Successfully deploy and maintain VoIP with these best practices

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Your VoIP vendor choice will influence your deployment and maintenance processes, but following general VoIP best practices will also increase the project's success. You can separate and handle voice traffic through standard, vendor-neutral protocols and design practices. Furthermore, general layout and design models for integrated VoIP and data networks provide a strong base for many VoIP deployments. VLANs, QoS, codecs, and compression will each help you build your VoIP solution atop your existing network. This document identifies and explains general best methods for deploying and maintaining successful VoIP networks.

VoIP Models

Regardless of your VoIP network's specific design, your should follow the standard guidelines appropriate for the overall VoIP network model. This document examines two widespread VoIP network models — a centralized design and a distributed design. Centralized models provide call setup and teardown through a VoIP core involving one or more soft-PBXs. A distributed model performs call setup and teardown at multiple locations. The location are normally arranged geographically with links for inter-site connectivity. The centralized model has the advantage of hardware and PSTN consolidation. The distributed model is best used with multiple sites of equivalent size where most traffic remains local.

Redundancy is a critical success factor for a centralized model and is usually accomplished with two or more VoIP servers that can each handle the total load. These servers should be dual-homed with two NICs going to separate switches. Backup power should be provided via a UPS with enough available load to outlast multiple brownouts as well as multi-hour blackouts. The entire phone system depends on these servers retaining connectivity.

While a centralized design will need to be highly resilient, it usually involves less hardware overall. PSTN connectivity can be consolidated at one location with a local gateway or redundant gateways. Organizations typically use a centralized model when they have a large central site, such a corporate headquarters, and several, smaller remote locations. WAN links connect sites into the main location and provide administration of voice communications. Call setup and teardown will be handled over the WAN links, but local site calling will remain local. The centralized model's disadvantage is the requirement to maintain connectivity to the central site from each location. WAN link redundancy is typically expensive, but you can mitigate the cost through local failover options. Many vendors offer redundancy capability, usually through a smaller-end PSTN gateway and minimal phone lines.

In a distributed model, multiple large sites usually require localized VoIP servers. These sites will pass inter-site voice calls through trunk lines that link the sites. The benefit here is a

VoIP Deployment and Maintenance Talking Points

- Choose a general VoIP network design that best fits your organization.
- Centralized VoIP networks let you consolidate hardware and PSTN connectivity.
- Centralized VoIP networks require a highly resilient central site.
- Distributed VoIP networks often require more hardware than centralized networks.
- Distributed VoIP networks don't require as much redundant connectivity as centralized networks.
- QoS allows you to separate voice and data traffic--treating each with distinct transmission policies.
- Compression reduces packet size and increases available bandwidth.
- Codecs that provide high call quality also require more bandwidth.
- The VoIP network should include redundant hardware, redundant links, and adequate power backups.
- Use standard network monitoring techniques and VoIP-specific tools to continuously monitor the network's health and identify problems.

decreased need for redundant connectivity. The distributed model may increase however, to the overall hardware costs and design complexity.

Network considerations

Design consideration will greatly influence the overall success when integrating VoIP with an existing network. Network equipment will support virtual LANs (VLANs), quality of service (QoS), and compression. Using VLANs, you can separate voice and data traffic but have them co-exists on the same medium. VLANs offer a simple way to provide a distinct level of service as well allowing the individual traffic types to be monitored.

QoS is a method of distinguishing IP packets so they can be treated by distinct policies. Your VoIP implementation should prioritize traffic to prevent existing data traffic from undermining voice communication integrity. This process requires matching specific fields within either the Layer 2 or Layer 3 headers inside the data units. Layer 2 QoS, for example 802.1p, allows for multiple levels of prioritization by tagging frames with certain fields. At the switch port level, QoS can then be provided to traffic entering the port. Therefore, as data and voice traffic simultaneously enter the device, voice traffic can be sent first and the data traffic queued for delivery.

On Layer 3 links, such as WAN links between sites, QoS can also be used to match on fields within the IP header. Again, this provides queues for data traffic for delivery while sending voice traffic with priority. As these QoS processes are fairly detailed, you must consider how much bandwidth you will allot to individual traffic types. For example, you could provide 75 percent of a WAN circuit to carry voice traffic, leave 5 percent for call setup and teardown activity, and reserve 20 percent exclusively for data traffic. You may also want to identify highly-critical data traffic that you place somewhere below voice, but above the general data queue. When dealing with limited bandwidth, you should allow ample time to create an effective QoS policy. Your QoS policy will ultimately determine inter-site voice quality as otherwise the network sends traffic on a first-come, first-served basis.

After QoS, compression and codes should be your next consideration. Compression will create smaller packets by shrinking duplicated information. Often seen on low-speed WAN links, compression increases the overall available bandwidth. Although, voice networks usually only compress packet headers, the practice will have a positive impact. The specific VoIP codec your solutions uses will also directly affect bandwidth. A high-quality codec like G.711 uses a sampling rate which translates in higher per-second bandwidth rates. G.711 will require approximately 90Kbps one-way for voice communication. G.729 on the other hand uses a lesser sampling rate, which results in lower call quality, but uses roughly 30Kbps. Many vendors support location-based codec decisions that allow you to use G.711 when making local network calls and G.729 when making calls known to transit a lower bandwidth link. The overall goal is to transmit the maximum number of voice calls while maintaining acceptable call quality.

Network resiliency

End-users expect reliable telephone service, and a successful VoIP solution requires a resilient and redundant network. When possible, the network should contain redundant hardware, redundant links, and adequate power backups. To meet the needs of a VoIP network, core network and VoIP hardware should run off UPS backups. If your VoIP solution uses power over Ethernet (PoE), the local network switches will also need power backups. These switches transmit power to the individual telephones, which will need power and connectivity during an outage.

A strong network design will often translate to a good VoIP design. Medium and large VoIP deployments should involve a close examination of the redundancy at the network's core. Redundancy measures may include dual Layer 3 switches that share default gateway duties through a standards-based or proprietary protocol. The network may also have redundant links from the access layer into the distribution and from the distribution into the core. Not all networks are built to such standards, but you should work to identify any mitigate any points of failure.

Maintaining a healthy VoIP network

You must continuously monitor and test the VoIP network to ensure sustained success. Although a properly designed voice network will initially respond well, quality can quickly degrade if problems are not detected and resolved. You should regularly monitor bandwidth usage and each hardware device's CPU and memory use. Using VoIP-specific management tools you can also monitor the voice network and catch problems basic management will miss.

VoIP vendors may also use the converged voice and data networks for problem reporting. On some VoIP systems, end users can submit feedback directly from their telephones. You can use this information to catch minor issues on which users may not open traditional help desk tickets.

An effective patch management process is also critical for VoIP network health. You must consistently update VoIP hardware and software as you would any other system. VoIP vendors regularly release code revisions and security updates that you must act upon.

The bottom line

Your overall design and implementation process will greatly affect your VoIP installation's success. Once you identify the VoIP model that best fits your organization, the specific design requirements should fall in line. Networking tools such as VLANs and QoS are requirements and can easily make or break VoIP call quality. Make your network strong enough to handle both day-to-day problems and occasional catastrophes. Finally, monitor and maintain your VoIP network to increase its reliability and effectiveness. When you send voice over the data network, quickly detecting and fixing problem is critical for success. Once deployed, VoIP traffic will likely become the most important information your network transmits. You should invest time and effort equivalent to this importance into planning, implementing and preserving your VoIP network.

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