Module I: Telecommunication Systems

EECE2141:

TELECOMMUNICATIONS FOR SOCIETY

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syllabus

Telecommunication Systems:

- Telephones
- Telephone System
- Facsimile
- Internet Telephony

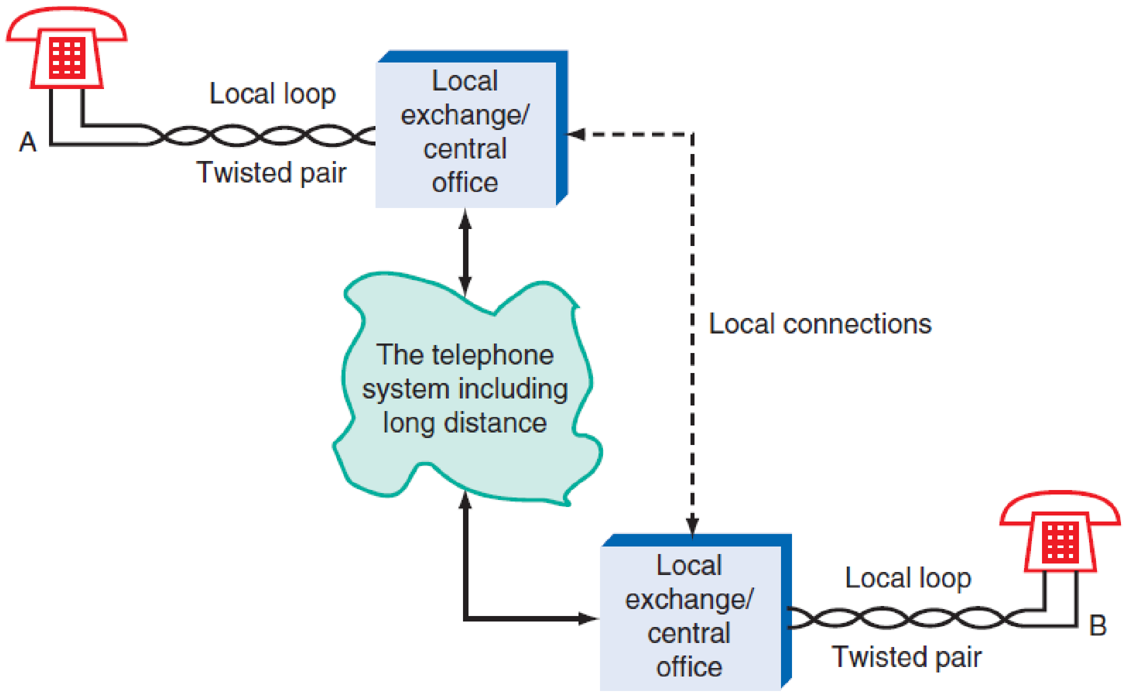
Telephones

- The original telephone system was designed for full duplex analog communication of voice signals
- Today, the telephone system is still primarily used for voice, but it employs mostly digital techniques, not only in signal transmission but also in control operations.
- unique identification code—the 10-digit telephone number
- The telephone system provides a means of recognizing each individual number and provides switching systems that can connect any two telephones

1.1 The Local Loop

- The two-wire, twisted-pair connection between the telephone and the central office is referred to as the local loop or subscriber loop.
- As many as 10,000 telephone lines can be connected to a single central office
- The circuits in the telephone and at the central office form a complete electric circuit, or loop.
- This single circuit is analog and carries both dc and ac signals.
- The dc power for operating the telephone is generated at the central office and supplied to each telephone over the local loop.
- The ac voice signals are transmitted along with the dc power.
- Full duplex operation, i.e., simultaneous send and receive, is possible.
- All dialing and signaling operations are also carried on this single twisted-pair cable.

The basic telephone system



1.2 Telephone Set

- A basic telephone or telephone set is an analog baseband transceiver.
- It has a handset that contains a microphone and a speaker, better known as a transmitter and a receiver.
- It also contains a ringer and a dialing mechanism. Overall, the telephone set fulfills the following basic functions.

The receive mode provides:

1. An incoming signal that rings a bell or produces an audio tone indicating that a call is being received
2. A signal to the telephone system indicating that the signal has been answered
3. Transducers to convert voice to electric signals and electric signals to voice.

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The transmit mode:

1. Indicates to the telephone system that a call is to be made when the handset is lifted

2. Indicates that the telephone system is ready to use by generating a signal called the dial tone

3. Provides a way of transmitting the telephone number to be called to the telephone system

4. Receives an indication that the call is being made by receiving a ringing tone

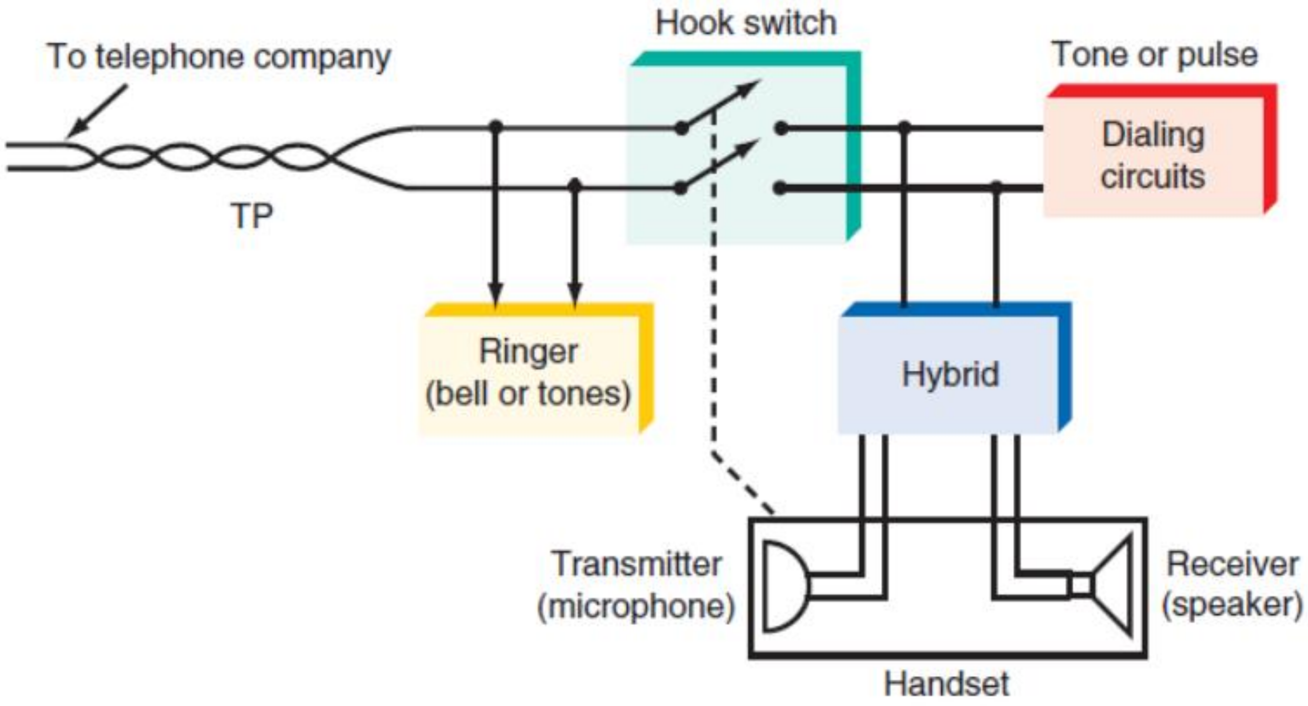
5. Provides a means of receiving a special tone indicating that the called line is busy

6. Provides a means of signaling the telephone system that the call is complete

- Some of the more advanced electronic telephones have other features such as multiple line selection, hold, speaker phone, call waiting, and caller ID.

- Fig. in the next slide is a basic block diagram of a telephone set.

- Detailed circuits for each of the blocks and their operation are described later when the standard and electronic telephones are discussed in detail.


To telephone company
TP
Hook switch
Tone or pulse
Dialing circuits
Ringer (bell or tones)
Hybrid
Transmitter (microphone)
Receiver (speaker)
Handset

Ringer:

- The ringer is either a bell or an electronic oscillator connected to a speaker.
- It is continuously connected to the twisted pair of the local loop back to the central office.
- When an incoming call is received, a signal from the central office causes the bell or ringer to produce a tone.

Switch Hook:

- A switch hook is a double-pole mechanical switch that is usually controlled by a mechanism actuated by the telephone handset.
- When the handset is "on the hook," the hook switch is open, thereby isolating all the telephone circuitry from the central office local loop.
- When a call is to be made or to be received, the handset is taken off the hook.
- This closes the switch and connects the telephone circuitry to the local loop.
- The direct current from the central office is then connected to the telephone, closing its circuits to operate.

Dialing Circuits:

- The dialing circuits provide a way for entering the telephone number to be called.
- In older telephones, a pulse dialing system was used.
- A rotary dial connected to a switch produced a number of on/off pulses corresponding to the digit dialed.
- These on/off pulses formed a simple binary code for signaling the central office.

DTMF:

- In most modern telephones, a tone dialing system is used. Known as the Dual-Tone Multi-Frequency (DTMF) system, this dialing method uses a number of push buttons that generate pairs of audio tones that indicate the digits called.

- Whether pulse dialing or tone dialing is used, circuits in the central office recognize the signals and make the proper connections to the dialed telephone.

Handset:

- This unit contains a microphone for the transmitter and a speaker for receiver.

- When you speak into the transmitter, it generates an electric signal representing your voice.

- When a received electric voice signal occurs on the line, the receiver translates it to sound waves.

- The transmitter and receiver are independent units, and each has two wires connecting to the telephone circuit. Both connect to a special device known as the hybrid.

Hybrid:

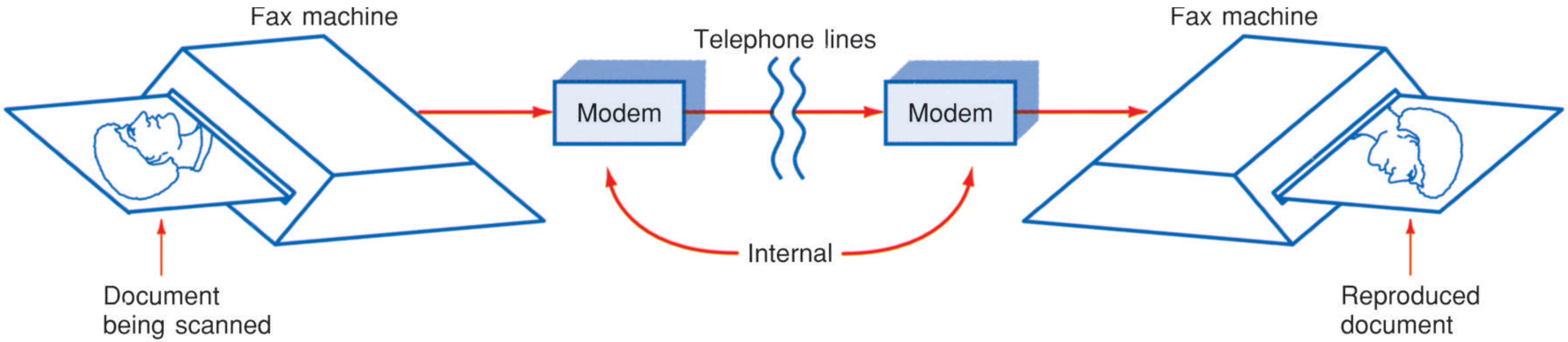
- The hybrid circuit is a special transformer used to convert signals from the four wires from the transmitter and receiver to a signal suitable for a single two-line pair to the local loop.

- The hybrid permits full duplex, i.e., simultaneous send and receive, analog communication on the two-wire line.

- The hybrid also provides a side tone from the transmitter to the receiver so that the speaker can hear her or his voice in the receiver. This feedback permits automatic voice-level adjustment.

Facsimile

- *Facsimile,* or *fax,* is an electronic system for transmitting graphic information by wire or radio.
- Facsimile is used to send printed material by scanning it and converting it to electronic signals that modulate a carrier to be transmitted over the telephone lines.
- Since modulation is involved, fax transmission can also take place by radio.
- With facsimile, documents such as letters, photographs, line drawings, or any printed information can be converted to an electric signal and transmitted with conventional communication techniques.
- The components of a fax system are illustrated in Fig. in the next slide.


Fax machine
Telephone lines
Fax machine
Modem
Modem
Internal
Document
being scanned
Reproduced
document

- Although facsimile is used to transmit pictures, it is not TV because it does not transmit sound messages or live scenes and motion.

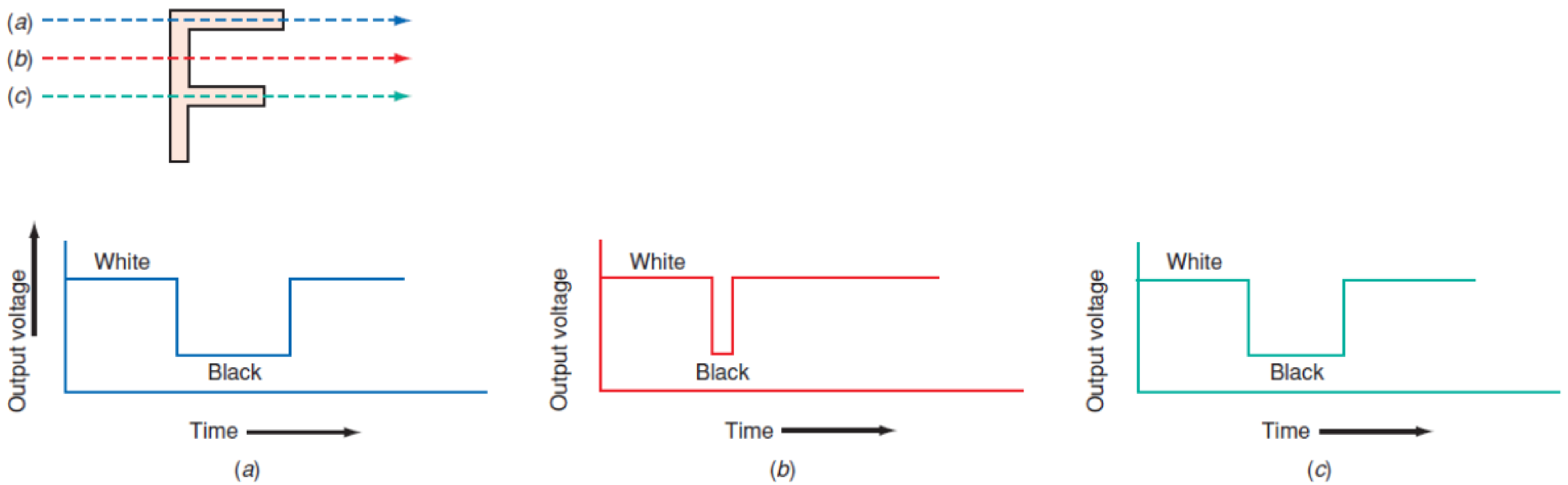
- However, it does use scanning techniques that are generally similar to those used in TV.

- A scanning process is used to break a printed document up into many horizontal scan lines that can be transmitted and reproduced serially.

How Facsimile Works

- The early facsimile machines scanned the document to be transmitted with a light and photocell arrangement.
- A scanning head consists of a light source and a photocell.
- A light source, focused to a tiny point with a lens system, was used to scan the document.
- The lens was also used to focus the reflected light from on the document onto the photocell.
- As the light scanned the letters and numbers in a typed or printed document or the gray scale in a photograph, the photocell produced a varying electronic signal whose output amplitude was proportional to the amount of reflected light.
- This baseband signal was then used to amplitude- or frequency-modulate a carrier in the audio frequency range.
- This permitted the signal to be transmitted over the telephone lines.

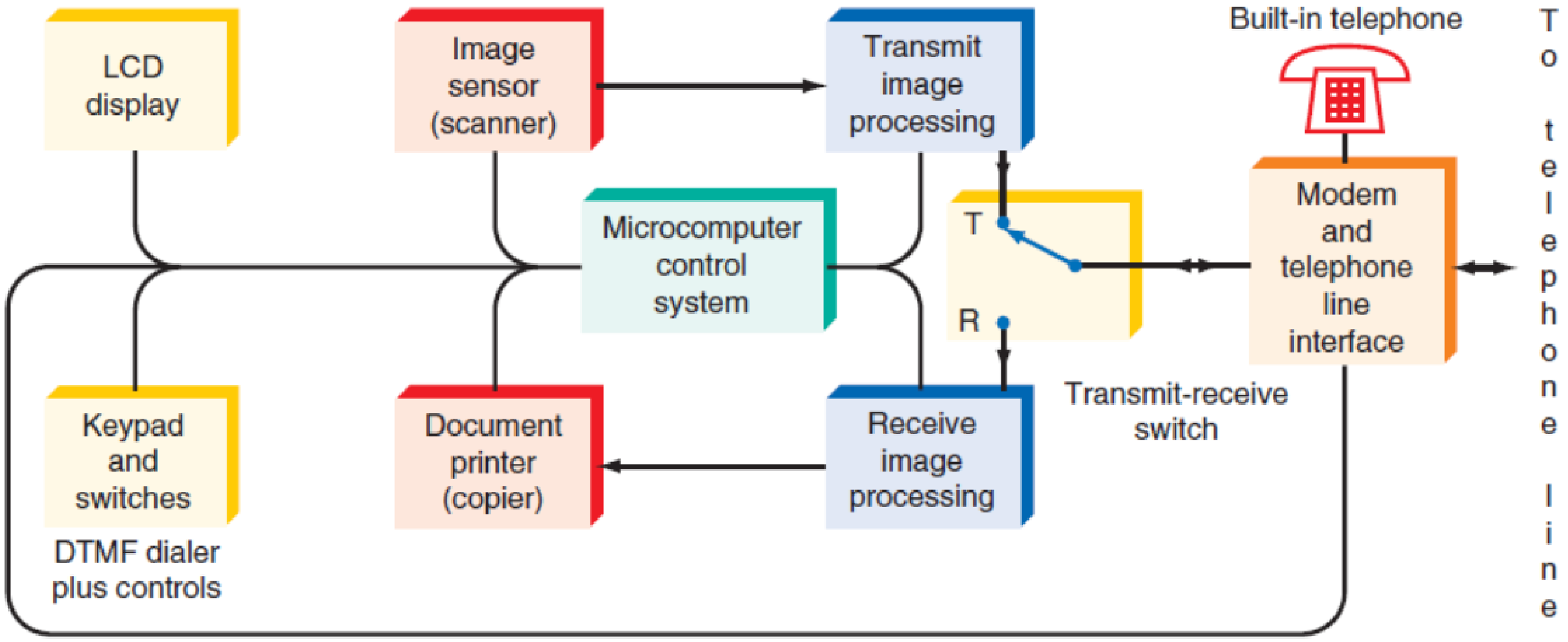
Output of a photosensitive detector during different scans.



- On the receiving end, a demodulator recovered the original signal information, which was then applied to a stylus.
- The purpose of the stylus was to redraw the original information on a blank sheet of paper.
- A typical stylus converted the electric signal to heat variations that burned the image into heat-sensitive paper.
- Today’s modern fax machine is a high-tech electro optical machine.
- Scanning is done electronically, and the scanned signal is converted to a binary signal.
- Then digital transmission with standard modem techniques is used.

Modern fax machine

Block diagram of modem fax machine.



- The transmission process begins with an image scanner that converts the document to hundreds of horizontal scan lines.
- Many different techniques are used, but they all incorporate a photo- (light-) sensitive device to convert light variations along one scanned line into an electrical voltage.
- The resulting signal is then processed in various ways to make the data smaller and thus faster to transmit.
- The resulting signal is sent to a modem where it modulates a carrier set to the middle of the telephone voice spectrum bandwidth.
- The signal is then transmitted to the receiving fax machine over the public switched telephone network.

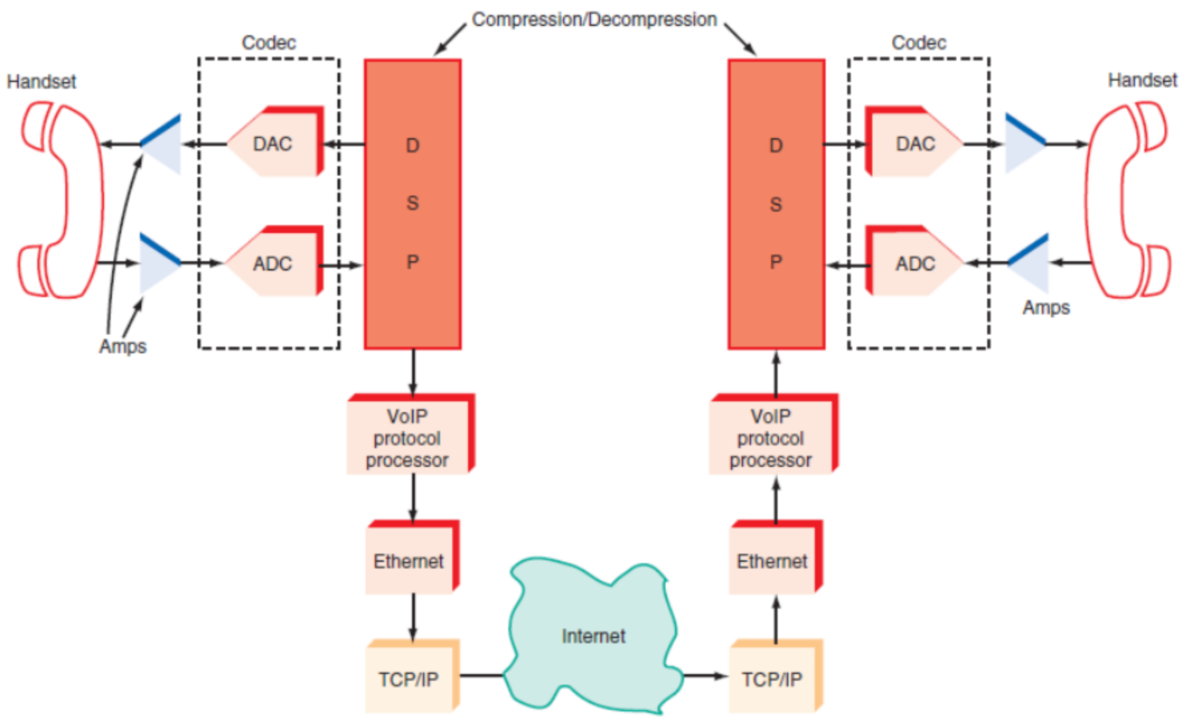
- The receiving fax machine's modem demodulates the signal that is then processed to recover the original data.
- The data is decompressed and then sent to a printer, which reproduces the document.
- Because all fax machines can transmit as well as receive, they are referred to as transceivers.
- The transmission is half duplex because only one machine may transmit or receive at a time.
- Most fax machines have a built-in telephone, and the printer can also be used as a copy machine.
- An embedded microcomputer handles all control and operation, including paper handling.

Internet Telephony

VoIP, or Voice over Internet Protocol, is a technology that allows you to make phone calls using a broadband internet connection instead of traditional analog phone lines. It converts your voice into digital data packets that are transmitted over the internet. This process is essentially the same as sending an email or a web page, but it's optimized for real-time communication

1. Conversion and Digitization
2. Packetization and Compression
3. Transmission over the Internet
4. Reassembly and Decoding
5. Playback

Signal flow in VoIP system



Telecommunication Systems

- VoIP, in effect, for the most part, bypasses the existing telephone system, but not completely.
- VoIP is a highly complex digital voice system that relies on high-speed Internet connections from cable TV companies, phone companies supplying DSL, and other broadband systems including wireless.
- It uses the Internet's vast fiber-optic cabling network to carry phone calls without phone company charges.
- This new telephony system is slowly replacing traditional phones, especially in large companies.
- It offers the benefits of lower long-distance calling charges and reduces the amount of new equipment needed, because phone service is essentially provided over the same local-area network (LAN) that interconnects the PCs in an organization.
- VoIP is rapidly growing in use and in the future is expected to replace standard phones in many companies and homes.

- A relatively wide bandwidth is needed to transmit this bit stream (64 kHz or more).
- To reduce the data rate and the need for bandwidth, the bit stream is processed by a voice encoder that compresses the voice signal. This compression is usually done by DSP either in a separate DSP processor chip or as hardwired logic on a larger chip.
- The output is at a greatly reduced serial digital data rate
- The type of compression used is determined by International Telecommunications Union standards.
- Various mathematical algorithms beyond the scope of this text are used.

- The 64-kbps digital signal is designated as standard G.711 and is better known as pulse-code modulation (PCM).
- Standard G.729a is the most common compression standard used and results in an 8-kbps digital voice signal.
- Another popular standard is G.723, which produces an even more highly compressed 5.3-kbps signal at the expense of some voice quality.
- Numerous other compression standards are used, and they are selected based upon the application.
- Most VoIP phones contain all the common compression standard algorithms in the DSP memory for use as called for.
- The signal is also processed in the DSP to provide echo cancellation, a problem in digital telephony.

- The resulting serial digital signal is put into a special packet by a microcomputer processor running a VoIP protocol and then transmitted by Ethernet over a LAN or via a high-speed Internet connection such as is available from a cable TV company or on DSL.
- From there the signal travels over standard available Internet connections using TCP/IP through multiple servers and routers until it comes to the desired location.
- At the receiving phone, the process is reversed.
- The Internet signal gets converted back to Ethernet, and then the VoIP processor recovers the original packet.
- From there, the compressed data is extracted, decompressed by a DSP, and sent to the DAC in the codec where the original voice is heard.

Problems with VoIP

- One of the main problems with VoIP is that it takes a relatively long time to transmit the voice data over the Internet.
- The packets may take different routes through the Internet.
- They all do eventually arrive at their intended destination, but often the packets are out of sequence.
- The receiving phone must put them back together in the correct sequence. This takes time.
- Even though the signals traverse the high-speed optical Internet lines at gigabit speeds, the packets pass through numerous routers and servers, each adding transit time or latency.
- This annoying latency is unacceptable to most. Keeping the latency below 150 ms minimizes this problem

Link Establishment:

- In the PSTN, the dialing process initiates multiple levels of switching that literally connects the calling phone to the called phone.
- That link is maintained for the duration of the call because the switches stay in place and the electronic paths stay dedicated to the call.
- In Internet telephony, no such temporary dedicated link is established because of the packetized nature of the system. Yet some method must be used to get the voice data to the desired phone.
- This is taken care of by a special protocol developed for this purpose.
- The initial protocol used was the ITU H.323. Today, however, a newer protocol established by the Internet Engineering Task Force (IETF) called the session initiation protocol (SIP) has been adopted as the standard.
- In both cases, the protocol sets up the call and then makes sure that the voice packets produced by the calling phone get sent to the receiving phone in a timely manner

Best wishes