Connecting with SIP to the public phone system

Most enterprises start using VoIP as a replacement for the traditional business PBX, and their emphasis is on deploying phones and servers within the enterprise network. However, at some point every VoIP network must "touch" the traditional telephone network, called the Public Switched Telephone Network (PSTN).

In a traditional PBX environment, the PSTN connects to the private company network using a well-defined interface controlled by the PSTN. In most cases, the interface is either an analog telephone line (sometimes called a POTS line, for "plain old telephone service," although there are many variations on these lines), an ISDN BRI circuit, or an ISDN PRI or channelized T1 circuit. These connections vary in their capabilities but all serve the function of making the link between a private telephone network and the rest of the world.

When a traditional PBX is replaced by a SIP-based VoIP network, it is usually 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 old interface to the PSTN that has always been used.

Since most, if not all, enterprises using VoIP also have high-speed and reliable Internet connectivity, a natural alternative to connecting using an on-premised media gateway is to simply connect to the PSTN using SIP itself. Rather than converting the VoIP side of the network to use 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, different, PSTN carriers for different types of calls (long distance, international, local calling, toll-free service), there are many different options in direct SIP connection. The easiest way to think about direct PSTN connectivity is to divide the problem into outgoing calls and incoming calls.

Placing Calls using SIP

There are many VoIP service providers now offering outbound SIP Call Termination services. Using SIP, you can select multiple providers that deliver the best rates for the specific parts of the world you call or you can select a single provider that delivers more advanced features at a premium price; the choice is yours.

The primary drivers for outbound SIP calling are cost reduction, flexibility, and scalability. Although long distance carriers have already reduced telecommunications costs dramatically, delivery of outbound calls using the Internet allows an enterprise to quickly and easily route calls based on the lowest possible cost. Because there is no installed equipment or telecommunications facility provided by the long distance provider, the barriers to change are also lowered, creating increased competition. More importantly, the ability to deliver calls to a SIP service provider in any state or country using existing Internet infrastructure allows the enterprise to select carriers based on a variety of factors, including cost, based on requirements that may change quickly.

The relative 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

Traditionally, enterprises pick their local exchange carrier (LEC) based on the geographic area they happen to have a large physical presence. If you're in Arizona, you usually get your inbound phone service from the incumbent LEC, Qwest, or one of a small number of competitive LECs that might be serving the area. These iLECs and cLECs 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. For enterprises, this means you can now look like you have a presence in a large city, even if you are located in the suburbs or perhaps even in a completely different country. SIP Call Origination makes the world a smaller place and allows any size enterprise direct access to the world market.

Most US enterprises traditionally divide their telephony service between local, intra-state, and inter-state, with 3 or more carriers providing the three services. In the same way, enterprises can choose between one or more Internet-based SIP carriers to provide different types of calling services.

Caveats

With new technology always comes with its share of concerns. For example, most SIP Providers do not (or cannot realistically) provide actual emergency services (911, 112, 999, etc). While this may or may not be a problem, because of cell phone and legacy telephone technology still being very available, the issue still needs to be addressed by both regulators and providers alike before SIP services can become completely relied upon. Several high-profile lawsuits have been filed recently that have highlighted 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 call delivery is Calling Party Number (CPN), often simply referred to as 'Caller*ID'. The CPN for a phone call is normally controlled by the LEC when a POTS-type call is presented to it, and small businesses don't care or control this. Larger businesses with ISDN PRI circuits have always been able to set their own CPN because of the capabilities within ISDN signaling. When connecting using SIP, a protocol element allows the business the capability to set their CPN, although service provider limitations and restrictions can 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. Because CPN is not under control of most people, an illusory penumbra of security has arisen around CPN as positively confirming the identity of a caller.

Any time that call delivery is made over the Internet, the question of Quality of Service (QoS) comes up. Because media streams are particularly susceptible to disruptions or delays in packet delivery compared to most other common Internet applications, it is important to carefully engineer the Internet connection to support business-class VoIP. At central sites with high-speed connections of DS3 (T3) speed or faster, support of 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, the jitter and predictability of the bandwidth may not be sufficient to consistently support multiple simultaneous business-class VoIP calls. Some type of bandwidth management function at the perimeter of the network can help to alleviate this problem, and many firewalls and routing devices designed for small office environments include QoS or bandwidth management features that can be used to improve the experience of users running VoIP over the Internet.