Connecting with SIP to the public phone system (PSTN)

Most enterprises start using VoIP as a replacement for the business PBX, with an emphasis on deploying phones and servers within the organization. However, at some point every VoIP network must “touch” the public switched telephone network (PSTN). This white paper outlines the steps needed to make that connection.

In a PBX environment, the PSTN connects to an organization’s private network using a well-defined interface controlled by the PSTN. In most cases, the interface is either an analog telephone line (sometimes called a plain old telephone service [POTS] line), an ISDN BRI or PRI circuit, or a channelized T1 line. These connections vary in their capabilities, but they all make the link between a private telephone network and the rest of the world.

With a SIP-based VoIP network, in contrast, it’s incumbent on the VoIP side to convert to one of the normal PSTN interfaces using some type of media gateway device. In effect, the VoIP network “pretends” to be a standard PBX and presents the same interface to the PSTN that a PBX would use.

Since most, if not all, enterprises using VoIP also have reliable high-speed Internet connectivity, a natural alternative to a media gateway is to connect to the PSTN using SIP itself. Rather than using older telephony interfaces, it is possible to deliver the VoIP call to the PSTN directly using SIP.

When connecting directly to the PSTN with SIP, enterprises have many alternatives to consider. Just as most companies have multiple PSTN carriers for different types of calls (long distance, international, local calling, toll-free service), there are many different options for direct SIP connection. The simplest way to think about direct PSTN connectivity is to divide the problem into outgoing and incoming calls.

Placing calls using SIP

Many VoIP service providers now offer outbound SIP call termination services, enabling organizations to pick and choose which service make the most sense. It’s possible, for example, to select different SIP providers for calls to specific parts of the world, or select a single provider that delivers more advanced features at a premium price.

The primary drivers for outbound SIP calling are cost reduction, flexibility, and scalability. Although long-distance carriers have dramatically reduced telecommunications costs, delivery of outbound calls using the Internet allows an enterprise to quickly and easily route calls based on the lowest possible cost. Because the long-distance provider installs no equipment, the barriers to change are also lowered. More importantly, the ability to deliver calls to a SIP service provider in any state or country using existing Internet infrastructure allows organizations to select different carriers, even if requirements change quickly.

The relatively low-bandwidth required by VoIP compared to other Internet applications also means that it is easy for enterprises to scale their calling, both up and down, as needed. By eliminating the need to provision facilities and install equipment every time calling patterns change, the enterprise gains a dramatic level of scalability and flexibility.

Receiving calls using SIP

Historically, enterprises have chosen local exchange carriers (Less) based on the geographic area where they happen to have a large physical presence. For example, if you’re in Arizona, you probably get your inbound phone service from the incumbent LEC, Qwest, or one of a few competitive LECs serving the area. These carriers provide local telephone numbers for inbound calling and deliver those calls via facilities provisioned between the LEC and the enterprise.

Using SIP call origination services over the Internet, you can select a provider that has telephone numbers anywhere in the world. This gives even small enterprises the ability to look like they have a presence in large cities, even if they’re in the suburbs or perhaps even in a completely different country. SIP call origination makes the world a smaller place, and gives enterprises of any size direct access to the world market.

Most US enterprises divide telephony service between local, intra-state, and inter-state carriers, with three or more carriers involved. In the same way, enterprises can choose between one or more Internet-based SIP carriers to provide different types of calling services.

Caveats

As with any new technology, SIP has its share of concerns. For example, providing emergency services (911, 112, 999, etc) can be very problematic. In the US, many SIP providers are required to provide E911 services. But until this is fully implemented, many providers simply inform their customers that emergency services are not available. While this may or may not be a problem, the issue still needs to be addressed by regulators and providers before SIP services can become completely relied upon. Several high-profile lawsuits have been filed recently, highlighting the importance that regulators and legislators put on a reliable 911 service, and this is an area that needs careful consideration by any enterprise considering an all-IP PSTN interconnect.

A second issue to consider when using outbound SIP is Calling Party Number (CPN), often referred to as "Caller*ID." CPNs for POTS-type calls are usually controlled by the LEC, beyond the control of small enterprises. Larger organizations with ISDN PRI circuits have always been able to set their own CPNs because of the capabilities within ISDN signaling.

SIP changes the CPN rules for organizations large and small. When connecting using SIP, a protocol element allows enterprises to set CPNs, although service provider restrictions may still apply. Depending on your point of view, the ability to set CPN to any arbitrary phone number may or may not be a good thing. Few people today control CPNs, giving rise to the illusion that CPN can positively confirm the identity of a caller.

Any time call delivery occurs over the Internet, the question of Quality of Service (QoS) comes up. Because media streams are highly susceptible to disruptions or delays in packet delivery, it is important to engineer Internet connections to prioritize VoIP traffic. At central sites with high-speed connections of DS-3 (T3) speed or faster, support for multiple simultaneous VoIP calls can usually be easily accommodated (depending, of course, on remaining bandwidth on the Internet connection). For smaller sites with DSL and cable-modem connections, jitter and unpredictable bandwidth availability can make it difficult to support multiple simultaneous business-class VoIP calls. Some type of bandwidth management function at the edge of the network can help alleviate this problem. Similarly, many firewalls and routers for small-office environments include QoS or bandwidth-management features can improve the experience of users running VoIP over the Internet.