SIP Migration

Although many enterprises are eager to adopt SIP, they see legacy PBX systems as a roadblock to the cost savings and enhanced applications that SIP would bring. This paper outlines how to begin a phased migration to SIP without immediately replacing your existing PBX equipment.

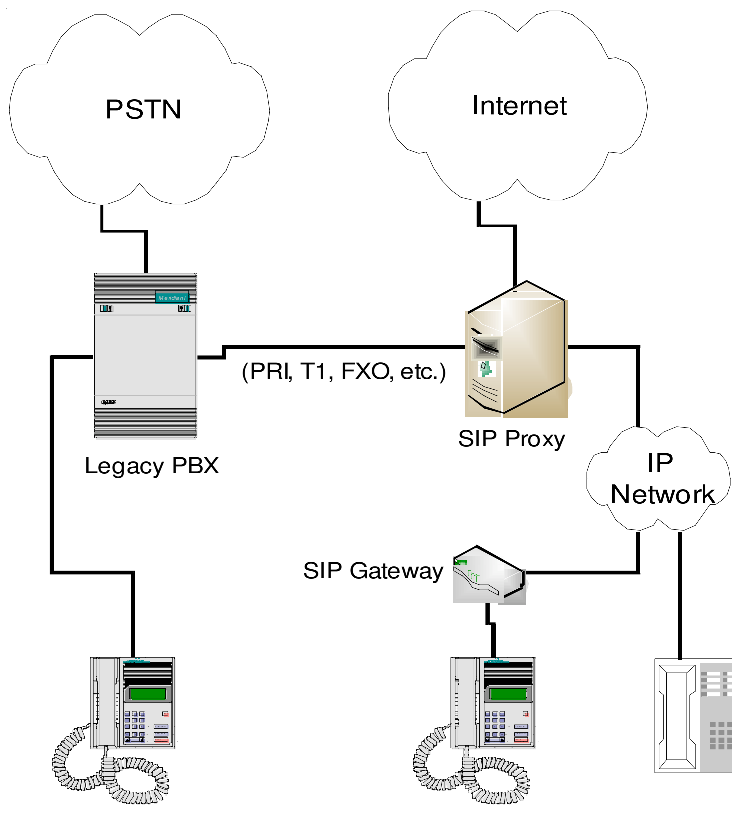
The cutover is a four-step process that enables an enterprise to migrate to a SIP based multimedia communications system while leveraging the existing investment in the public switched telephone network and not disappointing an installed based of enterprise users accustomed to a high level of voice quality and an extensive set of telephone options. These steps are:

1. Connecting a SIP proxy to an existing PBX
2. Migrating users and phones to the SIP proxy
3. Migrating your PSTN connection from your existing PBX to a SIP-based one
4. Retiring the PBX and the legacy phones.

Step 1.

This is where many enterprises stand today. In this mode of operation, the legacy PBX serves as the gateway to the traditional PSTN. To connect from the VoIP network to the PSTN, the SIP proxy routes any PSTN calls (as well as any calls to users remaining on the legacy PBX) over a high-capacity connection (such as a PRI ISDN line or a T1 line) to the legacy PBX.

This diagram demonstrates these methods. Initially you'll need to establish a connection from your new SIP-based call manager (or proxy server) to your legacy PBX-based call manager. You have several options to make this connection. You can use commercial standalone product, such as a SIP device with FXS/FXO ports or with T1 or PRI ISDN lines. Or you could use a board-level product that installs directly in a SIP proxy server with T1/PRI or FXO/FXS lines. Or, you may be able to make a SIP-based LAN connection between your SIP network and the legacy PBX with a built-in SIP/Ethernet option for the legacy PBX.



Step 2.

Now begins the migration of existing users and services to SIP. For example, you may first migrate all the handsets to SIP by either using SIP gateway devices or replacing the phones with native SIP handsets. A mixed approach is also common. For example, users who have "feature phones" might need the native SIP handset to keep their capabilities. Phones that didn't have features before can be easily connected to low or high-density SIP to traditional telephony adapters.

You'd then want to begin setting up SIP application servers that can perform some of the important features that previously resided in the PBX, such as voice mail and IVR functionality. Testing is needed to ensure that advanced applications, such as multi-party conferencing or paging, that were available on the PBX also function correctly with SIP devices.

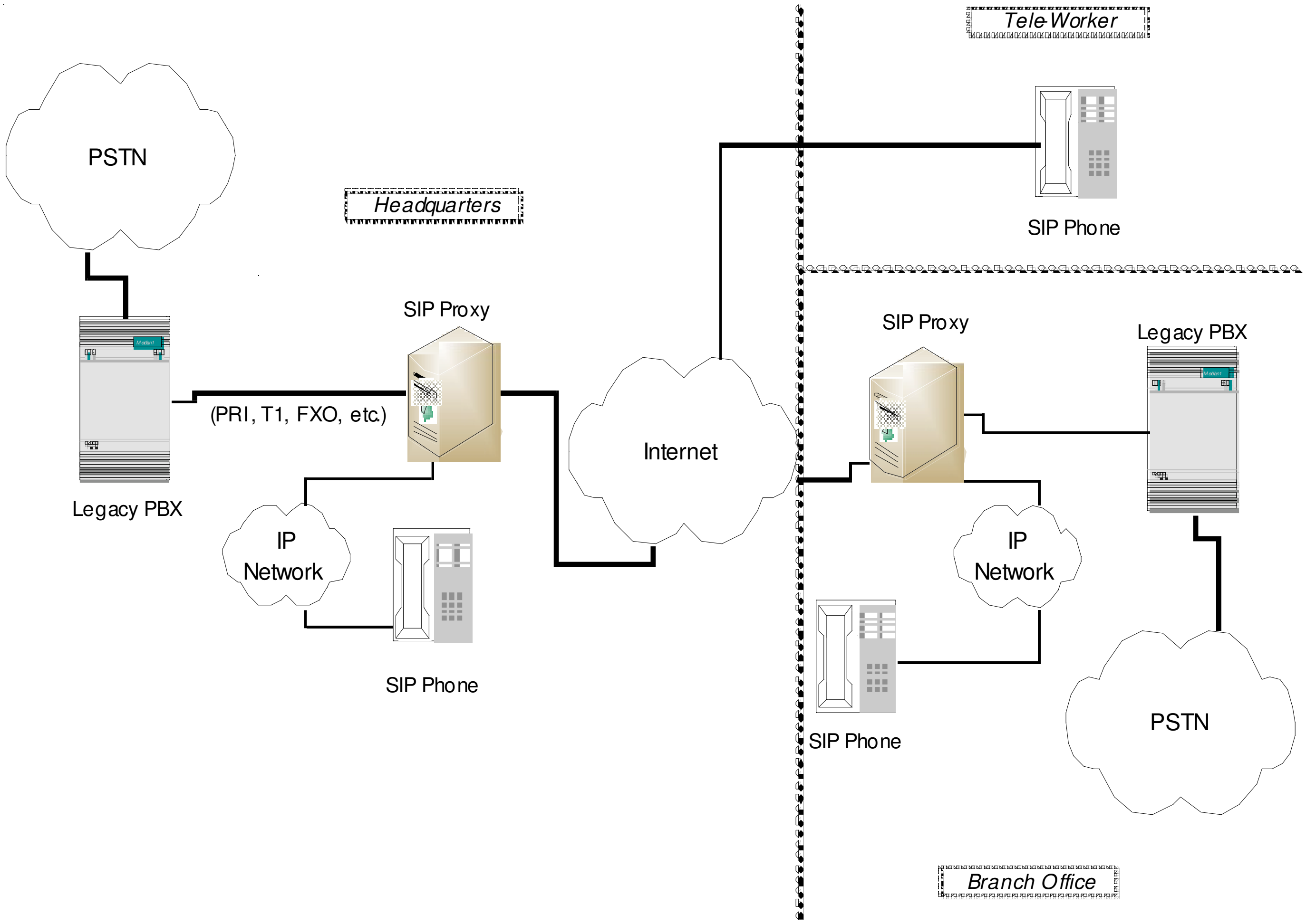
SIP Terms

SIP proxy – The SIP proxy provides the call routing functionality to your new phones. In this example the

SIP proxy is also functioning as a SIP gateway to the PBX, although they could easily be different devices.

SIP ATA – A SIP ATA (Analog Telephone Adapter, also known more properly as a SIP gateway) lets you use legacy telephones on the SIP network by converting calls from these phones into packetized voice traffic. For enterprises with large investments in handsets, this option provides a nice transition without having to re-educate the end-users. SIP ATAs from low-density 1-port devices up to devices that connect to 24 or more handsets with a single Ethernet connection.

SIP phone - Using a SIP-enabled phone gives you the most flexibility by letting you plug phones directly into your existing data network. SIP-enabled phones also can (and usually do) have more features than a simple analog phone, such as multiple line appearances, hold buttons, conferencing options, speaker phones, and other features common to “digital” phones on today’s business PBXs. SIP phones also let you more easily take advantage of the new SIP services offered. Initially the cost of these made this prohibitive for many enterprises; however that has changed considerably.



Step 3.

In Step 3, you will begin routing calls to the outside world through an Internet connection. More and more carriers are beginning to sell SIP-based long distance services at very appealing rates. However, the unpredictable nature of the Internet (with its high and variable jitter and error rates) along with security considerations may make this alternative impractical. Initially, you may choose to take a more conservative path and choose to use the Internet only for routing calls to your remote offices and your teleworkers, as shown in the diagram above, rather than using the public Internet for your PSTN calls.

Step 4.

Put the old PBX up for sale on eBay!