How Does A PBX Work?

The private branch exchange (PBX) is the system most often used by organizations today.

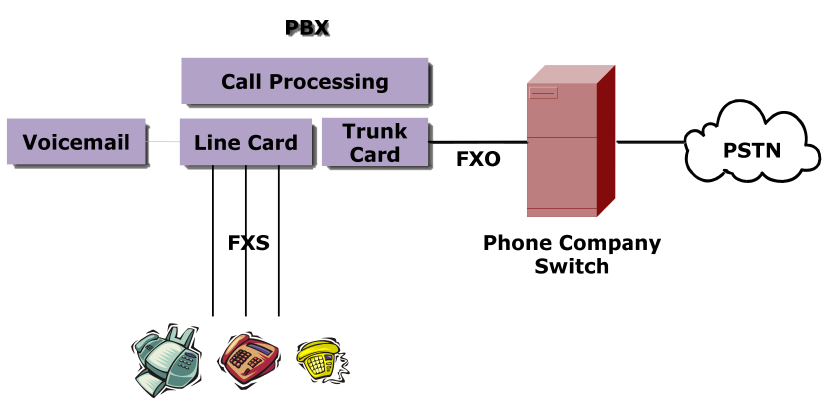
To understand how PBXs work, it is helpful to remember how calls were made before modern switches came along. Switchboard operators set up call between two phones. A call completed when the operator established a continuous circuit from the calling party to the called party. (This is why the phone system is often called a "switched" network.) Think of it as two tin cans connected by a string.

Switchboards consisted of a panel with cables and jacks. Operators plugged cables into plugs for each party. Setting up long distance or international calls was very complicated because operators had to establish continuous circuits across many different switchboards.

Call Switching, Processing and Signaling

Over time, operators were replaced by electronic switches. Switches do what the operators did, but they do it automatically.

Organizations operate PBXs, which are also switches. PBXs remember what users are doing at any moment and switch calls appropriately. The switch establishes a circuit between the parties, and the act of establishing this circuit is referred to as call processing or call control.



Call processing uses signaling protocols between the PBX and user handsets, adjacent PBXs, and the public switched telephone network (PSTN).

Customers moved to proprietary signaling because they wanted the features various PBX vendors offered. This made sense if customers received what they needed from only one vendor. However, the customer was locked into a permanent relationship with that vendor, and therefore could not take advantage of interoperability with products based on industry standards.

Line Interfaces

For a PBX to work, it needs to form a circuit between the called and the calling parties. There are two types of line interfaces for PBXs. Terminal line interfaces (also known as FXS) connect the PBX to telephone handsets. Trunk-line interfaces (also know an FXO), are used to connect the PBX to the central office (CO) exchange.

Terminal Line Cards

Telephone handsets connect directly to at least one port on a PBX card. Terminal line cards fall into two categories – analog and digital – and each supports only the corresponding analog or proprietary digital handsets.

Telephone Handsets

Styles of telephone handsets typically depend on a user's role in the organization. A manager might use a full-featured phone. Administrative assistants often require specialized multi-line sets and a broader set of telephony features.

Trunk Line Interfaces

Trunk interfaces are used to connect the PBX to the phone company and the PSTN, and enable communication with the outside world. Trunks can be analog (the so-called POTS, or plain old telephone service trunk) or digital (T-1, ISDN PRI in the U.S. and E-1 in Europe).

Trunk Features

Many trunk features are available. One feature that almost all organizations have is Caller Identification (Caller ID). Caller ID lets the called party see the calling party's name and/or telephone number before picking up the telephone handset, unless the calling party has specifically blocked this feature.

Another common feature is direct inward dial (DID). DIDs enable callers to contact users directly at unique phone numbers, without first contacting an automated attendant or operator. DIDs are assigned by the phone company to a customer and are routed to DID trunks connected to the PBX.

PBX systems can be connected to each other using either T-1 or analog interfaces. These interfaces were designed to interact with the phone company's switches; therefore, one of the PBX systems has to simulate CO signaling so that the two PBXs can communicate with each other.

Cabling

The cables pulled between telephone devices represent a significant portion of the investment in a phone system.

Category five (CAT5) twisted pair cable is popular because it can carry both voice and data traffic. The jack that links the cable to the desktop depends on whether the connection is to a telephone or a network device. The Ethernet NIC uses an RJ-45 plug and a standard analog telephone uses an RJ-11 plug.

The other end of the cable terminates close to the PBX, normally at a distribution frame. Distribution frames are rack-like structures where cables are threaded from one entry point to an appropriate exit point. The telephone engineer establishes the connection using a special-purpose tool that is used to push or punch the copper wire into a receiving contact. This is why distribution frames are sometimes referred to as "punch down blocks."

Dedicated phone networks increase reliability but decrease flexibility. Moves, adds and changes (MACs) in the PBX environment often require reconfiguring the wiring infrastructure. According to many telecom managers, 12 percent of users in a typical mid-size organization move, add, or change phones each year, at an average cost of $150 per MAC. MACs in a traditional PBX environment can be a significant, although often hidden, cost of ownership.

Basic Features and Functions

Telephone systems deliver certain features and functions. The following is a list of the typical features available to users and administrators.

- Call forward
- Call park
- Call pickup
- Call transfer
- Conference
- Hunt groups
- Last number redial
- Music or message on hold
- Mute call button
- Redials
- Shared line ringing
- Speakerphone

Value-added features are often included with telephone handsets to encourage customers to upgrade. Handsets therefore represent a large portion of the overall cost of owning a PBX. The feature lists associated with these handsets are fairly similar from one vendor to the next. Adding features along with a new handset requires significant engineering to the PBX, which increases costs.

Voicemail

Voicemail is not always integrated within the PBX but instead is an adjunct that resides outside the chassis and linked via several line interfaces. The cost of such applications is beyond the basic PBX purchase and significantly increases the price of the overall system.

Call Flows and Dial Plans

One of the most important decisions is how calls will be routed when the called party is not available. Will calls be transferred to an auto-attendant, an operator, an assistant, an off-site number, pager or mobile phone?

In evaluating call routing policies, it is essential to seek input from users, and especially high volume users and groups. Hunt groups and workgroups will often need to be defined for service centers and customer representatives.

(The term "hunt group" describes the way a call might be handled by the phone system. For example, if a call is not answered by a customer agent after a few rings, it will be forwarded to the next available phone in the agent group until it is picked up. If the call reaches the end of the available extensions without being picked up, it may be passed on to the group's voicemail.)

A call handling process also needs to be carefully planned out for outbound calls in such a way that, for any number dialed, a corresponding route is available for the call. Dial plans can become quite complex.

Automated Attendant

Auto-attendants provide a customizable way to route for incoming callers to the people they need. The application uses in-band signaling, called Dual Tone Multi-Frequency (DTMF) codes. DTMF assigns a certain sound frequency to each telephone key, so that when the dial pad keys are pressed, the auto-attendant "hears" these frequencies, interprets the information contained in these frequencies, and acts on that information.

For small businesses, the advantage of having an auto-attendant is the cost saving of not having to pay an operator. However, this feature can frustrate callers if the menu levels get too deep.

Comparing features of auto-attendant applications usually involves addressing such factors as:

- The number of menu levels that are needed

- Different menus depending on time of day and year
- Automatic updating of menus on a particular time/date
- How the system handles incorrect user input
- Whether users can go straight to a destination
- Whether users can bypass a long prompt or are they forced to wait
- Does the system provide directory search with name lookup
- Can calls be forward to workgroups or call center agents?

Call Detail Records and Billing

PBX systems typically generate detailed logs of calls on the system. These call logs can be saved to file for processing and analysis.

Because PBXs are isolated from the IT infrastructure, the generated call detail information is typically fed into a report engine that produces reports by department, by group, by usage cost.

This information can be used to answer questions such as the following:

- What calls are being made outside office hours and where are calls being placed?
- Which extensions are costing the organization the most money?
- What are the phone usage costs by department?

Caller Line Identification can be used to determine the duration of calls from specific customers. This information can be useful for basic customer billing or service level review.

More detailed statistics require a call center type system. The process of generating such reports can be outsourced to third parties who take the basic PBX data and convert it into useful reports. Some service organizations (legal, advertising, etc.) that bill by the hour use such call detail records as input into customer billing systems.