Carriers and Modulation

CS442

DIGITAL TRANSMISSION OF DIGITAL DATA

Review…

Baseband Transmission

Digital transmission is the transmission of electrical pulses. Digital information is binary in nature in that it has only two possible states 1 or 0. Sequences of bits encode data (e.g., text characters).

Digital signals are commonly referred to as <u>baseband signals</u>.

In order to successfully send and receive a message, both the sender and receiver have to agree how often the sender can transmit data (<u>data rate</u>).

<u>Data rate</u> often called <u>bandwidth</u> – but there is a different definition of bandwidth referring to the frequency range of a signal!
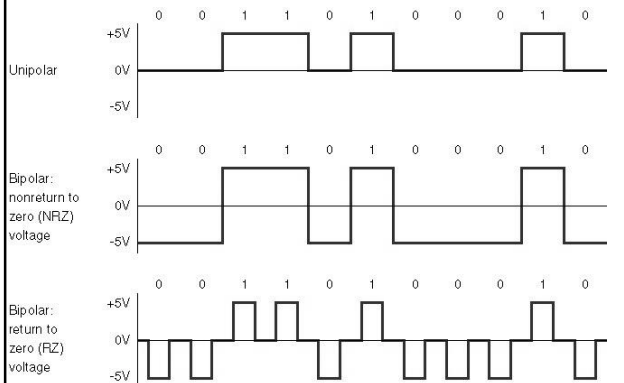
Baseband Transmission

With unipolar signaling techniques, the voltage is always positive or negative (like a dc current).

In bipolar signaling, the 1's and 0's vary from a plus voltage to a minus voltage (like an ac current).

In general, bipolar signaling experiences fewer errors than unipolar signaling because the signals are more distinct.

Baseband Transmission

Baseband Transmission

Manchester encoding is a special type of unipolar signaling in which the signal is changed from a high to low (0) or low to high (1) in the middle of the signal.

- More reliable detection of transition rather than level
  - consider perhaps some constant amount of dc noise, transitions still detectable but dc component could throw off NRZ-L scheme
  - Transitions still detectable even if polarity reversed

Manchester encoding is commonly used in local area networks (ethernet, token ring).
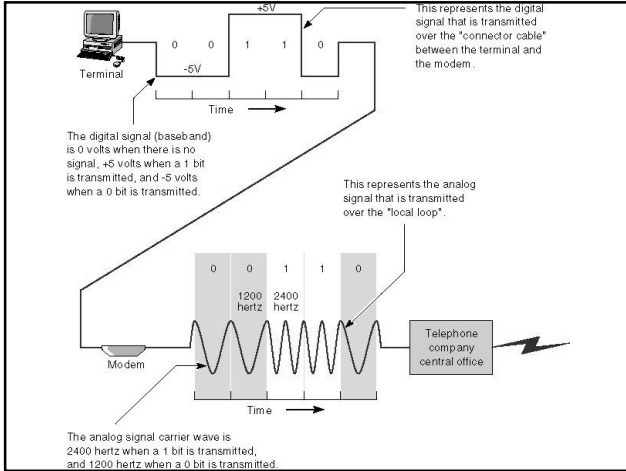
Manchester Encoding
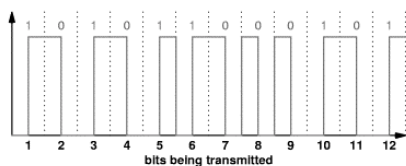
ANALOG TRANSMISSION OF DIGITAL DATA

Analog Transmission occurs when the signal sent over the transmission media continuously varies from one state to another in a wave-like pattern.
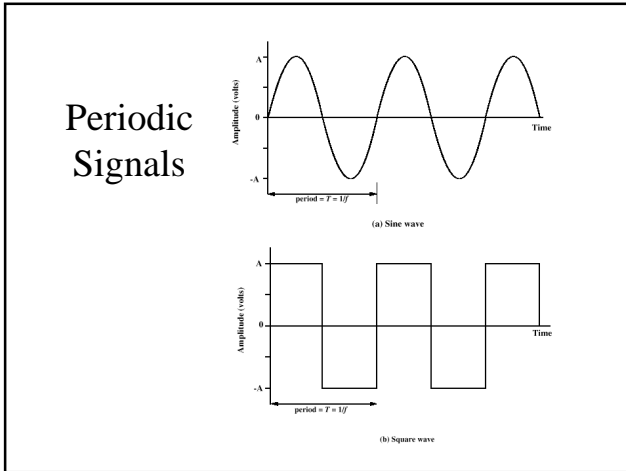
e.g. telephone networks, originally built for human speech rather than data.

Advantage for long distance communications: much less attenuation for analog carrier than digital

2

Slide 1 (Digital Data to Analog diagram)

+5V

0 0 1 1 0

-5V

Terminal

Time

This represents the digital signal that is transmitted over the "connector cable" between the terminal and the modem.

The digital signal (baseband) is 0 volts when there is no signal, +5 volts when a 1 bit is transmitted, and -5 volts when a 0 bit is transmitted.

This represents the analog signal that is transmitted over the "local loop".

0 0 1 1 0

1200 hertz 2400 hertz

Telephone company central office

Modem

Time

The analog signal carrier wave is 2400 hertz when a 1 bit is transmitted, and 1200 hertz when a 0 bit is transmitted.

Slide 2

Digital Data to Analog Transmission

Before we get further into Analog to Digital, we need to understand various characteristics of analog transmission.

Slide 3 (Periodic Signals)

Periodic Signals

A

Amplitude (volts)

0

Time

-A

period = T = 1/f

(a) Sine wave

A

Amplitude (volts)

0

Time

-A

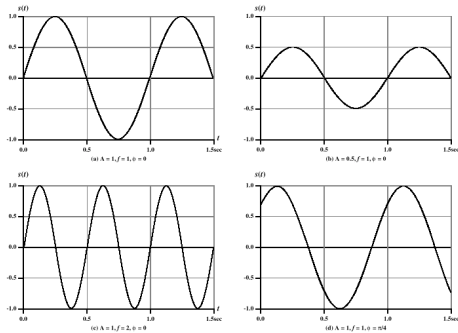period = T = 1/f

(b) Square wave

Slide 4

Sine Wave

- Peak Amplitude (A)
  - maximum strength of signal
  - volts
- Frequency (f)
  - Rate of change of signal
  - Hertz (Hz) or cycles per second
  - Period = time for one repetition (T)
  - $T = 1/f$
- Phase (ϕ)
  - Relative position in time, from 0-2*pi
- General Sine wave

$$s(t) = A\sin(2\pi f t + \phi)$$

Varying Sine Waves

(a) A = 1, f = 1, φ = 0
(b) A = 0.5, f = 1, φ = 0
(c) A = 1, f = 2, φ = 0
(d) A = 1, f = 1, φ = π/4

## Wavelength

- Distance occupied by one cycle
- Distance between two points of corresponding phase in two consecutive cycles
- λ = Wavelength
- Assuming signal velocity v
  - $\lambda = vT$
  - $\lambda f = v$
  - $c = 3*10^8$ ms^{-1} (speed of light in free space)

Frequency Domain Concepts

- Signal usually made up of many frequencies
- Components are sine (or cosine) waves
- Can be shown (Fourier analysis) that any continuous signal is made up of component sine waves
- Can plot frequency domain functions

Addition of Frequency Components
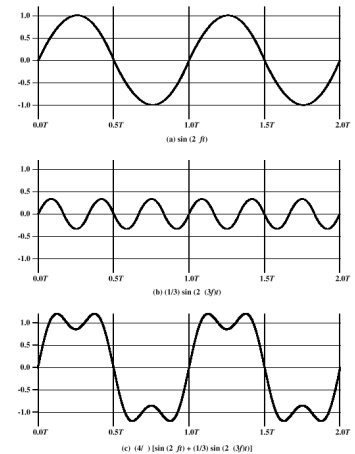
Notes:

2nd freq a multiple of 1st
1st called fundamental freq
Others called harmonics

Period of combined = Period of the fundamental

Fundamental = carrier freq

(a) sin (2 ft)

(b) (1/3) sin (2 (3f)t)

(c) (4/) [sin (2 ft) + (1/3) sin (2 (3f)t)]
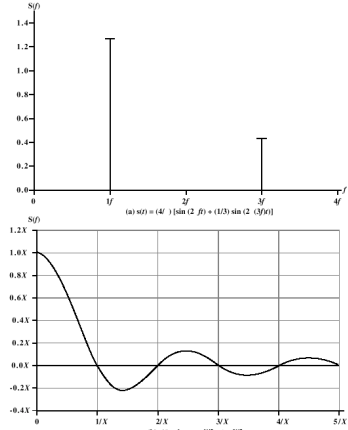
4

Frequency Domain

Discrete Freq Rep:

$$s(t) = 4/\pi[\sin(2\pi ft) + 1/3\sin(2\pi(3f)t)]$$

Any continuous signal can be represented as the sum of sine waves! (May need an infinite number..)

Discrete signals result in Continuous, Infinite Frequency Rep:

s(t)=1 from –X/2 to X/2

(a) s(t) = (4/π) [sin (2 ft) + (1/3) sin (2 (3f)t)]

(b) s(t) = 1   – X/2 ≤ t ≤ X/2

Data Rate and Bandwidth

- Any transmission system has a limited band of frequencies
- This limits the data rate that can be carried

- Spectrum
  - range of frequencies contained in signal
- Absolute bandwidth
  - width of spectrum
- Effective bandwidth
  - Often just *bandwidth*
  - Narrow band of frequencies containing most of the energy
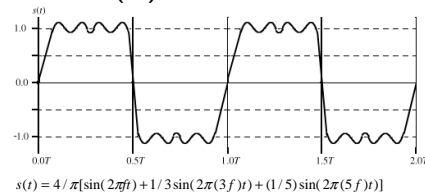
Example of Data Rate/Bandwidth

Want to transmit:
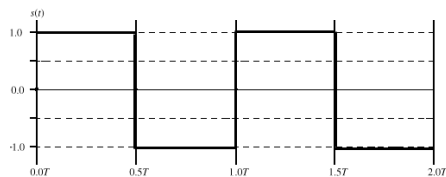
Let's say that f=1Mhz or 10^6 cycles/second, so T= 1microsecond

Let's approximate the square wave with a few sine waves:

Ex(1): Sine Wave 1
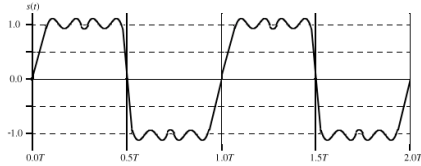

$$s(t) = 4/\pi[\sin(2\pi ft) + 1/3\sin(2\pi(3f)t) + (1/5)\sin(2\pi(5f)t)]$$

Bandwidth=5f-f =4f
If f=1Mhz, then the bandwidth = 4Mhz

T=1 microsecond; we can send two bits per microsecond so the data rate = 2 * 10^6 = 2Mbps

Ex(2): Sine Wave 1, Higher freq

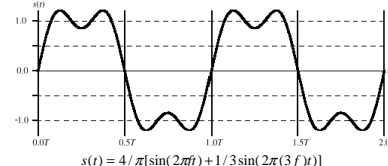$$s(t) = 4/\pi[\sin(2\pi f t) + 1/3\sin(2\pi(3f)t) + (1/5)\sin(2\pi(5f)t)]$$

Bandwidth=5f-f =4f
If f=2Mhz, then the bandwidth = 8Mhz

T=0.5 microsecond; we can send two bits per 0.5 microseconds
or 4 bits per microsecond, so the data rate = 4 * 10^6 = 4Mbps

Double the bandwidth, double the data rate!

Ex(3): Sine Wave 2

$$s(t) = 4/\pi[\sin(2\pi f t) + 1/3\sin(2\pi(3f)t)]$$

Bandwidth=3f-f =2f
If f=2Mhz, then the bandwidth = 4Mhz

T=0.5 microsecond; we can send two bits per 0.5 microseconds
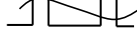or 4 bits per microsecond, so the data rate = 4 * 10^6 = 4Mbps

Still possible to get 4Mbps with the "lower" bandwidth, but our
receiver must be able to discriminate from more distortion!

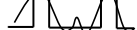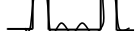## Bandwidth / Representation

2000 bps

B=500 Hz

B=1000 Hz

B=1700 Hz

B=4000 Hz

Increasing bandwidth improves the representation of the data signal.

500Hz too low to reproduce the signal. Want to maximize the capacity of the available bandwidth.

Multiplexers

- A <u>multiplexer</u> puts two or more simultaneous transmissions on a single communications circuit.
- Generally speaking, the multiplexed circuit must have the same capacity as the sum of the circuits it combines.
- The primary benefit of multiplexing is to save money.
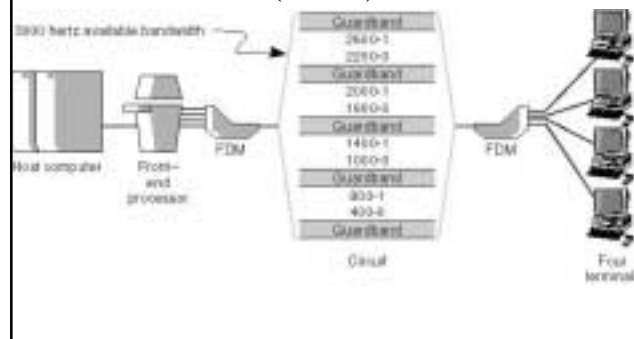
Multiplexed Circuit
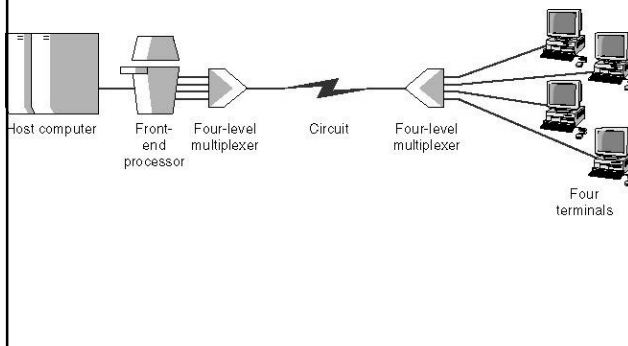
Multiplexing

There are three major types of multiplexers
- Frequency division multiplexers (FDM)
  - E.g. AM/FM Radio, Telephone
- Time division multiplexers (TDM)
  - ISDN
- Statistical time division multiplexers (STDM)
  - We'll cover later (maybe)
- Wavelength division multiplexing (WDM)
  - Used in optical carriers (colors carry signals)

Frequency Division Multiplexing (FDM)

- Frequency division multiplexers can be described as dividing the circuit "horizontally" so that many signals can travel a single communication circuit simultaneously.
- The circuit is divided into a series of separate channels, each transmitting on a different frequency.
- Guardbands are employed to keep one channel from leaking over into another channel.
- Frequency division multiplexers are somewhat inflexible because once you determine how many channels are required, it may be difficult to add more channels without purchasing an entirely new multiplexer.

Frequency Division Multiplexing (FDM)

Time Division Multiplexing (TDM)

- Time division multiplexing shares a circuit among two or more terminals by having them take turns, dividing the circuit "vertically."
- Time on the circuit is allocated even when data are not transmitted, so that some capacity is wasted when a terminal is idle.
- Time division multiplexing is generally more efficient and less expensive to maintain than frequency division multiplexing, because it does not need guardbands.

Time Division Multiplexing (TDM)
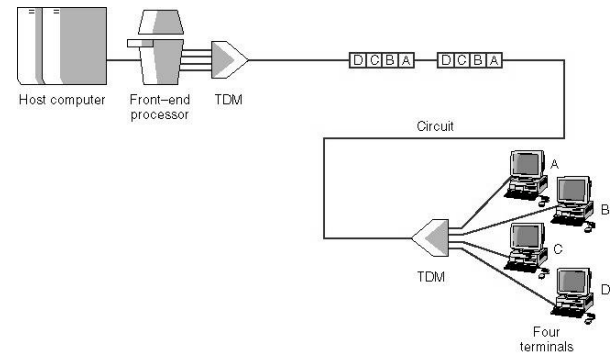
Transmission Impairments

- Signal received may differ from signal transmitted
- Analog - degradation of signal quality
- Digital - bit errors
- Caused by
  - Attenuation and attenuation distortion
  - Delay distortion
  - Noise

Attenuation

- Signal strength falls off with distance
- Depends on medium
- Received signal strength:
  - must be enough to be detected
  - must be sufficiently higher than noise to be received without error
- Attenuation is an increasing function of frequency; higher frequencies suffer from more attenuation. Can distort the signal.

- Solution: Equalization. Boost higher frequency components.

Delay Distortion

- Only in guided media
- Propagation velocity varies with frequency
  - Velocity highest near center frequency
  - Results in phase shift at different frequencies
  - "Overlapping" bits

- Solution: Equalization

Noise (1)

- Additional signals inserted between transmitter and receiver
- Thermal
  - Due to thermal agitation of electrons
  - Uniformly distributed
  - White noise
- Intermodulation
  - Signals that are the sum and difference of original frequencies sharing a medium

Noise (2)

- Crosstalk
  - A signal from one line is picked up by another
- Impulse
  - Irregular pulses or spikes
  - e.g. External electromagnetic interference
  - Short duration
  - High amplitude

What Causes Errors?
Summary of Errors and Noise:

| Source of Error	What Causes It	How to Prevent It.
Line Outages	Storms, Accidents	
White Noise	Movement of electrons	Increase signal strength
Impulse Noise	Sudden increases in electricity (e.g. lightning)	Shield or move the wires
Cross-Talk	Multiplexer guardbands too small, or wires too close together	Increase the guardbands, or move or shield the wires
Echo	Poor connections	Fix the connections, or tune equipment
Attenuation	Graduate decrease in signal over distance	Use repeaters or amps
Intermodulation Noise	Signals from several circuits combine	Move or shield the wires
Jitter	Analog signals change phase	Tune equipment
Harmonic Distortion	Amplifier changes phase	Tune equipment

Error Prevention

There are many ways to prevent errors:
- Shielding (adding insulation)
- Moving cables away from noise sources
- Changing multiplexing type (FDM→TDM)
- Tuning transmission equipment and improving connection quality
- Using amplifiers and repeaters
- Equalization
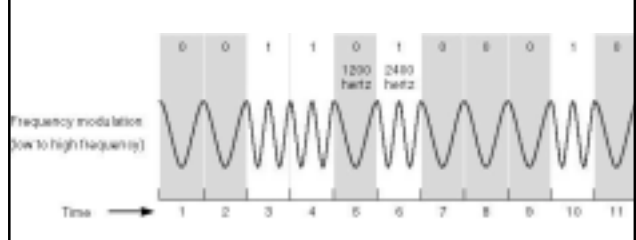- Leasing conditioned circuits

Modulation - Digital Data, Analog Signal

- Public telephone system
  - 300Hz to 3400Hz
    - Guardband from 0-300, 3400-4000Hz
  - Use modem (modulator-demodulator)
- Amplitude shift keying (ASK)
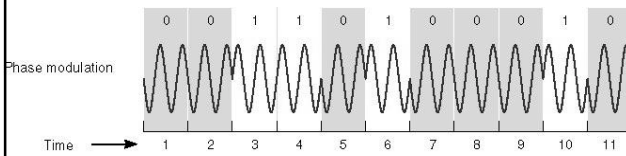- Frequency shift keying (FSK)
- Phase shift keying (PSK)

Amplitude Modulation and ASK
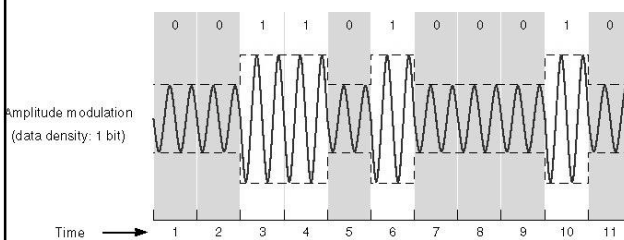

Frequency Modulation and FSK

Phase Modulation and PSK

Amplitude Shift Keying

- Values represented by different amplitudes of carrier
- Usually, one amplitude is zero
  - i.e. presence and absence of carrier is used
- Susceptible to sudden gain changes
- Inefficient
- Typically used up to 1200bps on voice grade lines
- Used over optical fiber

Frequency Shift Keying

- Values represented by different frequencies (near carrier)
- Less susceptible to error than ASK
- Typically used up to 1200bps on voice grade lines
- High frequency radio
- Even higher frequency on LANs using co-ax

FSK on Voice Grade Line
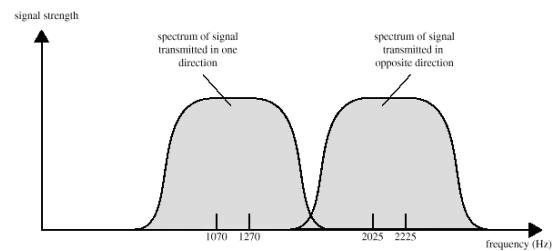
Bell Systems 108 modem

Figure 5.8 Full-Duplex FSK Transmission on a Voice-Grade Line

Phase Shift Keying

- Phase of carrier signal is shifted to represent data
- Differential PSK
  - Phase shifted relative to previous transmission rather than some reference signal

Sending Multiple Bits Simultaneously

Each of the three modulation techniques can be refined to send more than one bit at a time. It is possible to send two bits on one wave by defining four different amplitudes.

This technique could be further refined to send three bits at the same time by defining 8 different amplitude levels or four bits by defining 16, etc. The same approach can be used for frequency and phase modulation.

Sending Multiple Bits Simultaneously
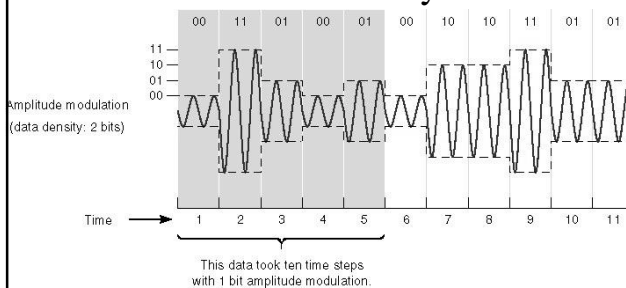
Sending Multiple Bits Simultaneously

In practice, the maximum number of bits that can be sent with any one of these techniques is about five bits. The solution is to combine modulation techniques.

One popular technique is quadrature amplitude modulation (QAM) involves splitting the signal into eight different phases, and two different amplitude for a total of 16 different possible values, giving us lg(16) or 4 bits per value.

2-D Diagram of QAM
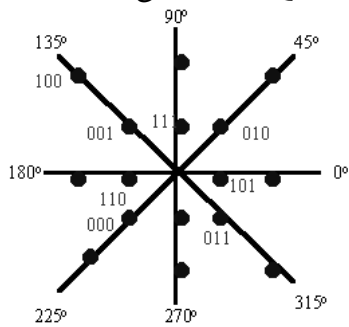
Sending Multiple Bits Simultaneously

Trellis coded modulation (TCM) is an enhancement of QAM that combines phase modulation and amplitude modulation.

The problem with high speed modulation techniques such as TCM is that they are more sensitive to imperfections in the communications circuit.

Bits Rate Versus Baud Rate Versus Symbol Rate

The terms bit rate (the number of bits per second) and baud rate are used incorrectly much of the time. They are not the same.

A bit is a unit of information, a baud is a unit of signaling speed, the number of times a signal on a communications circuit changes. ITU-T now recommends the term baud rate be replaced by the term symbol rate.

Bits Rate Versus Baud Rate Versus Symbol Rate

The bit rate and the symbol rate (or baud rate) are the same only when one bit is sent on each symbol. If we use QAM or TCM, the bit rate would be several times the baud rate.

Modem Standards

There are many different types of modems available today.

Most modems support several standards so that they can communicate with a variety of different modems.

Better modems can change data rates during transmission to reduce the rate in case of noisy transmission (<u>fast retrain</u>).

Modem Standards

Modem Standard	Maximum Symbol Rate	Maximum Bits per Symbol	Maximum Data Rate (bps)
V.22	1200	1	1200
	2400	1	2400
V.32	2400	4	9600
V.32bis	2400	6	14,400
V.34	3429	8.4	28,800
V.34+	3429	9.8	33,600

Modem Standards

- V.22
  - 1200-2400 baud/bps, FSK
- V.32 and V.32bis
  - full duplex at 9600 bps (2400 baud at QAM)
  - bis uses TCM to achieve 14,400 bps.
- V.34 and V.34bis
  - Works best for phone networks using digital transmission beyond the local loop to reduce noise. Up to 28,800 bps (TCM)
  - bis up to 36,600 with TCM

Modem Standards

- V.42bis

  data compression modems, accomplished by run length encoding, code book compression, Huffman encoding and adaptive Huffman encoding
  - MNP5 - uses Huffman encoding to attain 1.3:1 to 2:1 compression.
  - bis uses Lempel-Ziv encoding and attains 3.5:1 to 4:1.
  - V.42bis compression can be added to almost any modem standard effectively tripling the data rate.

Analog Data, Digital Signal

- Digitization
  - Conversion of analog data into digital data
  - Digital data can then be transmitted using digital signaling (e.g. Manchester)
  - Or, digital data can then be converted to analog signal
  - Analog to digital conversion done using a codec (coder/decoder)
  - Two techniques to convert analog to digital
    - Pulse code modulation / Pulse amplitude modulation
   - Delta modulation

Pulse Amplitude Modulation

Analog voice data must be translated into a series of binary digits before they can be transmitted.

With Pulse Amplitude Modulation, the amplitude of the sound wave is sampled at regular intervals and translated into a binary number.

The difference between the original analog signal and the translated digital signal is called quantizing error.
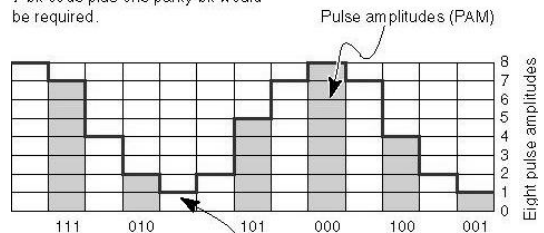
Pulse Amplitude Modulation

The signal (original wave) is quantized into 128 pulse amplitudes (PAM). In this example we have used only eight pulse amplitudes for simplicity. These eight amplitudes can be depicted by using only a 3-bit code instead of the 8-bit code normally used to encode each pulse amplitude.

Original wave

Eight pulse amplitudes
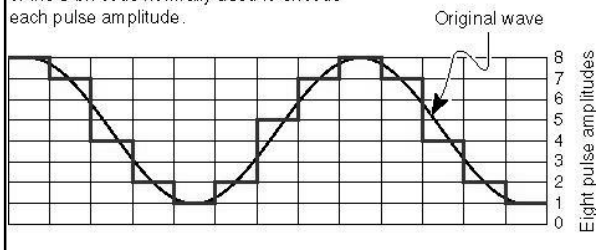
Pulse Amplitude Modulation

After quantizing, samples are taken at specific points to produce amplitude modulated pulses. These pulses are then coded. Because we used eight pulse levels, we only need three binary positions to code each pulse.[1] If we had used 128 pulse amplitudes, then a 7-bit code plus one parity bit would be required.

Pulse amplitudes (PAM)
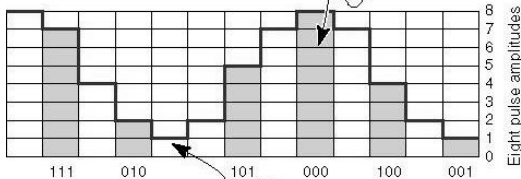
111   010   101   000   100   001

Eight pulse amplitudes

Pulse Amplitude Modulation

¹001 = PAM level 1	For digitizing a voice signal, 8,000 samples
010 = PAM level 2	per second are taken. These 8,000 samples
011 = PAM level 3	are then transmitted as a serial stream of
100 = PAM level 4	0s and 1s. In our case 8,000 samples times
101 = PAM level 5	3 bits per sample would require a 24,000
110 = PAM level 6	bps transmission rate. In reality, 8 bits per
111 = PAM level 7	sample times 8,000 samples requires a
000 = PAM level 8	64,000 bps transmission rate.

Pulse Amplitude Modulation

For standard voice grade circuits, the sampling of 3300 Hz at an average of 2 samples/second would result in a sample rate of 6600 times per second.

There are two ways to reduce quantizing error and improve the quality of the PAM signal.

– Increase the number of amplitude levels

– Sample more frequently (oversampling).

Pulse Code Modulation

Pulse Code Modulation is the most commonly used technique in the PAM family and uses a sampling rate of 8000 samples per second.

Each sample is an 8 bit sample resulting in a digital rate of 64,000 bps (8 x 8000).

Sampling Theorem: If a signal is sampled at a rate higher than twice the highest signal frequency, then the samples contain all the information of the original signal.

E.g.: For voice capped at 4Khz, can sample at 8000 times per second to regenerate the original signal.

Performance of A/D techniques

- Good voice reproduction via PCM
  - PCM - 128 levels (7 bit)
  - Voice bandwidth 4khz
  - Should be 8000 x 7 = 56kbps for PCM
    - (Actually 8000 x 8 with control bit)
- Data compression can improve on this
  - e.g. Interframe coding techniques for video
- Why digital?
  - Repeaters instead of amplifiers; don't amplify noise
  - Allows efficient and flexible Time Division Multiplexing over Frequency Division Multiplexing
  - Conversion to digital allows use of more efficient digital switching techniques

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Analog Data, Analog Signals

- Why modulate analog signals?
  - Higher frequency can give more efficient transmission
  - Permits frequency division multiplexing
- Types of modulation
  - Amplitude
  - Frequency
  - Phase
- Ex. Of analog/analog modulation : Spread Spectrum

Spread Spectrum

- Analog or digital data
- Analog signal
- Spread data over wide bandwidth
- Makes jamming and interception harder
- Frequency hopping
  - Signal broadcast over pseudorandom series of frequencies
- Direct Sequence
  - Each bit is represented by multiple bits in transmitted signal
  - Chipping code

Analog/Digital Modems (56k Modems)

The V.34+ modem is probably the fastest analog modem that will be developed.

The basic idea behind 56K modems (V.90) is to take the basic concepts of PCM both forwards and backwards.

The PSTN is already digitizing analog data, and sending it at 64Kbps. However, not all of these bit patterns are actually available for data (one bit used for control), so the maximum data rate becomes 56K.

The problem becomes the Analog to Digital conversion; assuming a 4Khz bandwidth coupled with quantization error limits us to the 33.6Kbps when performing an ADC conversion!

Solution: Eliminate one ADC conversions going downstream from the ISP; we still have to do a DAC conversion but this doesn't introduce quantization error (our V.90 modem must have the fine resolution to reproduce the original signal)

V.90 56K Modems
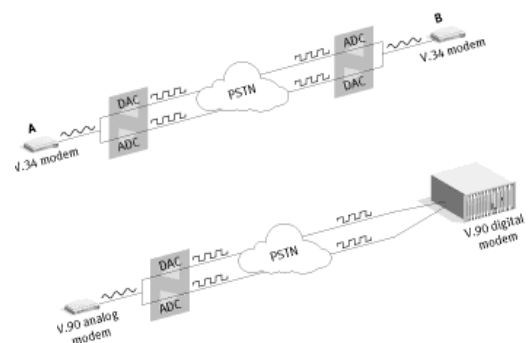
Analog/Digital Modems (56k Modems)

Noise is a critical issue. Tests found 56K modems to connect at less than 40 Kbps 18% of the time, 40-50 Kbps 80% of the time, and 50+ Kbps only 2 % of the time.

It is easier to control noise in the channel transmitting from the server to the client than in the opposite direction.

Because the current 56K technology is based on the PCM standard, it cannot be used on services that do not use this standard.

Cable Modems

- Much more complicated than normal modems
  - Tuner, decoder, modulator, demodulator, router, hub, etc.
  - Bus architecture – scalability issues
  - Downstream: 27Mbs per 6Mhz channel, QAM
  - Upstream: more noise, 3Mbps per 2Mhz channel, QPSK (2-3 bits per symbol)

xDSL

- Digital Subscriber Line – High bandwidth via ordinary copper lines
  - Typically range from 1.544 Mbps to 512 Kbps downstream and around 128 Kbps upstream but could have 8Mbps downstream, 768Kbps upstream for ADSL
  - Many variants: ADSL, ADSL-Lite, CDSL, HDSL, IDSL, SDSL, VDSL …
  - Need to be about <15000 feet from a CO
  - More scalable than cable modems
  - No real standards yet? (DSL-Lite somewhat, no splitter)

DSL Installation

Subscriber Network Element
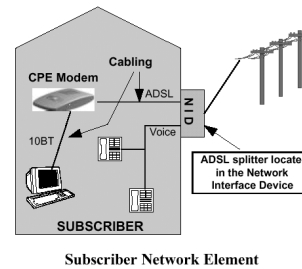
ADSL Splitter

Discrete Multitone (DMT) Linecode

POTS   4 KHz   1.1 MHz
32 QAM
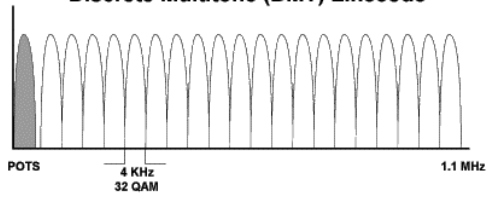
Data Rate = Number of channels x number of bit/channel x modulation rate

Theoretical maximum upstream data rate = 25 x 15 x 4 kHz = 1.5 Mb/s

Theoretical maximum downstream data rate = 249 x 15 x 4 kHz = 14.9 Mb/s