Analog vs. Digital Transmission

Compare at two levels:

1. Data—continuous (audio) vs. discrete (text)
2. Signaling—continuously varying electromagnetic wave vs. sequence of voltage pulses.

Also Transmission—transmit without regard to signal content vs. being concerned with signal content. Difference in how attenuation is handled, but not focus on this.

Look at Table 2.3 and Fig 2.13 in Stallings.

Seeing a shift towards digital transmission despite large analog base. Why?

- improving digital technology
- data integrity. Repeaters take out cumulative problems in transmission. Can thus transmit longer distances.
- easier to multiplex large channel capacities with digital
- easy to apply encryption to digital data
- better integration if all signals are in one form. Can integrate voice, video and digital data.

Analog Transmission

Phone System

Because of the role played by phone companies in data transmission, analog transmission has dominated communications for quite some time. Today, everyone agrees that fiber is the way to go. As much as we prefer fiber, analog communications will be with us for a long time. Consider the phone system. It is characterized by:

1. Low bandwidth: It carries a bandwidth of about 3 kHz. That is, the system only allows signals between 0-3kHz to pass through — all higher frequencies are chopped off. The 0-3kHz spectrum covers the most important frequencies of human voice, which is precisely what the phone system has been designed to carry.
2. High error rate: Relative to LANs, the error rate is roughly 11 orders of magnitude higher! Errors don't matter as much to analog communication, especially when voice is involved. For digital communications, of course, a 1-bit error can have devastating consequences.

The phone system is organized into a hierarchy:

- Local phones are connected to a *central office* over a 2-wire circuit. That is, only two wires are needed to connection your phone to the phone system.
- The 2-wire path is called the *local loop*, and is no longer than 10 km in length.
- An area code and the first three digits of a telephone number uniquely identify or *address* a central office.
- Calls between neighbors connected to the same central office go only through the central office.

  Each call ties up a circuit in the central office. Typically only a small fraction of the circuits can be used simultaneously. That is, the phone company plans for expected use rather than worst case use.
- Central offices connect to *toll offices* over *toll connecting trunks*, which typically have higher bandwidths then the local loop. In fact, higher bandwidth trunks carry multiple "voice grade" lines over a single physical channel.
- Toll offices are then connected in various ways.
- Fiber is increasingly connecting toll offices and central offices, but not the local loop. Why? It's not cost effective to replace wiring at the local loop level. For trunks, however, many phone customers essentially share the lines, making the costs worthwhile.

Digital Data/Analog Signals

Must convert digital data to analog signal. One such device is a *modem* to translate between bit-serial and modulated carrier signals.

To send digital data using analog technology, the sender generates a *carrier signal* at some continuous tone (e.g. 1-2 kHz in phone circuits) that looks like a sine wave. The following techniques are used to encode digital data into analog signals (Fig 2-18)

Resulting bandwidth is centered on the carrier frequency.

- *amplitude-shift modulation (keying)*: vary the amplitude (e.g. voltage) of the signal. Used to transmit digital data over optical fiber.
- *frequency-shift modulation*: two (or more tones) are used, which are near the carrier frequency. Used in a full-duplex modem (signals in both directions).
- *phase-shift modulation*: systematically shift the carrier wave at uniformly spaced intervals.

  For instance, the wave could be shifted by 45, 135, 225, 315 degree at each timing mark. In this case, each timing interval carries 2 bits of information.

  Why not shift by 0, 90, 180, 270? Shifting zero degrees means no shift, and an extended set of no shifts leads to clock synchronization difficulties.

Another variation, called *Quadrature Amplitude Modulation*, has the following characteristics:

Look at Figure 2-25. Can use:

- QAM. 4 phase shifts—2-bit encoding. (*Quadrature Phase-Shift Keying*)
- QAM-16. 4 phase shifts plus four amplitudes—4-bit encoding.

  Used with 2400 baud we get an effective data rate of 9600bps.
- QAM-64. 6-bit encoding.
- V.32 for 9600 bps and combine with error correction (Fig 2-26).
- V.34 12-bit encoding

Analog Data/Analog Signals

Can actually transmit analog data in a similar manner with amplitude-, phase- and frequency-modulated waves. Stallings Fig 4.20.

Two reasons:

- Transmission media may need to use a higher frequency than that used by the data (such as voice)
- Modulation permits frequency-division multiplexing.

Asymetric Digital Subscriber Lines

“broadband”—services with more bandwidth than standard telephone service. 56kbps modems just not enough.

Implemented by phone companies by connecting customers to a different kind of switch that does not attenuate frequencies as is done for standard voice lines.

Achievable bandwidth becomes a factor of the distance from the customer to the switch.

Fig. 2-28 shows allocation of bandwidth into 4KHz channels with more allocated for downstream (end office to home) than upstream (home to end office). Get higher data rates than standard phone service—up to 8Mbps downstream and 1Mbps upstream.

Fig 2-29 shows equipment configuration. Can also not have splitter and have a filter on each telephone line.

ADSL is just a physical layer standard allowing higher speed access to telephone customers.

Being upgraded to fiber optics into the home.

Digital Transmission

Digital transmission has several advantages over analog transmission:

1. Analog circuits require amplifiers, and each amplifier adds distortion and noise to the signal.
2. In contrast, digital amplifiers regenerate an exact signal, eliminating cumulative errors.An incoming (analog) signal is sampled, its value is determined, and the node then generates a new signal from the bit value; the incoming signal is discarded. With analog circuits, intermediate nodes amplify the incomiing signal, noise and all.
3. Voice, data, video, etc. can all by carried by digital circuits. What about carrying digital signals over analog circuit? The modem example shows the difficulties in carrying digital over analog.

Digital Data/Digital Signals

A simple encoding method is to use constant voltage levels for a "1" and a "0". Can lead to long periods where the voltage does not change.

Clock Synchronization

With digital transmission, one problem that continually arises is *clock synchronization*. The receiver must be able to determine when one bit time ends and the next one starts, so that it samples one pulse, rather than part of one pulse and part of the next.

Note: quartz clocks are not accurate enough. Eventually, the sender and receiver's clock will drift apart.

Possibilities:

1. include timing information in the data signal
2. use a separate channel (e.g., wire) to transmit timing information

Manchester encoding is one technique that provides clocking information. The encoding splits each sampling unit into 2 halves where:

- a binary "1" is sent as a high-low voltage sequence
- a "0" is sent as a low-high sequence
- because each sampling time contains one transition, the receiver can easily synchronize its clock to the sender's.

In a related technique, *differential Manchester encoding*, a "1" bit is indicated by the absence of a transition at the start of the bit time, while a "0" is indicated by the presence of a transition.

Drawback of Manchester encoding:

- half the bandwidth is wasted because it takes two transitions to represent one bit

Advantages:

- reduced complexity of transmitter and receiver components

See Fig 4-16.

Analog Data/Digital Signals

Although most local loops are analog, end offices increasingly use digital circuits for inter-trunk lines. A *codec* (coder/decoder) is a device that converts an analog signal into a digital signal. To convert analog signals to digital signals, many systems use *Pulse Code Modulation* (PCM):

1. PCM samples the 4kHz signal 8,000 times per second. Why? PCM takes advantage of Nyquist's result, sampling the 4kHz bandwidth signal at $2H = 8$ thousand times per second. (Here we assume the use of a standard voice grade line.)
2. Each sample measures the amplitude of the signal, converting it into an n-digit integer value.
3. The digital channel carries these n-digit encodings.

T1 Carrier

One popular product is Bell's T1 carrier (Figure 2-33)

1. It multiplexes 24 voice channels over one digital channel. That is, it carries 24 voice channels at the same time over one digital channel.
2. Each of the 24 analog inputs is sampled in round-robin fashion and its n-bit encoding is sent down the wire.
3. Each encoding consists of 7 bits of sampled data, plus 1 bit of signaling information (e.g., out-of-band information).
4. Each voice grade sub-channel carries (7 bits X 8000 samples) = 56kbps of data, plus 8000 bps of signaling information, requiring a digital data rate of 64kbps.
5. Samples are transmitted in 193-bit units (frames).
6. Each 193-bit frame consists of $24 \times 8 + 1$ bits of information; the extra bit of information carries synchronization information It alternates between a "0" and "1" allowing the receiver to verify that it is properly recognizing the start and end of frames.
7. A T1 channel has an aggregate carrying capacity of 1.544 Mbps.

International Standard

As for the international standard, CCITT felt that 8 kbps signaling was overkill, so their standard encodes digital signals differently:

1. In *Common Channel Signaling*, all 8 bits carry data, and the extra frame bit is used to carry framing and signaling information
2. *Channel Associated Signaling* is yet another variation on the same idea. Here, five of six samples carries 8 bits of data, while every sixth sample carries seven bit of data and one for signalling.

Note: Other Bell standards specify how T1 trunks are to be multiplexed over higher capacity trunks, such as: T2 (6.3 Mbps), T3 (44.7 Mbps) and T4 (274.2 Mbps).

Encoding Systems

It turns out that 8 bits of data can be reduced through *compression*. For compression, the assumption is that the signal changes relatively slowly compared to the sampling frequency:

1. In *differential pulse code modulation*, each sample contains the (signed) difference between the current and previous amplitude value.

   It only requires 5 bits, and works well in practice with voice traffic.
2. *Delta modulation* assumes that each sample differs by either +/- 1 relative to the previous sample, requiring only a single bit to represent each sample. See Figure 2-34.
3. *Predictive encoding* attempts to predict what the next sample will look like, transmitting the difference between the actual measured sample and the expected sample.

Transmission Multiplexing & Switching

Problem: Given a channel of large capacity, how does one subdivide the channel into smaller logical channels for individual users? *Multiplex* many conversations over same channel. Three flavors of solution:

Frequency division multiplexing (FDM): Divide the frequency spectrum into smaller subchannels, giving each user exclusive use of a subchannel (e.g., radio and TV). One problem with FDM is that a user is given all of the frequency to use, and if the user has no data to send, bandwidth is wasted — it cannot be used by another user.

Time division multiplexing (TDM): Use time slicing to give each user the full bandwidth, but for only a fraction of a second at a time (analogous to time sharing in operating systems). Again, if the user doesn't have data to sent during his timeslice, the bandwidth is not used (e.g., wasted).

Statistical multiplexing: Allocate bandwidth to arriving packets on demand. This leads to the most efficient use of channel bandwidth because it only carries useful data. That is, channel bandwidth is allocated to packets that are waiting for transmission, and a user generating no packets doesn't use any of the channel resources.

Now we see another reason why the phone system limits the bandwidth voice grade lines to 3kHz. FDM is used on the trunk lines, allocating 4 kHz to each channel.

Only 3 kHz is consistently usable, with 500 Hz of *guard bandwidth* on each end of the spectrum.

One common organization of channels is as follows:

- Bundle 12 voice grade lines into a unit called a *group.* A group carries signals in the 60-108 kHz spectrum.
- Combine 5 groups into *supergroup.*
- Combine 5 supergroups into a *mastergroup.*

Both TDM and FDM work well with *continuous* transmission, in which data is generated at a constant rate (e.g. voice). How well does it work for computer traffic? Not so well. Computer traffic is extremely *bursty*, characterized by alternating periods of idleness and heavy data transmission.

Circuit Switching

The phone system uses a technique called *circuit switching* (see Figure 2-39).

1. Once a call has been completed, the user sees a set of "virtual wires" between communicating endpoints.
2. The user sends a continuous stream of data, which the channel guarantees to deliver at a known rate.
3. Data transmission handled elegantly using TDM or FDM. Note that TDM/FDM work well because the data rate is predictable — the voice grade signal is sampled using PCM generating a steady stream of bits.
4. *Call setup* required before any data can be sent, allowing network to set up the path, allocate subchannels, etc. Call setup also used to decide who to charge for the call.
5. *Call termination* required when parties complete call, allowing the network to reclaim resources. At this point, a billing record is saved somewhere that records where the call was made, its duration, etc.

Advantages of circuit switching:

1. Fixed bandwidth, guaranteed capacity (e.g., no congestion).
2. Low-varience end-to-end delay (e.g., delay nearly constant).

Drawbacks:

1. Connection setup introduces delay before communication can begin.
2. User pays for circuit, even when not sending any data.
3. Other users cannot use bandwidth of other circuits that are not actually being used(e.g., in most conversations, only one person speaks at a time. Thus, half the underlying bandwidth is wasted!

Message Switching

Entire message stored at each node. Each message is received in its entirety before forwarding. A *store-and-forward* network (UUCP is a store and forward network).

Packet Switching

In contrast, *packet switching* systems use statistical multiplexing to make better use of a channel:

1. Data is sent in individual messages (packets).
2. Each message is forwarded from switch to switch, eventually reaching its destination.
3. Each switch has a small amount of buffer space to temporarily hold messages. If an outgoing line is busy, the packet is queued until the line becomes available.

Packet switching vs circuit switching:

1. (Current) packet switching system do not provide known delay or capacity characteristics. Some applications, like those making use of real-time voice and video, cannot tolerate high variation in delays.
2. If many sites send data at the same time, end-to-end delay increases. That is, per-user response and throughput drops as more users share a channel.
3. Packet switching utilizes resources more efficiently (similar to multiprocessing in operating systems). In particular, with circuit switching, bandwidth can be allocated but unused, as when no one talks.
4. Packet switching systems doesn't usually require opening a connection before sending data. This important for applications that send only a single packet of data; the cost of opening and closing a connection may exceed the cost of sending the data.
5. Billing algorithm more complex in packet switching systems. It's easy to bill for a connection, because one can figure out who to charge during the connection set up. With packet-switching, each packet must be accounted for individually.

Hybrid Switching

Hybrid switching systems attempt to combine the advantages of both approaches.

For instance, phone companies have developed *fast connect circuit switching* systems that establish connections quickly (e.g. on each interactive input line). However, there is still much debate as to whether these "fast" systems are really fast enough.

Another variation, *virtual circuits*, requires users to open a connection before sending data, but transmits packets. The call allows the network to establish a path, and once established, all packets follow the same path. Because all packets follow the same path, packets can be delivered in order, and accounting is simplified.

Mobile Telephone System

Section 2.6 in Tanenbaum. Lots of details.

Internet over Cable

Use fiber connections between head ends and fiber end points. Use a small frequency range for upstream data and a larger amount for downstream data (Fig 2-48).

Used for cable modems. Fig 2-49.

ADSL vs. Cable

Both use fiber backbone.

On end-points cable uses coax, ADSL uses twisted pair. Coax has higher bandwidth, but much of the bandwidth is being used for television transmission.

ADSL bandwidth is more consistent as cable bandwidth depends on the number of users. Much more independence for ADSL users than cable users.

Availability—not all users are close enough to end offices.