How IP Telephony Works

by Jeff Tyson

If you regularly make long-distance phone calls, chances are you've already used **IP telephony** without even knowing it. IP telephony, known in the industry as **Voice-over IP (VoIP)**, is the transmission of telephone calls over a data network like one of the many networks that make up the Internet. While you probably have heard of VoIP, what you may not know is that many traditional telephone companies are already using it in the connections between their regional offices.


This person is using a computer to talk to a friend in another state.

In this article, you'll learn about VoIP and the technology that makes it possible. We'll talk about VoIP's major protocols, about the various services provided and the low-cost, often free software that allows you to take advantage of them.

But first, let's discuss the fundamental problem with existing telephone networks -- namely, their reliance on **circuit switching**.

Circuit Switching

Circuit switching is a very basic concept that has been used by telephone networks for over 100 years. What happens is that when a call is made between two parties, the connection is maintained for the entire duration of the call. Because you are connecting two points in both directions, the connection is called a **circuit**. This is the foundation of the **Public Switched Telephone Network** (PSTN).
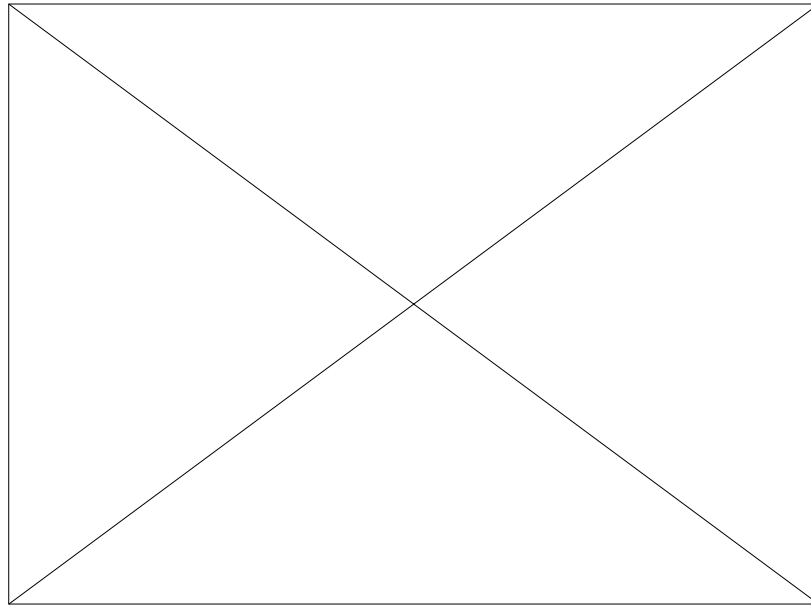
Click "Play" to see how circuit switching works.

Here's how a typical telephone call works:

1. You pick up the receiver and listen for a dial tone. This lets you know that you have a connection to the local office of your telephone carrier.
2. You dial the number of the party you wish to talk to.
3. The call is routed through the **switch** at your local carrier to the party you are calling.
4. A connection is made between your telephone and the other party's line, opening the circuit.
5. You talk for a period of time and then hang up the receiver.
6. When you hang up, the circuit is closed, freeing your line.

Let's say that you talk for 10 minutes. During this time, the circuit is continuously open between the two phones. Telephone conversations over the traditional PSTN are transmitted at a fixed rate of about 64 kilobits per second (Kbps), or 1,024 bits per second (bps), in each direction, for a total transmission rate of 128 Kbps. Since there are 8 kilobits (Kb) in a kilobyte (KB), this translates to a transmission of 16 KB each second the circuit is open, and 960 KB every minute it's open. So in a 10-minute conversation, the total transmission is 9600 KB, which is roughly equal to 9.4 megabytes (MB).

If you look at a typical phone conversation, much of this transmitted data is wasted. While you are talking, the other party is listening, which means that only half of the connection is in use at any given time. Based on that, we can surmise that we could cut the file in half, down to about 4.7 MB. Plus, a significant amount of the time in most conversations is **dead air** -- for seconds at a time, neither party is talking. If we could remove these silent intervals, the file would be even smaller.

Data networks do not use circuit switching. Your Internet connection would be a lot slower if it maintained a constant connection to the Web page you were looking at. Instead of simply sending and retrieving data as you need it, the two computers involved in the connection would pass data back and forth the whole time, whether the data was useful or not. That's no way to set up an efficient data network. Instead, data networks use a method called **packet switching**.

Packet Switching

While circuit switching keeps the connection open and constant, packet switching opens the connection just long enough to send a small chunk of data, called a **packet**, from one system to

another. What happens is this: The sending computer chops data into these small packets, with an address on each one telling the network where to send them. When the receiving computer gets the packets, it reassembles them into the original data.

Packet switching is very efficient. It minimizes the time that a connection is maintained between two systems, which reduces the load on the network. It also frees up the two computers communicating with each other so that they can accept information from other computers as well.
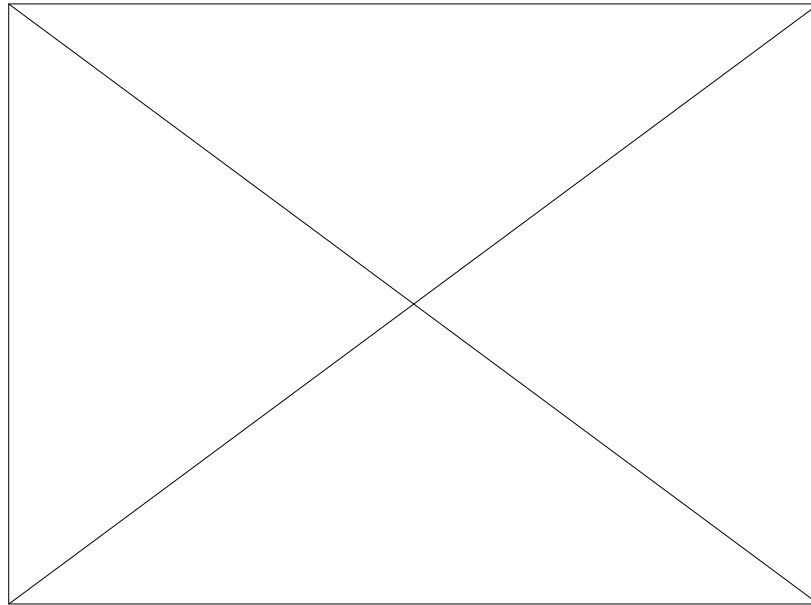
Click "Play" to see how packet switching works.

VoIP technology uses this packet-switching method to provide several advantages over circuit switching. For example, packet switching allows several telephone calls to occupy the amount of space occupied by only one in a circuit-switched network. Using PSTN, that 10-minute phone call consumed 10 full minutes of transmission time at a cost of 128 Kbps. With VoIP, that same call may have occupied only 3.5 minutes of transmission time at a cost of 64 Kbps, leaving another 64 Kbps free for that 3.5 minutes, plus an additional 128 Kbps for the remaining 6.5 minutes. Based on this simple estimate, another three or four calls could easily fit into the space used by a single call under the conventional system. And this example doesn't even factor in the use of data compression, which further reduces the size of each call.

Let's say that your company had equipment installed and a contract set up so that you can use VoIP. You have installed about a dozen telephones and a digital **private branch exchange** (PBX) in your office. A PBX is essentially a switch used to connect a number of phones (extensions) to each other and to one or more outside phone lines. In our example, the PBX is also a **gateway**.

Gateways are used to connect devices on two different types of networks so that they can communicate with each other. Our PBX is a gateway because it converts the standard circuit-switched signal from each phone into digital data that can be sent over a packet-switched, IP-based network. **IP** stands for Internet protocol, the language used by most data networks. Let's take another look at that typical telephone call, but this time using VoIP over a packet-switched network:

1. You pick up the receiver, which sends a signal to the PBX.
2. The PBX receives the signal and sends a dial tone. This lets you know that you have a connection to the PBX.
3. You dial the number of the party you wish to talk to. This number is then temporarily stored by the PBX.

4. Once you have entered the number, the PBX checks it to ensure that it is in a valid format.
5. The PBX determines whom to **map** the number to. In **mapping**, the number is attached to the IP address of another device called the **IP host**. The IP host is typically another digital PBX that is connected directly to the phone system of the number you dialed. In some cases, particularly if the party you are calling is using a computer-based VoIP client, the IP host is the system you wish to connect with.
6. A **session** is established between your company's PBX and the other party's IP host. This means that each system knows to expect packets of data from the other system. Each system must use the same protocol to communicate. The systems will implement two channels, one for each direction, as part of the session.
7. You talk for a period of time. During the conversation, your company's PBX and the other party's IP host transmit packets back and forth when there is data to be sent. The PBX at your end keeps the circuit open between itself and your phone extension while it forwards packets to and from the IP host at the other end.
8. You finish talking and hang up the receiver.
9. When you hang up, the circuit is closed between your phone and the PBX, freeing your line.
10. The PBX sends a signal to the IP host of the party you called that it is terminating the session. The IP host terminates the session at its end, too.
11. Once the session is terminated, the PBX removes the number-to-IP-host mapping from memory.

Probably one of the most compelling advantages of packet switching is that data networks already understand the technology. By migrating to this technology, telephone networks immediately gain the ability to communicate the way computers do. Of course, having the ability to communicate and understanding the methods of communication are two very different things. For telephones to communicate with each other and with other devices, such as computers, over a data network, they need to speak a common language called a **protocol**.

Protocols

There are two major protocols being used for VoIP. Both protocols define ways for devices to connect to each other using VoIP. Also, they include specifications for audio **codecs**. A codec, which stands for **co**der-**dec**oder, converts an audio signal into a compressed digital form for transmission and back into an uncompressed audio signal for replay.

The first protocol is **H.323**, a standard created by the **International Telecommunications Union** (ITU). H.323 is a comprehensive and very complex protocol. It provides specifications for real-time, interactive videoconferencing, data sharing and audio applications such as IP telephony. Actually a suite of protocols, H.323 incorporates many individual protocols that have been developed for specific applications.

H.323 Protocol Suite			
Video	**Audio**	**Data**	**Transport**
H.261	G.711	T.122	H.225
H.263	G.722	T.124	H.235
	G.723.1	T.125	H.245
	G.728	T.126	H.450.1
	G.729	T.127	H.450.2
			H.450.3
			RTP
			X.224.0

As you can see, full implementation of H.323 requires a lot of overhead. [Protocols.com: H.323](#) provides detailed information about the entire H.323 suite of protocols and how they relate to the

OSI Reference Model.

An alternative to H.323 emerged with the development of **Session Initiation Protocol (SIP)** under the auspices of the **Internet Engineering Task Force (IETF)**. SIP is a much more streamlined protocol, developed specifically for IP telephony. Smaller and more efficient than H.323, SIP takes advantage of existing protocols to handle certain parts of the process. For example, **Media Gateway Control Protocol (MGCP)** is used by SIP to establish a gateway connecting to the PSTN system. You can learn more about the architecture of SIP at Protocols.com: SIP.

Let's take a quick look at the various ways you can connect using VoIP.

Calling

There are four ways that you might talk to someone using VoIP. If you've got a computer or a telephone, you can use at least one of these methods without buying any new equipment:

- **Computer-to-computer** - This is certainly the easiest way to use VoIP. You don't even have to pay for long-distance calls. There are several companies offering free or very low-cost software that you can use for this type of VoIP. All you need is the software, a microphone, speakers, a sound card and an Internet connection, preferably a fast one like you would get through a cable or DSL modem. Except for your normal monthly ISP fee, there is usually no charge for computer-to-computer calls, no matter the distance.

The Net2Phone software client is easy to set up and use.

- **Computer-to-telephone** - This method allows you to call anyone (who has a phone) from your computer. Like computer-to-computer calling, it requires a software client. The software is typically free, but the calls may have a small per-minute charge.
- **Telephone-to-computer** - A few companies are providing special numbers or calling cards that allow a standard telephone user to initiate a call to a computer user. The caveat is that the computer user must have the vendor's software installed and running on his or her computer. The good news is that the cost of the call is normally much cheaper than a traditional long-distance call.
- **Telephone-to-telephone** - Through the use of gateways, you can connect directly with any other standard telephone in the world. To use the discounted services offered by several companies, you must call in to one of their gateways. Then, you enter the number you wish to call, and they connect you through their IP-based network. The downside is that you have to call a special number first. The upside is that the rates are typically much lower than standard long distance.

Although it will take some time to happen, you can be sure that, eventually, all of the circuit-switched networks will be replaced with packet-switching technology. IP telephony just makes

sense, in terms of both economics and infrastructure requirements. More and more businesses are installing VoIP systems, and the technology will continue to grow in popularity as it makes its way into our homes.