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E-Guide

IP Telephony for Mid-Sized Companies: Current trends and what's in store for the future

Although the benefits of IP telephony may have made implementation a no-brainer for mid-sized companies, new options and services present both challenges and opportunities. This e-guide highlights the top 5 trends and current outlook for enterprise IP telephony and how you can take advantage of SIP trunking to more easily benefit from improved collaboration and reduced costs.

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Top 5 trends in enterprise IP telephony

So you've completed your enterprise IP telephony implementation, your systems are running without incident, and every person you meet in the hall stops you to shake your hand for all that you have done to improve their lives. It's time to kick back and relax, right? Not quite. New options and services present challenges and opportunities to improve collaboration and reduce costs. So what comes next after full-scale enterprise IP telephony deployment?

In our research and consulting interviews with IT architects, we see the rise of five distinct trends in enterprise IP telephony shaping communications going forward:

1. **Unified communications:** While often a confusing term, we define UC as the joining of various real-time communications applications into a suite of integrated collaboration services. For most, this means tying telephony together with instant messaging, conferencing, unified messaging and video—enabling users to see each other's presence status and initiate any form of communications via a single application. Over 60% of companies have some UC implementation under way, often starting with IM-telephony-presence integration before moving on to additional applications.

2. **SIP trunking:** Replacing legacy PSTN access with SIP-based services can reduce costs and improve flexibility for call routing, mobility integration and disaster recovery. On average, companies save anywhere from 20-60% off their PSTN access bill when they move to SIP trunking. Those that centralize access realize the largest savings. A successful SIP trunking initiative does involve addressing some challenges, most notably the need to support fax, E911 and performance management across SIP trunks, as well as potentially revising your dial plan.

3. **Video:** Is video the next voice? Perhaps. As quality improves, prices fall, and workers are increasingly distributed, we are seeing an increase in video conferencing adoption. Desktop video is now an inherent feature in most UC platforms, while vendors including Avaya, Cisco, Microsoft and Polycom enable integration of UC desktop apps with room and immersive telepresence systems. Video hasn't quite emerged as a replacement for voice,

but we do see desktop video conferencing growing, primarily to enable distributed workers to join room-based meetings.

4. **Virtualization:** In the last few years, VoIP vendors including Avaya, Cisco, Microsoft, Mitel and Siemens have ported their IP PBX software to virtual appliances or general purpose hypervisors, enabling their customers to take advantage of lower infrastructure and operating costs. Now some of those same vendors are working to support voice and video via virtual desktop infrastructure (VDI). VDI raises some particular thorny challenges thanks to the need to localize voice/video encapsulation, but it also offers the potential to reduce capital and operational expenses at the desktop.

5. **Mobility:** One of the most frequent questions I get is, "can I get rid of my desktop phones?" Companies are actively looking at extending telephony and UC capabilities to mobile users across a range of smartphone and tablet devices. While the ability to eliminate the expensive desktop phone (and the required Ethernet infrastructure) is attractive, be aware that mobile voice services require careful attention to wireless LAN (WLAN) architecture. For those on public wireless services, voice quality still lags behind that of a hard phone. Still, we see growing use cases, especially for field workers, where simply provisioning an integrated mobile phone makes a great deal of sense.

Now is not the time to rest on your laurels. While the enterprise IP telephony market is indeed maturing, there are still significant opportunities to reduce costs, improve services and drive innovation in your organization.

The future of IP telephony still trapped in the PBX era

Of all the discussions I have with IT decision makers regarding Voice over IP (VoIP), the most common set of questions tends to revolve around the future of the IP PBX—and who can blame anyone for wondering? The telephony industry has had more innovation and transformation in the past 10 years than in the 30 years previous to it, which makes keeping up very challenging.

I fully expect the pace of innovation to continue and even accelerate, making it even more difficult for enterprises to anticipate what the future of the voice platforms will be and position themselves accordingly. I'd like to provide some insight into what I think the future of IP telephony and the IP PBX will be.

The failed promise of IP PBX

Deploying voice used to be easy. You plugged your phones into a PBX, and it all magically worked. The problem with these older PBX platforms was that they were highly inflexible, inefficient and had to be deployed on a location-by-location basis. There was no way to extend features past the installed location, and they had to be run on an entirely separate network.

Then along came the IP PBX, which was supposed to be the panacea to all of our telephony woes. By converging voice and data networks, the cost of running telephony would drop through the floor. We would get more functionality and users would be more productive. It would be a win for IT, the users and the company—but this was not the case for many organizations.

Depending on who you talk to, the amount of cost savings tends to vary. The upfront costs are high, and in many cases, there isn't all that much new functionality. We make calls, but we could do that before. So what's held us back?

I think the main drag on the industry is that we really have not left the PBX era. Sure, all of the new systems are built on IP. We have new (and expensive) IP phones, but architecturally, not much has changed. A large number of the organizations I have worked with deploy VoIP by replacing every PBX or key system with an IP PBX. So at the end of the day the entire architecture is the same; we're just using IP as the transport mechanism instead of the PSTN, but at the most basic level, we've spent a great deal of energy trying to make our IP systems look and act like the old TDM systems.

The evolution of the PBX

The first wave of evolution away from the PBX era followed along the same path as computing. We deployed the new platforms, but kept the architecture the same. Much of this was due to gaining comfort with the new systems, but the IP PBX platforms themselves weren't ready to support a different architecture.

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The next wave of VoIP will be similar to what happened in the computing world. Call servers will become data-center based resources that use the corporate WAN to deliver services to the branches and other workers. Think of any other application in the organization, like email or CRM systems—all of these reside in the data center and are delivered over the network.

The technology evolution has also contributed to this trend. All of the new voice platforms—such as Avaya Aura, Cisco Unified Communications Manager and Microsoft Lync—are designed to be deployed this way. They follow application and Web principles to allow greater scale and efficiency.

The other important thing to note about these platforms is that they do more than just initiate and tear down calls. A next-generation IP PBX should be thought of as an IP session manager and not a call server. Calls are one of the features enabled by the IP session, but the session brings the ability to layer on video, presence and other multimedia functions.

Since the session is at the IP layer, it becomes highly mobile and can be moved from a phone to a tablet to a PC with relative ease. This will bring with it another level of cost savings and start to fulfill the long-term vision of what the move to IP was supposed to be: connecting any communications tool to any user on any device. But in order for that to happen, there needs to be another evolutionary step, which we are just starting to see.

IP PBX: Opening minds to open platforms

In the long term, I think the industry will stop thinking about the PBX as a closed system and instead see it as set of open platforms that work together to deliver the necessary functionality.

To clarify my point, I'll again use the application world as an example. In most large organizations, application strategies are the same. An underlying services layer provides all of the common functions, such as authentication and directory services. Above this layer, applications are built on a few platforms, such as .NET or Java. Middleware is then used to tie it all together. Though this is a gross simplification of how applications work, and there are all kinds of exceptions, this is the primary route most companies take.

Similarly in communications, what we will eventually have is a common-services layer for things like presence and user identity, some of which will be shared with other applications. A set of platforms will sit above this layer—perhaps one for voice, one for video, one for social media—with a higher-level, abstraction layer for the middleware functionality. In fact, Avaya's ACE, VOSS and Alcatel-Lucent's OpenTouch are good examples of some early communications middleware.

Though we're a long way off from this, it is the direction the industry is headed. Are we out of the PBX era? Hardly, but we're rapidly moving that way.

SIP trunking primer

Session Initiation Protocol, or SIP, is a signaling protocol for initiating multimedia usage in a network, including video, voice or chat. SIP establishes the end system to be used and handles call transfer and termination. An important aspect of SIP is SIP trunking, a use of VoIP that can minimize communication costs and maximize network usage with unified communications.

What is SIP trunking and why is it important?

SIP trunking is a service provided by an Internet telephony service provider (ITSP) that uses SIP to connect a traditional private branch exchange (PBX) to the Internet. In traditional telephony, the phone company delivers services over a "trunk" that connects the PBX to the public-switched telephone network (PSTN), and this trunk carries the phone calls from the corporation to the public. With SIP trunking, the physical wires in the trunk are replaced with a converged voice and data network. This eliminates the need for an expensive media gateway and reduces long-distance charges. SIP trunking combines voice, video and data in a single line, eliminating the need for a separate physical medium for each one.

SIP trunking consists of three main elements: the ITSP or SIP trunking provider, the IP PBX, and a border element, which facilitates connectivity among an enterprise IP network, the PSTN, and an external IP-carrier network. The border element, which could be a SIP-capable firewall or a switch to transfer calls in and out of the PSTN, is generally managed by the service provider.

For example, a Los Angeles-based customer service rep at a company employing SIP trunking places a phone call to a client in Chicago. The call is then converted to an IP call (or originates as one) before it leaves the office, travels mostly over the service provider's IP network, then drops onto the PSTN at the termination point. Because most of the call traveled over the service provider's IP network rather than the PSTN, traditional long-distance calling charges won't apply, and the cost of the call will be a fraction of what it would have been if placed over a traditional PBX.

The benefits of SIP trunking

Thanks to this level of cost savings, SIP trunking benefits any enterprise with a PBX that connects all internal users and where the employees make long-distance calls on a regular basis. With the commonly used ISDN, each circuit has a maximum of 30 channels. But with SIP trunking, there is flexibility in line usage because companies don't have to buy capacities -- they can have as many or as few users as they want.

Companies can also save on the costs of media gateways because gateways needed to connect to the PSTN will reside at the ITSP. SIP trunking is also a more efficient alternative to an ISDN or other traditional TDM lines, such as BRI. An ISDN sends digital voice and data over the traditional copper wire, requiring the monthly line rental of ISDN circuits. Because SIP trunks eradicate the need for costly PSTN gateways and traditional TDM lines by using an IP connection to the ITSP, enterprises do not incur costs besides what they pay for the ITSP.

Another benefit of SIP trunking is in bandwidth utilization. Both telephony and Internet lines often utilize bandwidth at a low rate. With the telephony line, telephone usage is generally characterized by several hours a day with many calls and several hours a day with very few calls. Internet usage is rather erratic, with bursts of use throughout the day, thus wasting the capacity at the time that usage is low, for both types of lines. With SIP trunking, combining these to a converged line, you can optimize bandwidth for average consumption rather than peak usage. Quality of service (QoS) settings can allow for prioritizing between voice and data and can ensure that the capacity you need is always available when you need it.

SIP trunking also delivers real-time communications applications – such as IM, presence, video conferencing, and application sharing -- in a cost-efficient, reliable way. With SIP trunking, these real-time communications applications improve collaboration and productivity by reducing human latency and increasing awareness of employee locations, especially for initiating ad hoc meetings. For example, presence allows a hospital administrator to know where the hospital's doctors are at all times, in case of an emergency.

Least cost routing (LCR)

Creative use of SIP trunking can allow an enterprise to reduce calling costs even further, if it employs the use of least cost routing (LCR) and multiple ITSPs. LCR is the process by which a company chooses the path of an outbound communication by the price of the call, opting for the lowest price. If a company uses multiple SIP trunks from different service providers based on geographic locations and time zones, each call can be routed to the cheapest service provider based on country codes, saving a significant amount of money on international calls.

In terms of ROI, this depends on the company's size. Most medium to large companies have multiple locations and often use international calling, so businesses in this scenario can save 50% to 60% of their usual communication costs, and ROI can be achieved within 12 months.

Addressing the challenges of SIP trunking with best practices

With all IP-to-IP connections, security can be an issue because they are always connected to the Internet. SIP trunks are vulnerable to standard signaling and media security issues, as well as peer-to-peer issues if enterprises trust others to provide authentication. However, SIP server and proxy technologies can effectively manage the flow of SIP traffic, as long as there is a reliable IT manager to deal with the security threats with verification and authentication policies.

Another hurdle with SIP trunking is deploying and managing equipment from different vendors. As in any unified environment, mixing equipment based on the SIP protocol from different vendors can cause interoperability problems stemming from incompatibility. Two IP PBXs may seem to be compliant to the same requests for comment (RFC), but because of the open-editing nature of RFCs, the RFC definition for SIP has become overly flexible and vague.

To help battle this problem, SIP Forum, an IP communications industry association, has introduced SIPconnect. This is a recommended set of industry interoperability guidelines designed to facilitate the connection between SIP-enabled IP PBXs and SIP-enabled service

provider networks and to establish industry-wide norms. It specifies VoIP protocols/features, a reference architecture and implementation rules. By using SIPconnect, enterprises can solve interoperability problems.

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