PRI Handoff Considerations when Moving to SIP Trunking

In This Paper

- PRI handoff is needed to realize the benefits of using VoIP and SIP
- Sites with TDM PBXs need special PRI handoff consideration
- Level 3® Voice Complete includes SIP Trunking with native PRI handoff
Introduction

Many organizations are moving their sites over to SIP Trunking services to take advantage of the cost benefits of voice over IP (VoIP), productivity and business benefits of using new IP-based communications and collaboration applications, and to leverage the inherent business continuity and disaster recovery (BCDR) capabilities of using SIP Trunking and IP services.

Frequently, the one sticking point preventing a full-scale migration of all sites is that companies must decide how to incorporate sites that still have legacy Private Branch Exchanges (PBXs). Specifically, most older PBXs cannot be directly connected to a SIP service. For a site to realize the benefits of using VoIP and SIP there must be a PRI handoff between the SIP service and the site’s PBX.

PRI Handoff Considerations

Many companies moving to IP voice and IP-based unified communications (UC) and collaboration applications will find that they have a mix of sites; some have IP PBXs and some still rely on Time Division Multiplexer (TDM) PBXs.

Using a SIP Trunking service at a site with an IP PBX is pretty straightforward.

The other sites, the ones with the TDM PBXs, need special consideration. There are a handful of ways companies can deal with this issue. Each approach has its own appeal and points to consider. As is the case with the adoption of any new technology, a cost/benefit analysis must be performed to determine whether one approach versus another should be used at each site. The options for bringing a site that currently has a TDM PBX into an IP-based infrastructure include:

Do nothing approach: Some companies simply have not upgraded their installed base of TDM PBXs at every office site. With tight IT budgets over the past few years, some companies have focused their attention on other areas rather than replacing functioning TDM PBXs. It’s the old “if it ain’t broke, don’t fix it” mentality.

Even considering the positives (e.g., lower communications costs, increased productivity through simpler collaboration, etc.) that can come from moving phone traffic to an IP network and integrating voice into more powerful UC applications, a company may determine that the costs to bring a particular site into the fold exceeds the expected benefits. As a result, management may decide that it makes no sense to extend SIP Trunking to all sites.

This avoids new CAPEX and OPEX costs. However, voice communications to and from the sites will continue to require switched circuits and be carried over
the public switch telephone network (PSTN). Furthermore, the workers at these sites will not be able to use VoIP and UC SIP-based productivity features that make it easier to route calls, set up teleconferences, and more easily collaborate.

**Install an IAD or gateway:**
Excluding all sites with TDM PBXs may not be practical. Staff in larger regional offices and key workers in other remote sites might need the full features of SIP-based VoIP and UC applications to do their jobs.

One way to incorporate such sites into an IP communications network is to use a gateway router or Integrated Access Device (IAD) to connect a legacy PBX to a SIP Trunking service. With this approach, those systems can be included in a converged IP network carrying voice and data traffic.

There are several points to consider when opting for this approach.

First, there are CAPEX and OPEX costs associated with acquiring, installing, administering, and maintaining IADs and gateways in each site. If maintaining communications is business-critical for a particular site, a company would need to deploy and support redundant IADs and gateways to address BCDR issues. Specifically, to ensure a site can maintain communications if an outage or disruption occurs, the company would need to have a backup gateway/IAD to provide a secondary path for the VoIP traffic to flow.

The costs of adding a primary and perhaps secondary gateway or IAD in a site might be impractical to incur and lead a company to exclude some smaller sites from its IP communications network.

A second point to consider with this approach is that there is a mixed impact on end users in the sites where gateways or IADs are used. Workers retain their old phones (which also means the company does not have to buy new IP phones), but they cannot leverage all of the advanced call routing, call setup, and call delivery features of a native SIP-based system.

**Replace the old PBX:** For some sites with TDM PBXs, the best alternative to connect the site to a SIP Trunking service is to replace the PBX.

Naturally, there are cost issues to consider. Deploying an on-premises IP-PBX requires capital outlay, and there are operational costs to cover installation, configuration, administration, and maintenance. New IP-based phones would also be needed.

Such an upgrade is a major investment. For larger offices, the costs might make sense given the communications savings that could be realized moving telecom traffic to an IP network. But some sites simply would not warrant such an upgrade.

**Select a provider that offers native PRI handoff:** The choices above have cost implications, a potential worker productivity impact, or other limitations. Sites where the PBX is not replaced or a gateway/IAD is installed cannot reduce their communications costs by virtue of having voice and data traffic collapsed onto a single, shared IP network. Additionally, these sites are denied access to the full benefits of IP voice and collaboration applications.

“Using SIP Trunking at a site with an IP PBX is pretty straightforward.”
This leads some companies to look for and evaluate another approach: To select a provider that offers native PRI handoff services. Rather than investing in more equipment to provide a TDM PBX site with access to SIP Trunking services, this approach would have the provider handle the conversion in its cloud.

Native PRI handoff lets companies take advantage of using an IP network for its voice traffic and offers the full benefits of SIP features without buying and configuring additional CPE. Additionally, a company might be able to leverage built-in resiliency of the provider’s network to ensure high availability communications for its large offices and sites.

Using a provider that offers native PRI handoff helps in several ways. There is no need to exclude sites from the IP network. Workers can take advantage of the full range of benefits and productivity enhancements offered in IP-based voice, UC, and collaboration applications. There is no need to install and maintain IADs, gateways, or to replace TDM PBXs.

**Level 3 as your technology partner**

A SIP Trunking service provider that supports native PRI handoff helps companies include more sites with legacy PBXs into an IP communications network. This can help cut communications costs, increase reliability and uptime, lower risk, reduce administrative burdens, and increase productivity by leveraging the benefits of IP communications and collaboration applications.

One provider that delivers the SIP features needed in companies today is Level 3.

Level 3® Voice Complete includes SIP Trunking with native PRI handoffs performed in its cloud. Consequently, SIP can be delivered directly to any of those lingering legacy TDM PBXs without added endpoint gateway equipment. Providing the handoff in Level 3’s cloud enables failover of SIP signal trunk groups or PRI trunk groups.

Additionally, Level 3 Voice Complete delivers the benefits of SIP technology, complemented with a rich set of features for flexible, reliable communications.

The Level 3 network has the efficiency and scalability to provide a highly available system with true nationwide coverage.

With other carriers, a company might need to develop its own disaster recovery and business continuity strategy. Multiple paths must be selected, additional equipment put in place, and processes worked out to ensure uptime. Level 3 Voice Complete service builds in
enterprise-class BCDR. It offers network diversity and routing failover.

An additional point to note is that E-911 is included. Level 3 offers 911 services for end-user service locations within its voice footprint. The offering also supports nomadic users for Microsoft Lync voice and UC deployments. The solution takes advantage of Level 3’s SIP Trunking and E-911 networks, enabling Microsoft Lync users to move throughout the workplace while still providing location-specific 911 information to the correct emergency responders.

With this combination of advanced features, Level 3 Voice Complete service provides the platform necessary to migrate fully to SIP with the assurance that DR requirements are being satisfied.

To learn more about the way Level 3 SIP Trunking with native PRI handoff can help connect sites with legacy PBXs, visit: http://www.level3.com/en/products-and-services/voice/enterprise-voice/sip-services/

“Native PRI handoff lets companies get the full benefits of SIP without buying additional CPE.”