Overview of the PSTN

The ***Public Switched Telephone Network*** (PSTN) has been evolving ever since Alexander Graham Bell made the first voice transmission over wire in 1876.

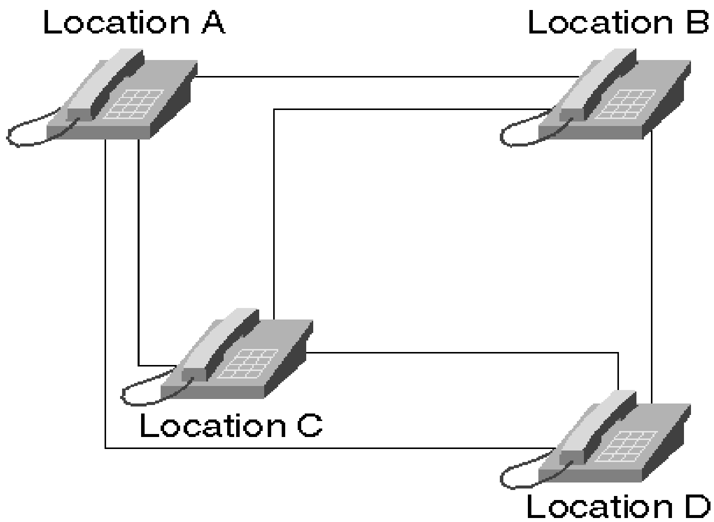
The Beginning of the PSTN

The first voice transmission, sent by Alexander Graham Bell, was accomplished in 1876 through what is called a ***ring-down*** circuit. A ring-down circuit means that there was no dialing of numbers; instead, a physical wire connected two devices. Basically, one person picked up the phone and another person was on the other end (no ringing was involved).

Over time, this simple design evolved from a one-way voice transmission, by which only one user could speak, to a bi-directional voice transmission, whereby both users could speak. The concept of dialing a number to reach a destination, however, did not exist at this time.

To further illustrate the beginnings of the PSTN, see the basic four-telephone network shown in Figure 1. As you can see, a physical cable exists between each location.

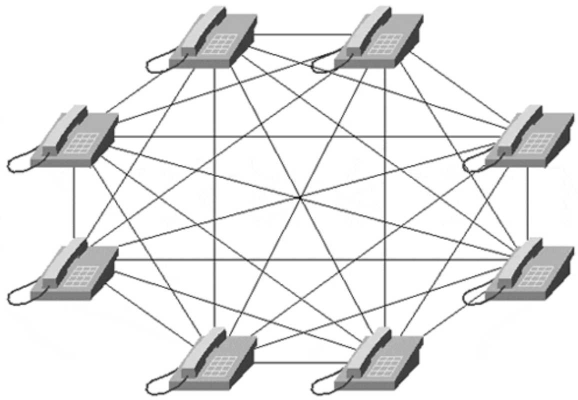
Figure 1. Basic Four-Phone Network



Place a physical cable between every household requiring access to a telephone, however, and you'll see that such a setup is neither cost-effective nor feasible (see Figure 2).

To determine how many lines you need to your house, think about everyone you call as a value of N and use the following equation: [N * (N–1)/2], As such, if you want to call 10 people, you need 45 pairs of lines running into your house.

Figure 2. Physical Cable between All Telephone Users

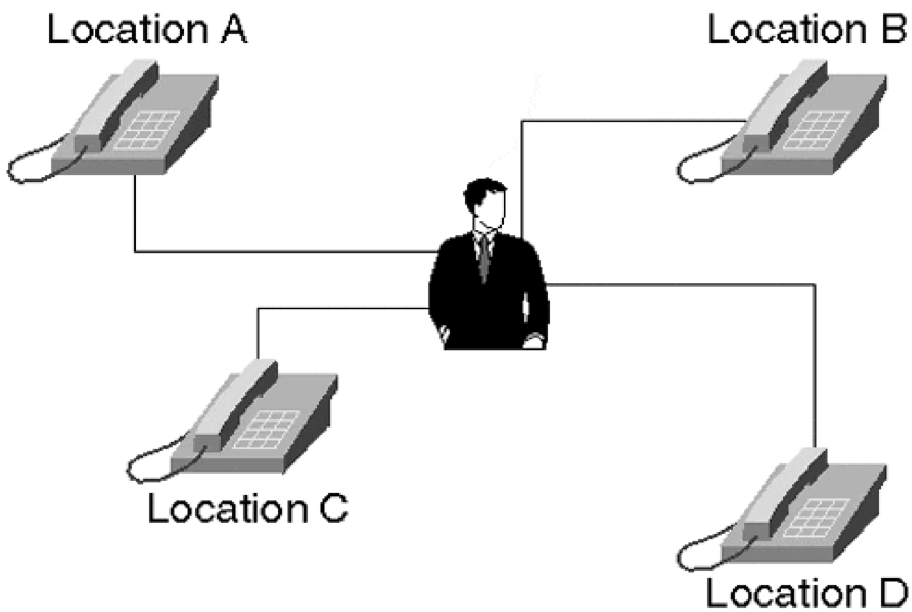


Due to the cost concerns and the impossibility of running a physical cable between everyone on Earth who wanted access to a telephone, another mechanism was developed that could map any phone to another phone.

With this device, called a ***switch***, the telephone users needed only one cable to the centralized switch office, instead of seven.

At first, a telephone operator acted as the switch. This operator asked callers where they wanted to dial and then manually connected the two voice paths. Figure 3 shows how the four-phone network example would look today with a centralized operator to switch the calls.

Figure 3. Centralized Operator: The Human Switch



Now, skip ahead 100 years or so—the human switch is replaced by electronic switches. At this point, you can learn how the modern PSTN network is built.

Understanding PSTN Basics

Although it is difficult to explain every component of the PSTN, this section explains the most important pieces that make the PSTN work. The following sections discuss how your voice is transmitted across a digital network, basic circuit-switching concepts, and why your phone number is 10 digits long.

Analog and Digital Signaling

Everything you hear, including human speech, is in analog form. Until several decades ago, the telephony network was based on an analog infrastructure as well.

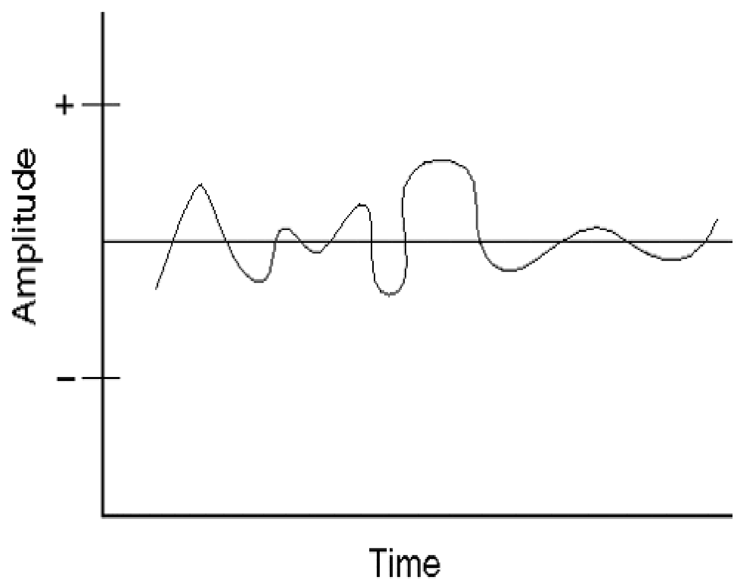
Although analog communication is ideal for human interaction, it is neither robust nor efficient at recovering from line noise.

In the early telephony network, analog transmission was passed through amplifiers to boost the signal. But, this practice amplified not just the voice, but the line noise as well. This line noise resulted in an often unusable connection.

Analog communication is a mix of time and amplitude.

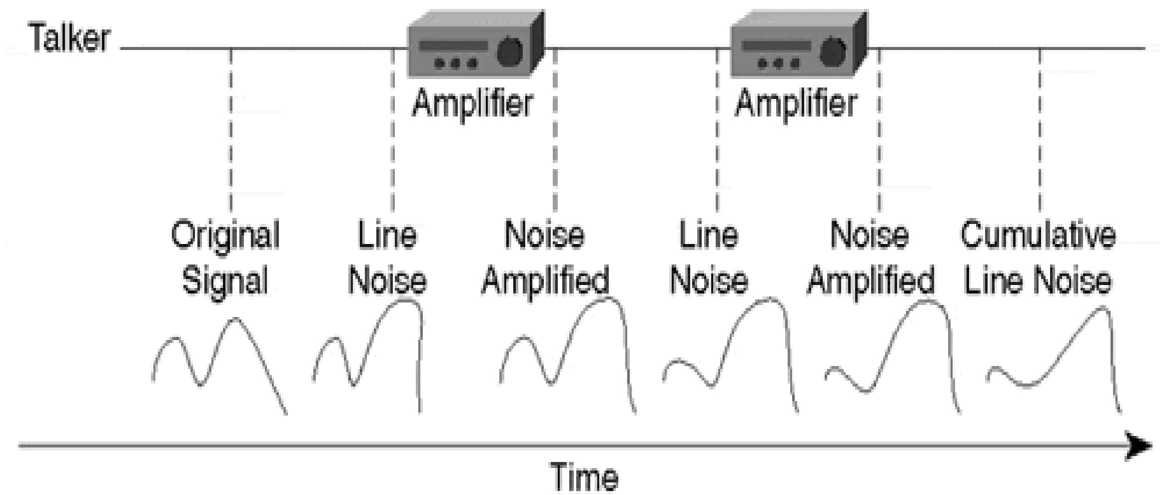
Figure 4, which takes a high-level view of an analog waveform, shows what your voice looks like through an oscilloscope.

Figure 4. Analog Waveform



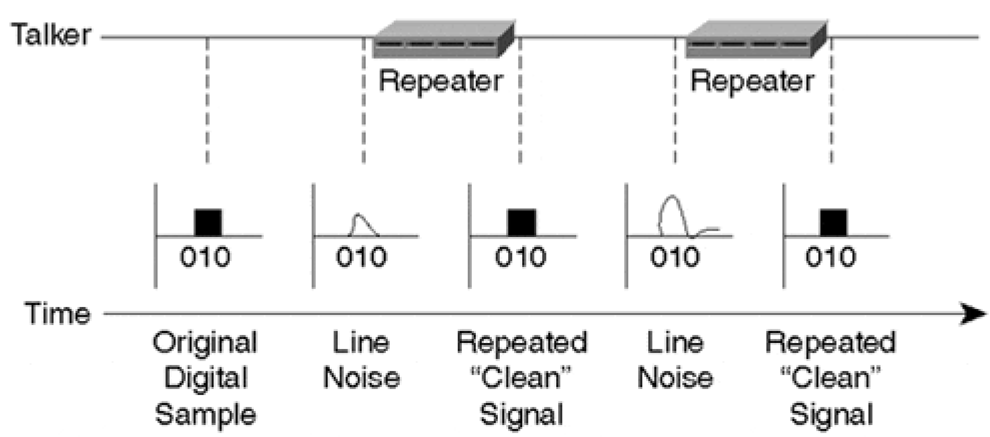
If you were far away from the ***end-office-switch*** (which provides the physical cable to your home), an amplifier might be required to boost the analog transmission (your voice). Analog signals that receive line noise can distort the analog waveform and cause garbled reception. This is more obvious to the listener if many amplifiers are located between your home and the end office switch. Figure 5 shows that an amplifier does not clean the signal as it amplifies, but simply amplifies the distorted signal. This process of going through several amplifiers with one voice signal is called ***accumulated noise***.

Figure 5. Analog Line Distortion



In digital networks, line noise is less of an issue because repeaters not only amplify the signal, but clean it to its original condition. This is possible with digital communication because such communication is based on 1s and 0s. So, as shown in Figure 6, the ***repeater*** (a digital amplifier) only has to decide whether to regenerate a 1 or a 0.

Figure 6. Digital Line Distortion



Therefore, when signals are repeated, a clean sound is maintained. When the benefits of this digital representation became evident, the telephony network migrated to ***pulse code modulation*** (PCM).

Digital Voice Signals

PCM is the most common method of encoding an analog voice signal into a digital stream of 1s and 0s. All sampling techniques use the ***Nyquist theorem***, which basically states that if you sample at twice the highest frequency on a voice line, you achieve good-quality voice transmission.

The PCM process is as follows:

- Analog waveforms are put through a voice frequency filter to filter out anything greater than 4000 Hz. These frequencies are filtered to 4000 Hz to limit the amount of crosstalk in the voice network. Using the ***Nyquist theorem***, you need to sample at 8000 samples per second to achieve good-quality voice transmission.
- The filtered analog signal is then sampled at a rate of 8000 times per second.
- After the waveform is sampled, it is converted into a discrete digital form. This sample is represented by a code that indicates the amplitude of the waveform at the instant the sample was taken. The telephony form of PCM uses eight bits for the code and a logarithm compression method that assigns more bits to lower-amplitude signals.

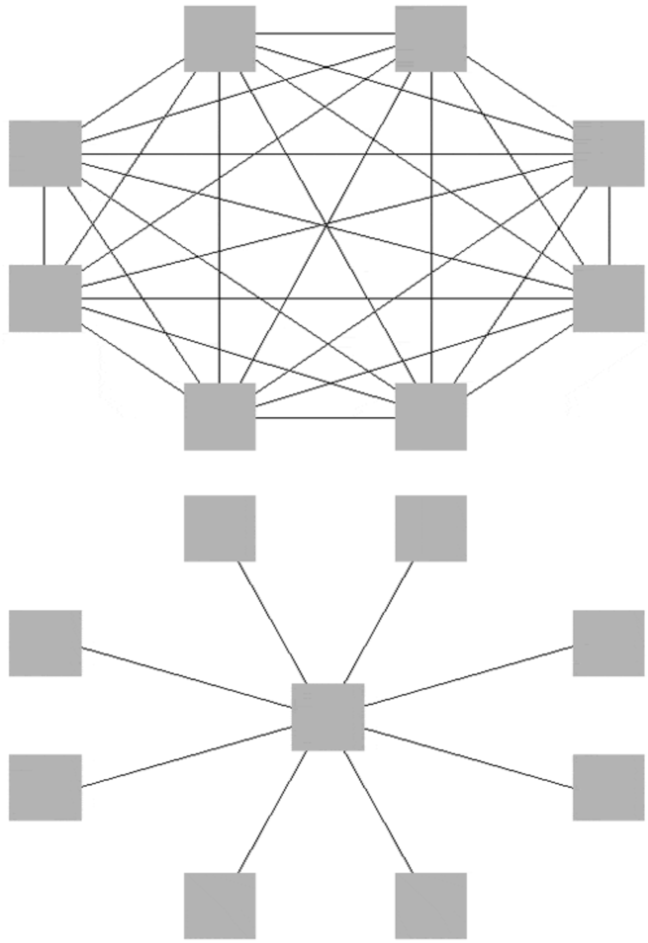
If you multiply the eight-bit words by 8000 times per second, you get 64,000 bits per second (bps). The basis for the telephone infrastructure is 64,000 bps (or 64 kbps).

Local Loops, Trunks and Inter-switch Communication

The telephone infrastructure starts with a simple pair of copper wires running to your home. This physical cabling is known as a ***local-loop***. The local loop physically connects your home telephone to the central office switch (also known as a ***Class-5-switch*** or ***end-office-switch***). The communication path between the central office switch and your home is known as the ***phone-line***, and it normally runs over the local loop.

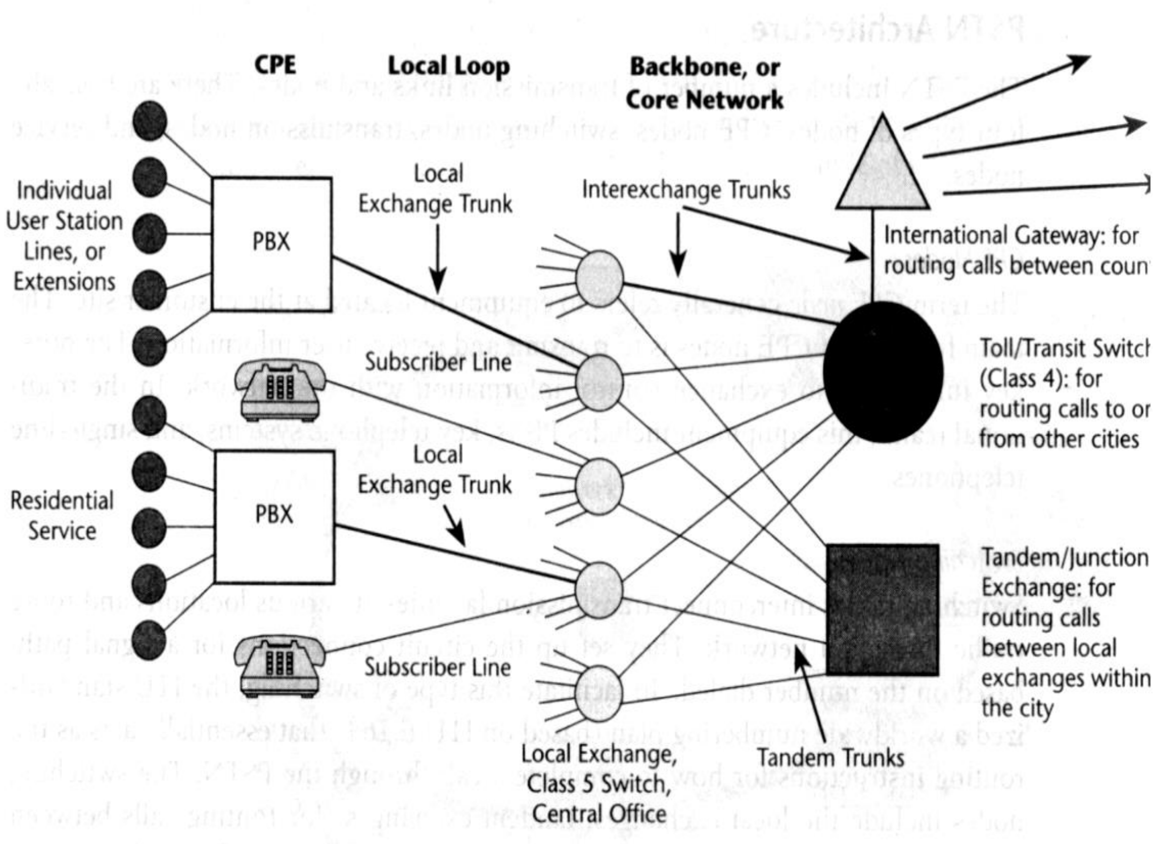
The communication path between several central office switches is known as a ***trunk***. Just as it is not cost-effective to place a physical wire between your house and every other house you want to call, it is also not cost-effective to place a physical wire between every central office switch. You can see in Figure 7 that a meshed telephone network is not as scalable as one with a hierarchy of switches.

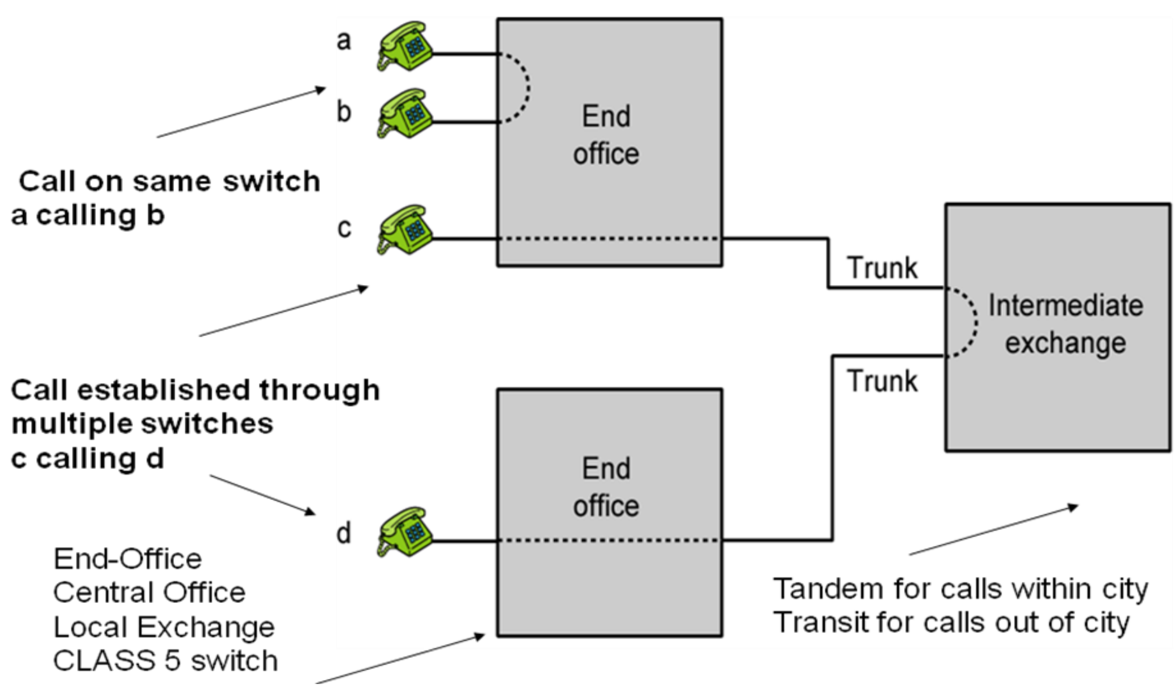
Figure 7. Meshed Network Versus Hierarchical Network

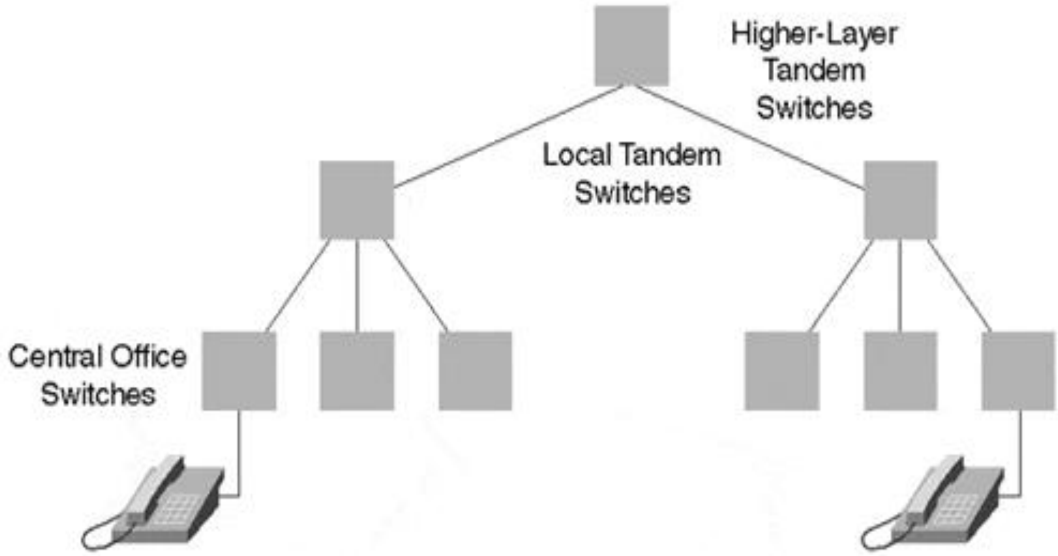


Switches are currently deployed in hierarchies. End office switches (or central office switches) interconnect through trunks to ***tandem switches*** (also referred to as ***Class-4-switches***). Higher-layer tandem switches connect local tandem switches. Figure 8 shows a typical model of switching hierarchy.

Figure 8. Circuit-Switching Hierarchy







Central office switches can be directly connected to each other. Where the direct connections occur between central office switches depends to a great extent on call patterns. If enough traffic occurs between two central office switches, a dedicated circuit is placed between the two switches to offload those calls from the local tandem switches.

PSTN Signaling

Generally, two types of signaling methods run over various transmission media. The signaling methods are broken into the following groups:

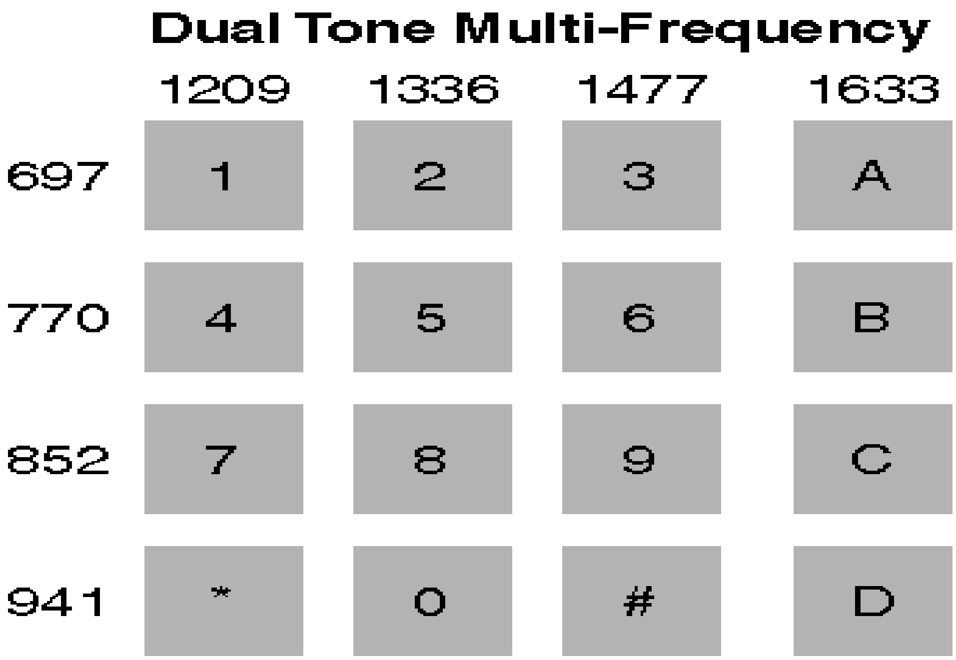
- **User-to-network signaling—** This is how an end user communicates with the PSTN.
- **Network-to-network signaling—** This is generally how the switches in the PSTN intercommunicate.

1. User-to-Network Signaling

Generally, when using ***twisted copper pair*** as the transport, a user connects to the PSTN through ***analog***, Integrated Services Digital Network ***(ISDN)***, or through a ***T1*** carrier.

The most common signaling method for user-to-network analog communication is ***Dual Tone Multi-Frequency (DTMF)***. DTMF is known as ***in-band*** signaling because the tones are carried through the voice path. Figure 9 shows how DTMF tones are derived.

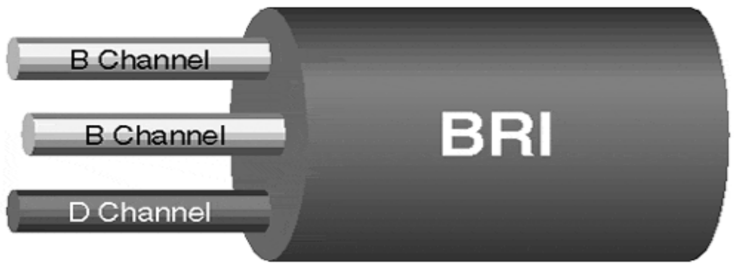
Figure 9. Dual Tone Multi-Frequency



When you pick up your telephone handset and press the digits (as shown in Figure 9), the tone that passes from your phone to the central office switch to which you are connected tells the switch what number you want to call.

Integrated System Digital Network **(ISDN)** uses another method of signaling known as ***out-of-band***. With this method, the signaling is transported on a channel separate from the voice. The channel on which the voice is carried is called a ***bearer*** (or B channel) and is 64 kbps. The channel on which the signal is carried is called a data channel (D channel) and is 16 kbps. Figure 10 shows a Basic Rate Interface (BRI) that consists of two B channels and one D channel.

Figure 10. Basic Rate Interface



Out-of-band signaling offers many benefits, including the following: Signaling is multiplexed, a lower post dialing delay, additional features, such as higher bandwidth, are realized, and call completion is greatly increased.

In-band signaling suffers from a few problems, the largest of which is the possibility for ***lost tones***.

2. Network-to-Network Signaling

Network-to-network communication is normally carried across the following transmission media:

- ***T1/E1*** carrier over ***twisted pair***

  T1 is a 1.544-Mbps digital transmission link normally used in North America and Japan.

  E1 is a 2.048-Mbps digital transmission link normally used in Europe.

- ***T3/E3, T4*** carrier over ***coaxial cable***

  T3 carries 28 T1s or 672 (64-kbps = T0) connections and is 44.736 Mbps.

E3 carries 16 E1s or 512 (64-kbps = E0) connections and is 34.368 Mbps.

T4 handles 168 T1 circuits or 4032 4-kbps connections and is 274.176 Mbps.

- ***T3, T4*** carrier over a ***microwave link***
- Synchronous Optical Network ***(SONET)*** across ***fiber*** media

SONET is normally deployed in ***OC-3***, ***OC-12***, ***OC-48*** and ***OC-768*** rates, which are 155.52 Mbps, 622.08 Mbps, 2.488 Gbps and 39.813 Gbps, respectively.

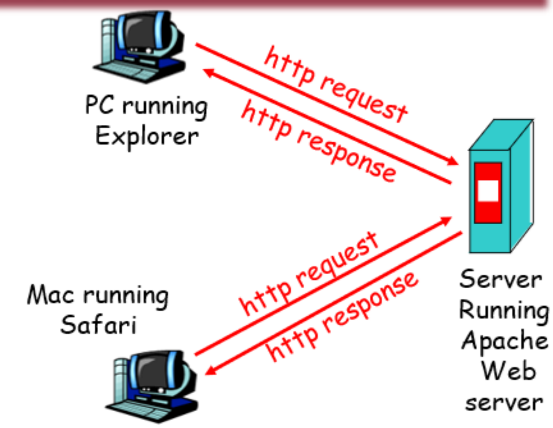
Network-to-network signaling types include ***in-band*** signaling methods such as Multi-Frequency ***(MF)*** and Robbed-Bit-Signaling ***(RBS)***. MF is similar to DTMF, but it utilizes a different set of frequencies. As DTMF signals from a home to an end office switch, but MF signals from switch to switch.

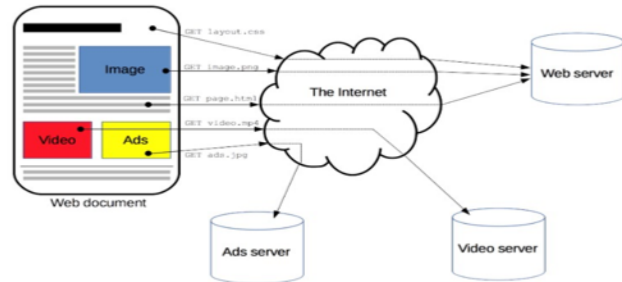
Network-to-network signaling also uses an ***out-of-band*** signaling method known as ***Signaling System 7*** (SS7) (or ***C7*** in European countries).

The Secret of the World Wide Web: the HTTP Standard

HTTP: hypertext transfer protocol

- WWW's application layer protocol
- client/server model
  - *client:* browser that requests, receives, and "displays" WWW objects
  - *server:* WWW server, which is storing the website, sends objects in response to requests
- http1.0: RFC 1945
- http1.1: RFC 2068
  - Leverages same connection to download images, scripts, etc.





The HTTP Protocol: More

http: TCP transport service:

- client initiates a TCP connection (creates socket) to server, port 80
- server accepts the TCP connection from client
- http messages (application - layer protocol messages) exchanged between browser (http client) and WWW server (http server)
- TCP connection closed

http is "stateless"

- server maintains no information about past client requests

Why?

Protocols that maintain session "state" are complex!

- past history (state) must be maintained and updated.
- if server/client crashes, their views of "state" may be inconsistent, and hence must be reconciled.
- RESTful protocols are stateless.

Lecture 1-33

HTTP Example

Suppose user enters URL www.cs.uiuc.edu/

(contains text, references to 10 jpeg images)

1a. http client initiates a TCP connection to http server (process) at www.cs.uiuc.edu. Port 80 is default for http server.

1b. http server at host www.cs.uiuc.edu waiting for a TCP connection at port 80. "accepts" connection, notifying client

2. http client sends a http ***request message*** (containing URL) into TCP connection socket

3. http server receives request messages, forms a ***response message*** containing requested object (index.html), sends message into socket

time

HTTP Example (cont.)

4. http server closes the TCP connection (if necessary).

5. http client receives a response message containing html file, displays html, Parses html file, finds 10 referenced jpeg objects

6. *Steps 1-5 are then repeated for <u>each</u> of 10 jpeg objects*

time

For fetching referenced objects, have 2 options:

- **non-persistent connection:** only one object fetched per TCP connection
  - some browsers create multiple TCP connections *simultaneously* - one per object
- **persistent connection:** multiple objects transferred within one TCP connection