Using SIP Trunking in a Legacy PBX or KTS Environment to Lower Costs

Technology Brief

The adoption of VoIP technology reflects the ongoing corporate mandate to provide a more efficient work environment at a lower cost. SIP trunking, a new delivery option for VoIP in the enterprise, can provide additional cost advantages by reducing charges on more types of phone service: local, long-distance, and international. SIP trunking also allows businesses to benefit from the multimedia capabilities of the SIP protocol and extend the ways they communicate with their business partners beyond basic voice in the future.

SIP Trunking's Market Appeal

SIP trunking can appeal to a broad market that encompasses large enterprises and small-to-medium-sized businesses (SMBs). Most businesses have separate voice and data services that are delivered over separate PSTN circuits, and broadband circuits configured for internet access. Often both the voice and data services are delivered via digital trunks (T1 circuits). The PSTN access circuits are configured for traditional voice protocols, PRI, or primary rate ISDN, while the data circuits are configured as bonded clear channel circuits with full symmetrical 1.544MB data transmission for broadband internet access.

SIP trunking can enable a business to reduce or eliminate circuits configured for PSTN access, and to leverage expanded broadband circuits to carry its voice traffic too.

Converging TDM lines can generally save about $700 to $1000 per month [PhonePlus] as a business only need pay for a T1/E1 line and the number of SIP trunks needed for voice calls. Some SIP trunk providers (often called Internet Telephony Service Providers or ITSPs) are advertising significant savings (50% to 70%) over current PSTN local and long distance rates. Along with access savings, support costs tend to be lower also because fewer lines are in use and need to be maintained.

The SIP Trunking Market Expands

When SIP trunking was first introduced, ITSPs focused on delivering the service to companies adopting IP-PBX technology equipped for direct support of SIP trunks, making IP-PBX adoption a viable transition point from PSTN to SIP trunking service. This kind of deployment (and market opportunity for SIP trunking) continues to grow as more organizations upgrade to IP-PBX systems when replacing legacy PBXs, but remains limited to those ready and willing to replace their phone systems.

As of this writing, a large part of the installed PBX market (estimated at more than 75%) still has traditional digital- or TDM-based PBX or Key Telephone System (KTS) equipment that is working well. Because the direct IP interconnection of SIP trunking to legacy equipment is not possible for these customers, the benefits of SIP trunking are less obvious to them. To reach these customers, ITSPs are actively working with experienced VoIP gateway providers whose equipment can help accelerate their growth by opening a market for them at companies with legacy PBX and KTS equipment.

If a VoIP gateway is used for SIP trunking, it is installed in front of the legacy phone system and provides a standard PSTN trunk interface to a company's phone system. The PBX is then programmed to route some or all calls over the SIP trunking provider network, allowing the provider to handle outbound local, long-distance, and international voice traffic for the customer. Each call travels most of the time on the SIP trunking provider network instead of on the PSTN and drops down to the PSTN "at the last mile," allowing dramatic cost savings.

In general, the initial investment in a VoIP gateway is not seen as cost prohibitive, and Return On Investment (ROI) can often be realized within months, depending on a customer's call volume. The investment also allows for considerable ongoing monthly savings beginning as soon as SIP trunk service is in place.

Inbound call service can be transitioned to an ITSP as well, facilitating a complete transition away from a PSTN service provider. Many ITSPs have developed a full suite of DID and toll-free services and remote exchange service, and they can help their customers leverage number portability legislation to ease a transition.

Technical Considerations

In this section, we will discuss some technical considerations concerning bandwidth, gateway and PBX interoperability, and Quality of Service (QoS).

Bandwidth

SIP trunks require approximately the same bandwidth as the traditional PSTN trunks they replace. SIP trunks running the G.711 coder (standard voice quality at 64 kbps) require a full Primary Rate Interface (PRI) span of clear channel service (1.544 mbps) to deliver 18 simultaneous voice calls, compared to 23 channels for a single PRI running a TDM protocol such as NI2 over the same link. If compression is raised using a G.729a coder, the number of simultaneous calls can be increased to about 28 per clear channel PRI span.

Generally, the ITSP can provision service over an existing public internet broadband link used by a business, but bandwidth requirements need to be considered. If an enterprise is already consuming all available broadband throughput for data, additional internet broadband access provisioning is necessary to use SIP trunking (and should be considered when calculating ROI).

Gateway and PBX Interoperability

If a VoIP gateway is used for SIP trunk service, few changes are necessary for PBX or KTS environments. The gateway emulates the PSTN Central Office trunk service, delivering T1/E1 PRI signaling and bearer channels to the PBX digital trunk ports. But if new routing tables are established based on a mix of SIP trunk services and PSTN services, then changes would be required.

Interoperability testing is important, and should be considered before a gateway is purchased. Normally vendors do extensive interoperability testing with leading PBX and KTS equipment, and with the access technology used by ITSPs because SIP implementations still vary considerably.

Availability

One benefit offered by SIP trunking is its transparency to end users who dial into the converged system in the same way as into a traditional system. Naturally, it is important for a converged service using SIP trunking to be as reliable as the system it replaces.

High levels of availability can be provided with a failover capability in the gateway. Failover allows calls to be routed over SIP trunks when they are available, and to backup PSTN trunks if they are not or if the broadband link that services the SIP trunks is unavailable. A gateway equipped with failover relays is required to provide failover availability, and a backup PSTN trunk service must also be maintained. Both of these items will increase costs and must be factored into a company's ROI calculations, but such additional capabilities are considered essential for certain business applications such as a customer service center.

Using a VoIP Gateway for SIP Trunking

The Dialogic® 2000 Media Gateway Series (DMG2000 Gateways) are very well-suited for use as VoIP gateways for SIP trunking. They have been thoroughly tested for interoperability with leading PBX and KTS equipment, are field-proven, and two current models in the series include failover technology. Dialogic is also in the process of testing the interoperability between the DMG2000 Gateways and the access technology used by leading ITSPs.

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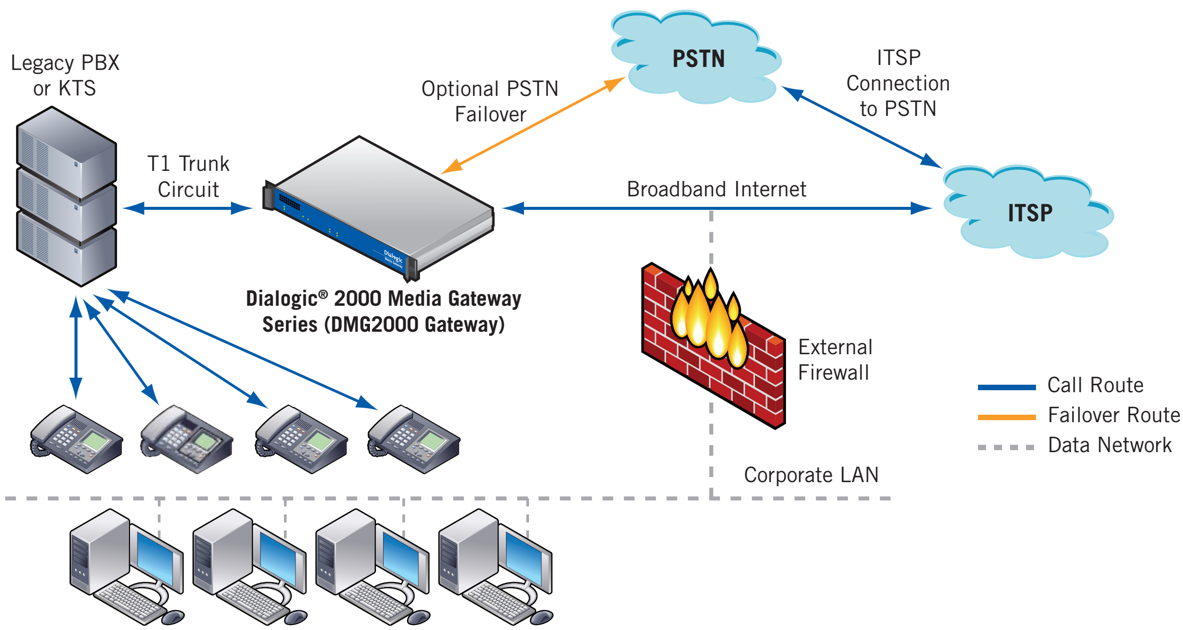

Figure 1. DMG2000 Gateway Enables SIP Trunking Service

In Figure 1, a DMG2000 Gateway with a failover capability is shown in a sample configuration in which it enables SIP trunking as a replacement for (or as a least-cost-routing option to) PSTN trunk service. The gateway is placed between a legacy PBX or KTS and the broadband internet connection provisioned by the ITSP to relay calls, and between the PBX or KTS and a backup PSTN line for failover. The PBX passes calls to the gateway in the same way as it would pass calls to its traditional PSTN Central Office T1/E1 trunk service, and the gateway routes these calls using SIP to the broadband internet connection provisioned by the ITSP. Failover relays in the gateway and additional T1/E1 ports provide a backup route to the PSTN if the SIP connection degrades or is unavailable.

More Information about the Dialogic® 2000 Media Gateway Series

For more information about gateways in the Dialogic 2000 Media Gateway Series (DMG2000 Gateways), visit http://www.dialogic.com/products/gateways/DMG2000.htm.

Reference

[PhonePlus] Figure cited in Tara Seals, "*SIPping into Something More Comfortable: SIP Trunking Offers Real SMB Opportunity*" (02/29/08) at http://www.phoneplusmag.com/articles/813/sip_trunking_offers_real_smb_opportunity.html

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