the legacy phones. The legacy telephone system remains the primary system as certain calls (unique campaigns, toll free numbers, etc.) are routed to the new phones. While the majority of agents are still completing calls on the legacy system, the VoIP system can be tested.

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This approach allows the organization to have a control group of VoIP agents taking either test or live calls in order to work out any kinks with a much smaller user group and customer base. Over time, this deployment can be widened until all agents have successfully migrated to the new system.

A hot cut may work better for other organizations where over the course of a night or weekend, after business hours, the old system is shut down and the new system is brought on line. When users arrive the next business day, the new system is in place. In any plan, preparing for contingencies in the deployment strategy is critical to ensure little or no downtime in the network migration to VoIP.

Implementing a testing process should quickly identify problems with the new system. Creating a test plan is important, such as calling local, long distance, toll free and international numbers to ensure set-up time and call quality is as expected, as well as to ensure behavior remains unchanged for the user.

Beyond testing, disaster recovery must be addressed prior to implementation to avoid service disruptions. In a VoIP deployment, both voice traffic and data traffic traverse the same network, so an outage is often more disruptive than a standard data outage. By using the Internet or private provider network as the converged data transmission network, one could at least answer the telephone, informing customers that the systems are temporarily unavailable.

Options for disaster recovery may be dual Internet connections from diverse providers, separate network equipment that interfaces the data network, and even traditional telephone back up using analog phones and local lines from a local telephone provider.

STEP 5 – MONITOR AND MAINTENANCE

Monitoring the VoIP network is critical to the success of the migration and the overall experience of the contact center agent and calling customer. To ensure success, there should be a constant awareness of the utilization, network efficiency and call quality.

To validate that sufficient capacity exists, the bandwidth utilized should be matched against predicted bandwidth. Additionally, call quality should be measured to ensure the highest level of call delivery. Customers and potential customers see call quality as a direct representation of the organization and will greatly impact the company's credibility.

Historically, call quality tests consisted of having human testers listen to VoIP conversations and then rate the average call quality from several listeners. Today, VoIP call quality can be measured using industry standard synthetic testing known as Mean Opinion Scores (MOS).

MOS is much more cost effective and yields a more rapid response while still providing a measurement of the overall call quality. Organizations that find examples of poor MOS results can troubleshoot specific areas of the LAN or network connectivity. As a result, they are more likely to improve their quality scores and therefore the caller's experience.

CONCLUSION

While the steps outlined here are high level and require customization to fit the needs of the organization, following a systematic plan should ensure a smoother migration and an excellent user experience for both the caller and the user while offering the organization all of the added benefits of VoIP.

Get in touch with the skilled staff at inContact© for more details on VoIP, and the many flexible implementation options and benefits to your organization.

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