TIME TO START SIP-ING?

Can a SIP-based solution add value for your contact center? A look at the key opportunities and considerations.

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Session Initiation Protocol is among the latest contact center technology innovations that carry the promise of delivering unique capabilities to benefit your company and your customers. But is it for real? Is there a compelling case for you to go down the path of using this relatively new technology?

If saving money and adding flexibility and agility to your inbound and outbound voice circuits sounds useful, then SIP trunking may make sense for your center. You may also have opportunities for SIP-based integration and interoperability. But like any new technology, you need to understand the tradeoffs and risks. SIP has implications for hardware configurations, capacity, voice quality, availability, security, network features, disaster recovery and redundancy. Here's what you should consider before taking a sip, or perhaps a gulp!

SIP Defined

Session Initiation Protocol (SIP) is a text-based signaling protocol that establishes and tears down sessions for various streams of data supporting voice, video, text and more on an IP network. It was established by the Internet Engineering Task Force (IETF) to improve peer-to-peer communication, or endpoint-to-endpoint, communication. As such, SIP can facilitate communication among endpoints, such as a PBX and SIP phone, or two people having an Instant Messaging session through desktop PCs. SIP is similar to HTTP for web communications in that it is modular and extensible to enable a wide range of applications. It is replacing existing VoIP protocols (e.g., H.323 and H.248) developed by the International Telecommunications Union (ITU) for many VoIP applications.

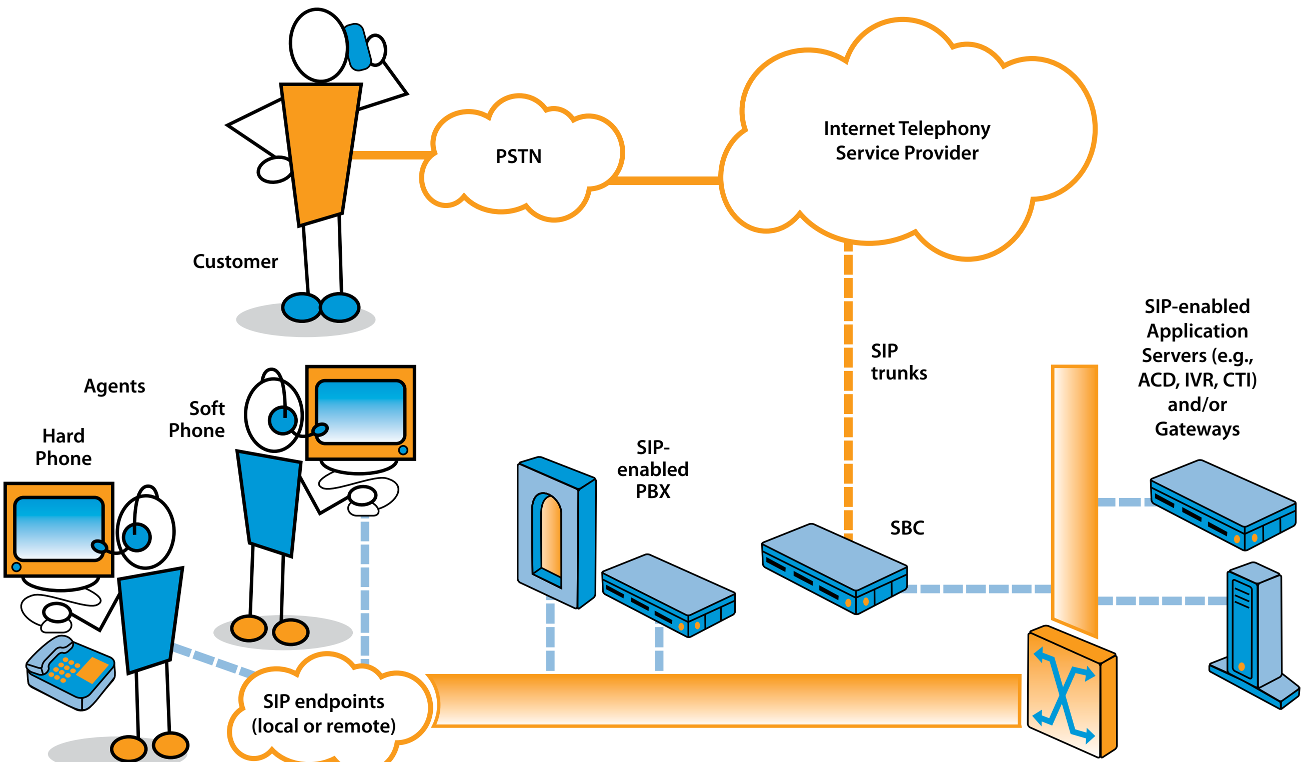
SIP Opportunities

The buzz on SIP for the contact center revolves around a variety of services and features. Platforms built on SIP support **routing** of all media types (voice, video, email, chat, IM, faxes, SMS or workflow object) using a common signaling protocol and administrative tool. They break the media silos that impact routing, reporting and administration across contact types, and enable the move from a "call center" to a "contact center."

Companies used to have to purchase high-cost proprietary phones (whether IP or TDM) from their PBX vendor. Now they can purchase SIP **phones** from a variety of third-party vendors, introducing more competition and lower cost alternatives. Typically, the vendors enable soft phone and/or hard phone options based on this standard.

Another area to leverage SIP is with **interconnectivity** among contact center applications (e.g., IVR, ACD and CTI). SIP standards and APIs alleviate the complexity of moving data about customers and contacts among applications, and connecting contacts to endpoints. Screen pops and IVR integration can be simpler and more cost-effective using SIP. In a multisite contact center environment, you may have PBX and ACD systems in place from a variety of vendors. While the ultimate end-state is a centralized architecture supporting all sites, getting to that point can take time. Leveraging SIP interconnectivity between disparate enterprise and contact center equipment vendors enables you to use a common routing and reporting engine, as well as IVR, CTI, QM and other applications. This interoperability is a boon for companies that have made independent decisions by site or business unit or have inherited diverse platforms through acquisition. With the right partnership, SIP could even improve outsourcer integration.

Perhaps the "big daddy" of SIP buzz is SIP **trunking** to add agility and lower costs for the trunking into the center (e.g., toll-free) and/or the connectivity among locations. Converting from digital T1s to SIP trunking using voice compression allows for up to eight times the number of conversations on the same size circuit. And unlike T1s, SIP capacity can be bought at any increment (not just 24 or 23 additional voice paths). Some studies show a 30% to 50% reduction in overall transport costs. You also have the opportunity for one carrier to supply Internet access, toll-free, long-distance and local service, thus providing more favorable rates than service provisioning through separate vendors. You can use common "trunks" to support the



SIP Application Servers include functions such as proxy, location, registrar, and redirect to facilitate connecting user agents (on SIP endpoints)

variety of media and adjust capacity by media as needed to support hourly, daily or seasonal peaks in interaction volumes. In addition, when connecting remote sites to central hubs using SIP trunking, you do not waste "idle" circuits at remote offices.

Is your interest peaked with the buzz? If so, it is important to understand the impacts of some of the decisions that can give you significant savings. The more functionality you use, the more critical it is to consider the cost and benefit tradeoffs. In the end, what you already have in place and how you size and configure your SIP trunking will impact the potential savings available to you.

SIP Trunking Considerations

When considering SIP **trunking**, you likely will need a Session Border Controller (SBC) at your entry point for security. You will need to determine who will provide the SBCs: you, the carrier, the equipment vendor, or some combination thereof. You will need to identify where you place the SBCs—your data centers or theirs—and assign responsibility for monitoring and support. You will need SIP gateways at each site; the more disparate systems, the more you will need. Keep in mind that a standard is not always fully interoperable across vendors, so choose your solutions carefully and include provisions for adequate testing.

When looking for savings with **capacity**, combining voice and data into a common pipeline triggers the need to look at capacity differently. What if you are already using VoIP? What if you already have centralized trunking? In either case, you may already receive the bulk of the savings potential. Instead, your opportunity may be in connecting to remote sites with SIP trunking. When calculating capacity, size for the maximum of what you will put on SIP trunks, such as toll-free, direct-dial inbound, outbound for the center as well as the enterprise and remote offices, Internet, data communications, email and conferencing. And don't forget video. Once you combine media, your peak periods may change from what you typically see with a focus on voice alone.

Consider **traditional toll-free and direct-dial features** as you make the switch to SIP trunking. If you are using network services such as Time of Day/Day of Week, percent allocation, prompting, etc., you may need to move those functions to your IVR and ACD. You need to work with your SIP provider to understand how the delivery of SIP-based interactions works among your data centers. This comes into play with the always important **disaster recovery and redundancy**. SIP could aid in your agility for backup scenarios, moving contacts to other data centers. You will want to plan for redundant access among data centers. If using SIP trunks to remote sites, consider local access by site as a backup option, as well.

Managing change is a huge consideration when making the move to SIP trunking for your contact centers and/or remotes sites. This can be a big project with efforts to identify all your toll-free numbers and local dial numbers. You may need to do a "Resp Org" change to reassign management of the numbers from one party to another. Not all local direct-dial numbers into your center can be ported to a new carrier.

NOW MAY BE THE BEST TIME TO REVISIT NETWORK CARRIER OPTIONS

YOUR EXISTING NETWORK VENDOR may not beat down your door to encourage you to switch to SIP trunking. After all, your gain may be their loss (you save money, they lose it). The best way to reduce costs when migrating to SIP trunking is to switch carriers, or put your network services out to bid. Smaller or other alternative players may be more eager to get your business by promoting their SIP offerings. Make sure that you understand which carriers (large and small) can offer SIP without losing your existing features, while delivering solid reliability and quality.

Other SIP Considerations

As you pursue SIP, you will also need to look at considerations for **routing, phones and interconnectivity**. These items are all part of the solution you use for core contact center technology—whether from Cisco, Avaya, Genesys, Interactive Intelligence or others. You will want to understand their positioning with SIP. Is it fundamental to their architecture going forward? How are they using it in each of these three areas? Do you have to buy their phones, or are there third-party options? Can you use their routing engine with a diverse set of platforms and have them all "talk" to each other using SIP? Are there any limitations on applications integration? If you are migrating an existing system, understanding the interoperability and how the traditional world fits with the new SIP (and VoIP) world will make it much easier for you when it's time to implement and test in your new environment.

Take a Taste

As SIP gets more traction, pay attention, watch and learn. Put it to work on behalf of your center and your customers where it makes sense. But keep in mind that you don't necessarily need to jump on the band wagon, especially if you are already leveraging VoIP or have centralized (and optimized) trunking. Carefully assess the benefits you will gain and any tradeoffs that compromise key goals for your customer interactions. If you do pursue SIP, be sure that IT and the contact center work together, along with your vendors, to talk about what they are doing with SIP, why, what the benefits (and risks) will be *in your specific environment* considering the number of sites, size and which capabilities you will use. At a minimum, stay on top of what is happening with SIP so you are prepared to leverage it when replacing your contact center technology.

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QUALITY AND RELIABILITY CONSIDERATIONS FOR SIP TRUNKING

Those who have migrated to VoIP are already aware that voice quality and reliability are different (read: not as good) from traditional TDM-based systems. When packetizing voice conversations and sending them across the same circuits as other data, problems can (and do) exist. There are ways to remediate these issues, and terms like “MOS score” and “five nines” get bantered about. What do they mean?

MOS (mean opinion score) is a way to put a number to a subjective measure: How does the call sound to you? MOS scores are 1-5, with one being the worst and five being the best. Numbers of 4 and higher are sometimes referred to as “toll quality.” With proper Quality of Service (QoS) elements in place to prioritize voice traffic, a MOS score of 4 is achievable on VoIP and SIP and is fully adequate for any conversation.

For reliability, some vendors offer guarantees with their solution as a percentage of “up-time.” The legacy TDM systems were very reliable and circuit failures from telco vendors were rare. Traditional systems and network services typically achieved 99.999% reliability (five-nines), or about 5 minutes of downtime per year. As systems and trunks move to VoIP architectures and SIP-based protocol, more failures do occur. A common reliability guarantee with these systems and network services is 99.99%, which translates to almost an hour of downtime per year. Less reliable systems can result in hours of outage per year.

So ask yourself: How important is staying connected to your customers 24x7x365, and ensuring high-quality voice conversations? Take steps to ensure that your needs can be met.

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