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Scaling a VoIP Network

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Analyzing the challenges in implementing any IT or Networking system, the primary issue is scalability – if you solve this issue, other concerns will be resolved automatically. Building a VoIP network for 10 users is different to building it for 1,000 users, and it completely differs from building it for 1,000,000 users. With the enterprise IP Telephony implementations growing to hundreds of thousands of users, and the carrier class 5 VoIP services growing to millions of users – we start to face the real scalability issues.

Scalability of VoIP networks is far more challenging than scalability of any other service over an IP network. The industry has been trying to bridge over fifty years of experience in implementations of TDM telephony networks, in less than a decade. In addition, any VoIP network must face the challenges of sharing the infrastructure with other applications like data and video.

As per various advanced technologies, most VoIP early adopters started with a small pilot. A small IP-PBX enabling a small number of advanced users, while all other users still use the TDM-based PBX, or a small pilot of residential customers using a VoIP carrier service, while keeping their old TDM-based analog line in case of any failure.

Most of these pilots were successful. Good voice quality, stable service and minor problems were experienced further to the initial deployment. It was only once those customers started growing their VoIP networks that they begun facing the issues behind scaling a VoIP network.

Today, when VoIP is being used in almost every new telephony network within enterprises and service providers, the networks must be planned to scale from its initial stages. A number of factors must be taken into account allowing the network to scale: voice quality, quality of service, redundancy, security, interoperability and manageability.

Voice Quality

Not all IP networks are created equally. After the IP Network is installed, it constantly evolves. An IP network that provides excellent performance at any given point of time can provide poor characteristics at another time. On the other hand, a voice call running on an IP network cannot tolerate large delays, jitters, and low bandwidth. A CEO wanting to move from TDM to VoIP expects the same level of voice quality from his IP phone and network that he has become accustomed to when using his old TDM phone and network.

The past 15 years has seen a number of standards and proprietary algorithms being developed in order to provide good voice quality over the ever-changing IP network. The algorithms deal with echo, double talk, delay, jitter, packet loss and many other dynamic characteristics of a VoIP call. Some of these characteristics are well defined as a part of well known industry codecs, and some are dependent on the specific implementations of the Media Gateway and/or IP Phone vendors.

An important lesson learnt in this instance, is that in order to achieve consistent high quality of voice over an IP network, the VoIP equipment (Media Gateways and IP Phones) must be chosen and tested to include the best mechanisms and algorithms. In order to achieve high-quality voice, it is imperative to use high quality VoIP termination equipment. Nothing less will do the job.

Quality of Service

The highest quality, VoIP termination equipment, cannot operate with a poorly designed IP network. Quality of Service (QoS) is one of the key attributes of a well designed IP network. A voice call is one of the most delay and jitter sensitive IP applications available.

A network built to scale to support voice, data and video simultaneously, must be designed with QoS in mind. The minimum required is to be able to distinguish between packets carrying the different types of traffic, and providing preferential treatment to the voice packets.

Three main mechanisms must be implemented and integrated into the network to support QoS: classification, prioritization, and policing or shaping. The key for scalability in this specific area is implementing these services end-to-end, in every hop of the networks.

With small networks, high bandwidth can substitute these mechanisms, but in order to scale – these mechanisms must be implemented. Classification (identification of the application and marking its precedence) must happen as close as possible to the packet origin. Prioritization (giving priority to the voice application over any other) must be impelemented in any interface of the network, and Policing or Shaping (dropping or decreasing the priority of high volumes of traffic) must occur at all congestion points of the network.

Redundancy

A failure in a voice network is unacceptable. Life with TDM networks adapted us to expect approximately 100% availability dial tone. In order to achieve this same availability in ever-growing VoIP networks, redundancy must be applied. In fact, the geographical independency of an IP service allows the VoIP network to be designed with more redundancy than a TDM telephony network.

Two levels of redundancy must be applied: network redundancy and service redundancy.

Network redundancy can be applied by creating at least two paths between any two IP network elements (switches and routers) creating a network with no single point of failure. Most mission critical IP networks are built this way today, and those networks were proven to provide very high availability.

Service redundancy is being provided by duplicating any server on the network, preferably, locating the two servers in two dispersed locations. Call servers, billing servers, unified messaging servers, and media servers are only a small number of the servers required for a VoIP service to run. All of them can be duplicated with ease, supporting hot redundancy without service interruption.

Network-based redundancy (by duplicating the system) is always preferred over a system level redundancy (by duplicating components of the server), but in user access systems, system level redundancy is the only option and must be implemented.

Security

Unlike in TDM networks, where the telephony network was almost 100% isolated from the data network and the internet, most VoIP networks have a significant amount of interface points with the data network. Exposed to dangers in the Internet world, a poorly designed network can be easily exposed to security threats, such as denial of service, computer viruses, and data theft.

However, the technologies available today, in addition to a well designed network, can offer a higher level of security as compared to the TDM world. The main security threats that VoIP networks can be exposed to include: Eavesdropping (the ability to listen to another party's telephone conversation without authorization), Impersonation (the ability to impersonate someone else and to have a phone call with the other party), Fraud (have a call on the network without paying), Local denial of service (the ability to block phone calls from or to a specific location out to the network) and Network denial of service (the ability to bring down the network or large portions of a telephony network).

Most, if not all of these threats can be prevented by implementing state-of-the-art security mechanisms that are part of all leading VoIP equipment today. Call control, voice stream and network management traffic can all be authenticated and encrypted using standard mechanism such as IPSec, SRTP, AES and various other algorithms and protocols. The implementation of these mechanisms, require suitable VoIP equipment and the careful design and maintenance of the network from the first day, while growing the network.

Interoperability

Although VoIP protocols are being standardized, and numerous networks are built using standard protocols, interoperability is still an issue and will be challenging for years to come.

Most leading enterprise IP telephony vendors still use proprietary protocols which are not interoperable with other protocols. In the carrier environment, IMS is a promising architecture, designed to standardize carrier voice networks, however it is still in the development stages.
In order to build a VoIP network to scale, it must be taken into account that a large factor of uncertainty exists. Access technologies, voice and video codecs, network and terminal devices, and network protocols are all evolving and changing.

Choosing the correct VoIP vendor that can provide a true, open system that can support a large variety of control protocols, codecs and access technologies is a key for allowing the VoIP network to scale into the future.

New services are being introduced daily, and most are distributed by niche vendors. In order to integrate those into an existing VoIP network, the network must be designed with interoperability in mind.

Manageability

Small networks do not require a large number of management tools. A small network can be managed using the embedded web management tools which are currently part of all vendor's equipment. On the other hand, growing the network, comprehensive management tools become more and more critical. In very large networks, provisioning becomes critical too. In order to scale a VoIP network, the management infrastructure must be designed and planned from the ground up. The right VoIP equipment and the right management tools must be present in order to allow quick provisioning, configuration, problem isolation and diagnostics.

Summary

VoIP technology is replacing TDM on a global scale. Designing a VoIP network to scale involves detailed planning and taking many factors into account. A well designed VoIP network can scale to support millions of users with very high voice quality, availability and manageability. As with other technologies, it is easy to take a wrong turn. However, if a technology is optimized, you will enjoy the benefits of saving on operational costs, while providing extensive services and a 21st century experience to your users.

About the Author

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Haim Melamed has 15 years' experience in the networking industry and is currently the Channel Marketing Director at AudioCodes. He is specifically responsible for the worldwide marketing of AudioCodes' solutions via channel partners. Before joining AudioCodes, Haim worked at Cisco Systems for seven years, where he led the technical and marketing activities in the Israeli, Cypriot and Maltese markets. Before joining Cisco, Haim worked for 3 years in the communications division of Team Computers and Systems, one of the leading Israeli IT systems integrators, as the pre-sales support manager. Haim Started his career at the computer center of the Israeli Defense Force, where he served for 4 years as a network architect and a project manager.

About AudioCodes

AudioCodes Ltd. (NASDAQ: AUDC) enables the new voice infrastructure by providing innovative, reliable and cost-effective Voice over Packet technology and Voice Network products to OEMs, network equipment providers and system integrators. AudioCodes provides its customers and partners with a diverse range of flexible, comprehensive media gateway and media processing technologies, based on VoIPerfect™ – AudioCodes' underlying, best-of-breed, core media gateway architecture. The company is a market leader in voice compression technology and is a key originator of the ITU G.723.1 standard for the emerging Voice over IP market.