PSTN Options

There are multiple supported ways in FreePBX to connect your PBX to the PSTN (Public Switch Telephone Network): VoIP Trunks, Analog Cards, Digital Cards, and Gateways.

VoIP Trunks

SIP Trunks provide the most flexibility in call routing and redundancy. The FreePBX Project is funded in part by SIPStation.com service, so please consider that for your phone service needs.

The most popular type of VoIP Trunk is a **SIP trunk**. SIP (Session Initiation Protocol) establishes an internet or IP connection from your PBX to your provider. A SIP trunk cuts out expensive legacy analog trunks and typically allows you to purchase voice services at a much lower cost than can be provided by your local telco. On FreePBX systems, SIP is also the preferred method of connecting your PBX to your phones, and it enables you to easily add remote extensions. There are multiple other VoIP protocols, such as IAX, SCCP "Skinny", Skype, T.38, and Jingle, but for the most part you will not need to worry about these to set up your basic FreePBX. The FreePBX project offers a top-rated SIP Trunking Service, and choosing our service will help support the development of the project!

But if you must keep your voice services with your local provider, you will need to find out how those services are being delivered. The FreePBX Marketplace has Analog Cards, Digital Cards and Gateways available for purchase. If ordered with a FreePBX Appliance, your card will be shipped preinstalled.

Analog Cards or Digital Cards?

Analog Cards

Many smaller businesses receive their existing service through analog trunks. In most markets, these analog lines can be ported to SIP Trunk providers for a cost savings. If this isn't available, or you want to keep some analog lines on site for other services (alarms, emergency 911 call routing, external fax machines, modems etc.), then you will need an analog card. Analog lines are the traditional copper pairs coming from the local phone company. If using Analog lines, you will need to install an analog card in your server. Before you can build your system, you will need to find out how many incoming lines you have (FXO ports). If you are going to be using analog devices on your PBX, you will want to know how many devices you will need to connect (FXS Ports).

> **What is FXO vs. FXS?** FXS and FXO are the name of ports used by analog phone lines (**POTS**- Plain Old Telephone Service).

> **FXS**- **Foreign eXchange Subscriber** interface is the port that actually delivers the analog line to the subscriber/phone/endpoint. In other words, it is the 'plug on the wall' that delivers a dial tone, battery current and ring voltage. You can use it to connect analog phones and devices to your PBX and re-use your analog phones with the VoIP phone system.

> **FXO**- **Foreign eXchange Office** - To connect POTS lines to an IP phone system, you need an FXO port. This allows you to connect the incoming line from your local telephone company to the PBX and then translate the analog phone line to a VoIP call.

As an example, if you are setting up a PBX with four phone lines from your local telco, you would need to order a card with 4 FXO ports.

Digital Cards

A digital connection can put the same amount of call paths on a single pair of copper wires as you could get with dozens of pairs of copper wires running only analog services.
For larger installations, a popular delivery method of the local telco is to install a digital trunk where the audio is transmitted digitally instead of via analog. Depending on where you are in the world, there are multiple methods to deliver the digital connection. (Please note that T1/E1/J1 cards are capable of carrying voice and data, but for the purposes of setting up a PBX, we will only discuss voice services.)

In the United States these digital connections are called T1s. (In Europe they are E1s and in Japan they are J1s.) These provide voice service by digital channels. A Traditional T1 will provide 24 channels (E1 has 32). The providers then assign the channels (aka "call paths") using a level of service (BRI) or (PRI). The predominant service type in North America and Japan is PRI and will provide 23 call paths, as one channel is reserved for signaling (the setup and termination of calls between the PBX and the PSTN). Europe (and the rest of the world) gives access to 30 channels and one signaling channel. One time slot on the E1 is used for synchronization purposes.

A typical "digital" installation in the United States would include the purchase of a T1/PRI card, which would provide you with up to 23 call paths for use with your VoIP. Note that when purchasing a PRI from your local carrier, you would also need to purchase DID (Direct Inward Dialing) numbers. Usually these will be purchased in blocks of numbers. Since not all numbers will be in use at the same time, you can purchase more

DIDs than you have call paths, allowing you to provide more phone numbers to your system. Do an internet search on "Erlang" if you would like to know more about the mathematics behind implementing the number of call paths, endpoints and concurrent users.

Hardware Echo Cancellation?

Echo.Echo...ho...ho...oo! Echo is not traditionally a problem on traditional analog systems. However, once you bring VoIP into the mix, it can be a nightmare if not handled properly. VoIP systems often introduce latency, which analog systems do not have. VoIP systems will also be converting audio between different codecs and different types of systems. The result of this can be an echo in your conversation, which can be extremely frustrating to your end users. There are two primary ways you can combat this problem: software and hardware.   If using only software echo cancellation, all of the echo cancellation is processed by your PBX server's CPU. You can purchase cards with or without hardware-based echo cancellation. Hardware echo cancellation can be more successful, because it removes the burden of echo cancellation from the PC. Hardware echo cancellation is also advantageous when handling large call volumes or a high number of channels that would otherwise stress the CPU and result in the potential for poor audio quality.

Gateways

Gateways are dedicated devices that have any combination of PRI, FXS, FXO ports and convert them to SIP to be used by your PBX.