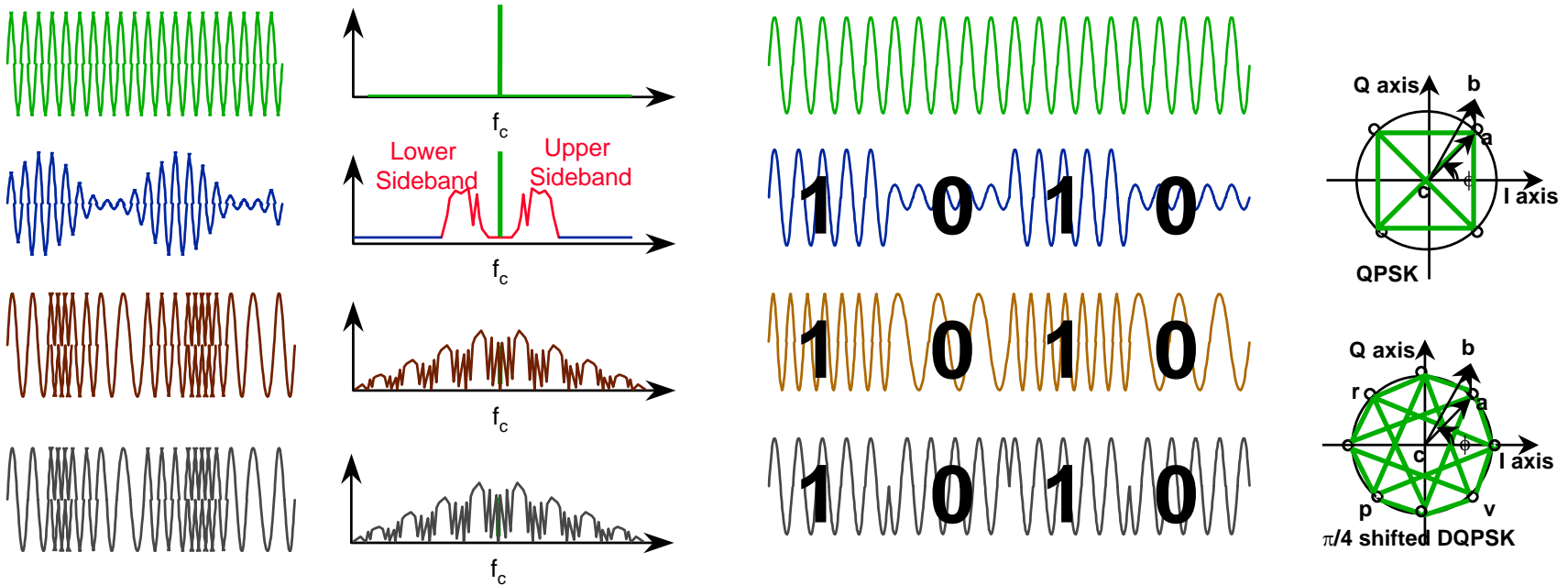
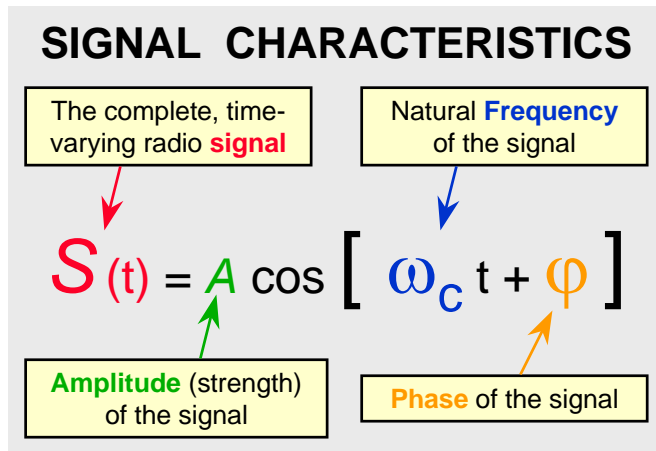


Chapter 2

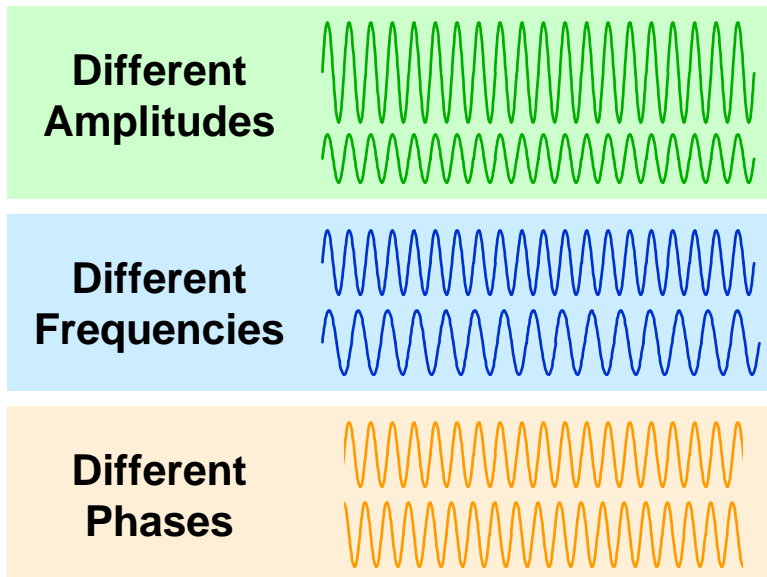
Wireless Systems: Modulation and Signal Bandwidth



Characteristics of a Radio Signal

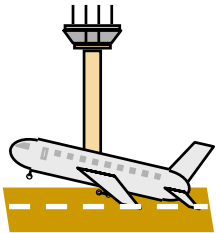
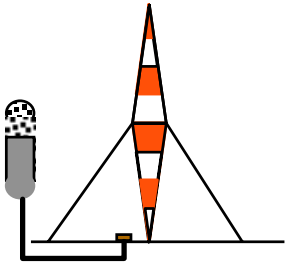
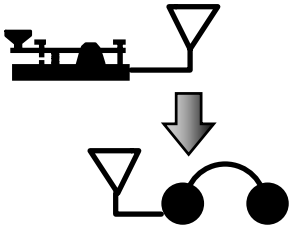


Compare these Signals:



- The purpose of telecommunications is to send information from one place to another
- Our civilization exploits the transmissible nature of radio signals, using them in a sense as our “carrier pigeons”
- To convey information, some characteristic of the radio signal must be altered (i.e., ‘modulated’) to represent the information
- The sender and receiver must have a consistent understanding of what the variations mean to each other
- RF signal characteristics which can be varied for information transmission:
 - Amplitude
 - Frequency
 - Phase

The Emergence of AM: A bit of History

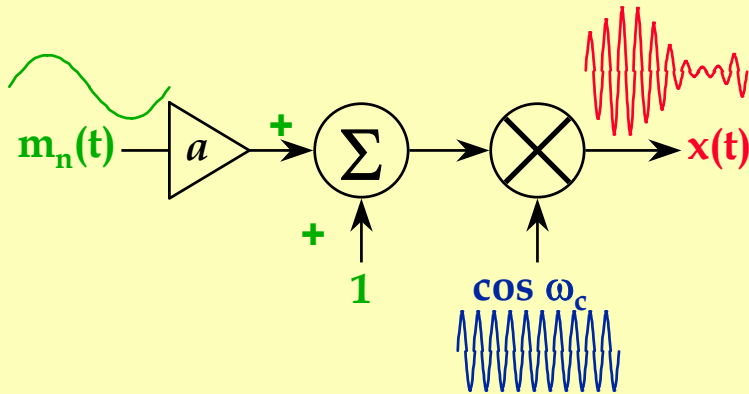


SSB
LSB **USB**

- The early radio pioneers first used binary transmission, turning their crude transmitters on and off to form the dots and dashes of Morse code. The first successful demonstrations of radio occurred during the mid-1890's by experimenters in Italy, England, Kentucky, and elsewhere.
- Amplitude modulation was the first method used to transmit voice over radio. The early experimenters couldn't foresee other methods (FM, etc.), or today's advanced digital devices and techniques.
- Commercial AM broadcasting to the public began in the early 1920's.
- Despite its disadvantages and antiquity, AM is still alive:
 - AM broadcasting continues today in 540-1600 KHz.
 - AM modulation remains the international civil aviation standard, used by all commercial aircraft (108-132 MHz. band).
 - AM modulation is used for the visual portion of commercial television signals (sound portion carried by FM modulation)
 - Citizens Band ("CB") radios use AM modulation
 - Special variations of AM featuring single or independent sidebands, with carrier suppressed or attenuated, are used for marine, commercial, military, and amateur communications

Amplitude Modulation (“AM”)

TIME-DOMAIN VIEW of AM MODULATOR



$$x(t) = [1 + am_n(t)]\cos \omega_c t$$

where:

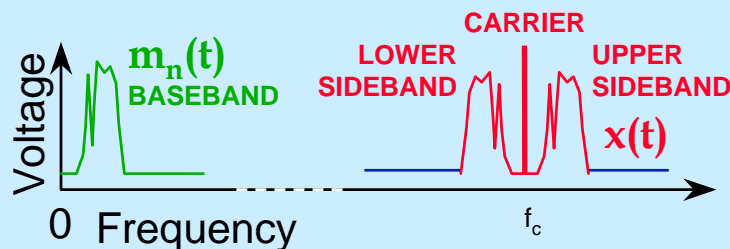
a = modulation index ($0 < a \leq 1$)

$m_n(t)$ = modulating waveform

$\omega_c = 2\pi f_c$, the radian carrier freq.

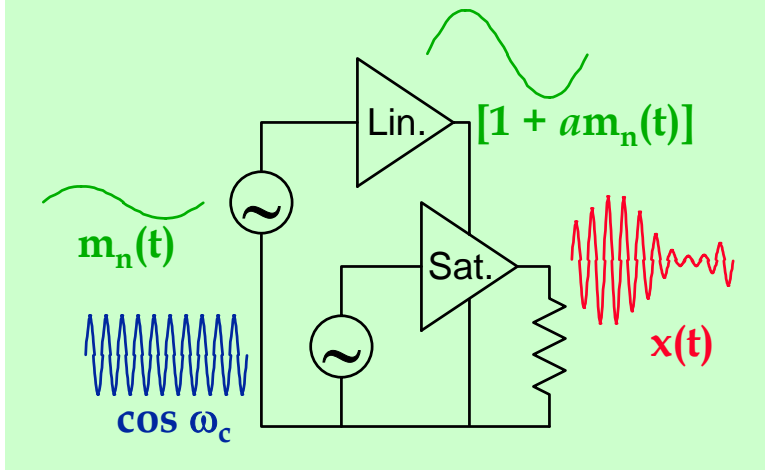
- AM is “linear modulation” -- the spectrum of the baseband signal translates directly into sidebands on both sides of the carrier frequency
- Despite its simplicity, AM has definite drawbacks which complicate its use for wireless systems:
 - Only part of an AM signal’s energy actually carries information (sidebands); the rest is the carrier
 - The two identical sidebands waste bandwidth
 - AM signals can be faithfully amplified only by linear amplifiers
 - AM is highly vulnerable to external noise during transmission
 - AM requires a very high C/I (~30 to 40 dB); otherwise, interference is objectionable

FREQUENCY-DOMAIN VIEW

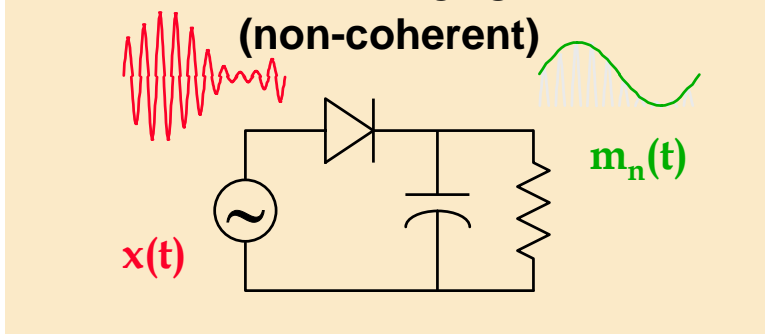


An AM Modulator and Detector

TIME-DOMAIN VIEW: AM MODULATOR



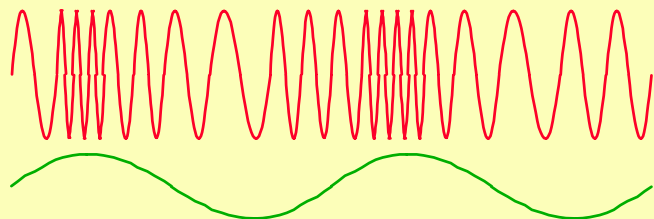
TIME-DOMAIN VIEW: AM DETECTOR (non-coherent)



- AM modulation can be simply accomplished in a saturated amplifier
 - superimpose the modulating waveform on the supply voltage of the saturated amplifier
- AM de-modulation (detection) can be easily performed using a simple envelope detector
 - example: half-wave rectifier
 - this “non-coherent” detection works well if $S/N > 10$ dB.
- AM demodulation can also be performed by coherent detectors
 - incoming signal is mixed (multiplied) with a locally generated carrier
 - enhances performance when S/N ratio is poor (< 10 dB.)

Frequency Modulation (“FM”)

TIME-DOMAIN VIEW



$$S_{FM}(t) = A \cos \left[\omega_c t + \int_{t_0}^t m_\omega m(x) dx + \phi_0 \right]$$

where:

A = signal amplitude (constant)

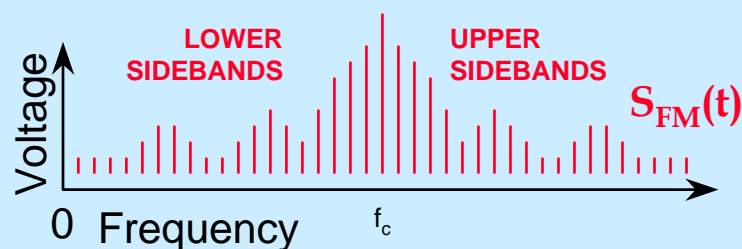
ω_c = radian carrier frequency

m_ω = frequency deviation index

$m(x)$ = modulating signal

ϕ_0 = initial phase

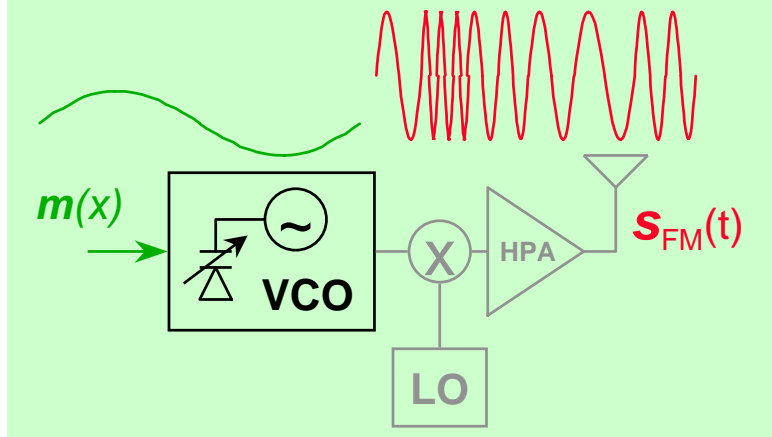
FREQUENCY-DOMAIN VIEW



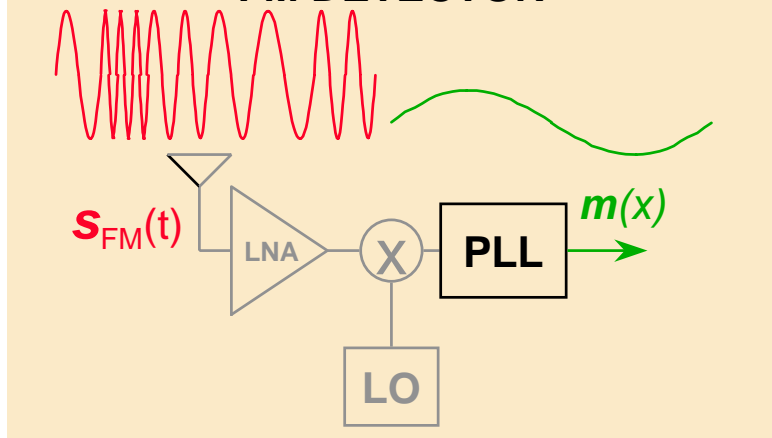
- Frequency Modulation (FM) is a type of *angle* modulation
 - in FM, the instantaneous frequency of the signal is varied by the modulating waveform
- Advantages of FM
 - the amplitude is constant
 - simple saturated amplifiers can be used
 - the signal is relatively immune to external noise
 - the signal is relatively robust; required C/I values are typically 17-18 dB. in wireless applications
- Disadvantages of FM
 - relatively complex detectors are required
 - a large number of sidebands are produced, requiring even larger bandwidth than AM

An FM Modulator and Detector

TIME-DOMAIN VIEW: FM MODULATOR



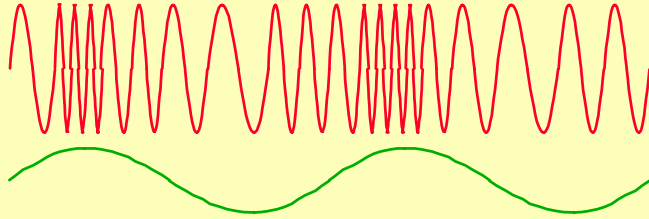
TIME-DOMAIN VIEW: FM DETECTOR



- FM modulation can be accomplished in tuned or voltage-controlled oscillator
 - the modulating signal varies a reactance (varactor, etc.) or otherwise changes the frequency of the oscillator
 - the modulation may be performed at a low intermediate frequency, then heterodyned to a desired communications frequency
- FM de-modulation (detection) can be performed by any of several types of detectors
 - Phase-locked loop (PLL)
 - Pulse shaper and integrator
 - Ratio Detector

Phase Modulation (“PM”)

TIME-DOMAIN VIEW



$$s_{PM}(t) = A \cos \left[\omega_c t + m_\omega m(x) + \varphi_0 \right]$$

where:

A = signal amplitude (constant)

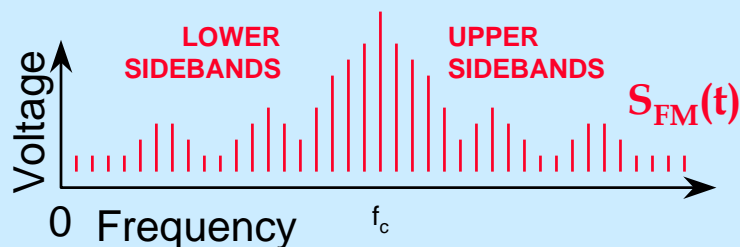
ω_c = radian carrier frequency

m_ω = phase deviation index

$m(x)$ = modulating signal

φ_0 = initial phase

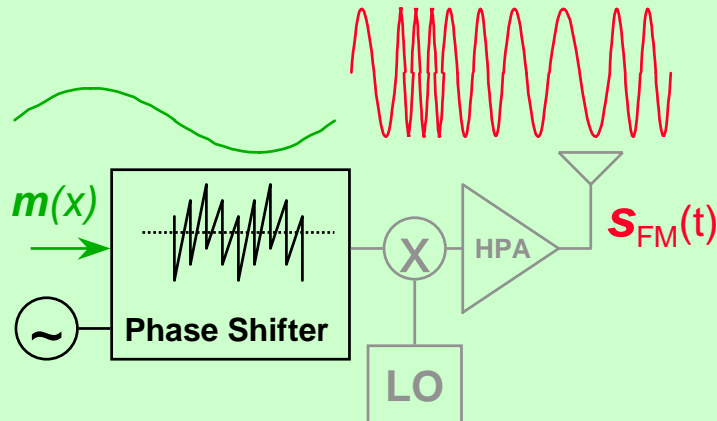
FREQUENCY-DOMAIN VIEW



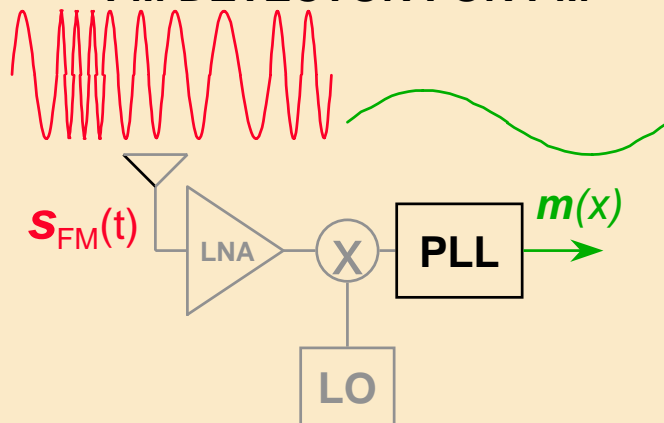
- Phase Modulation (PM) is a type of *angle* modulation, a “sister” of FM
 - the instantaneous *phase* of the signal is varied according to the modulating waveform
- Advantages of PM: similar to FM
 - the amplitude is constant
 - simple saturated amplifiers can be used
 - the signal is relatively immune to external noise
 - the signal is relatively robust; required C/I values are typically 17-18 dB. in wireless applications
- Disadvantages of PM
 - relatively complex detectors are required
 - a large number of sidebands are produced, requiring even larger bandwidth than AM

Generating and Detecting Phase Modulation

TIME-DOMAIN VIEW: PHASE MODULATOR



TIME-DOMAIN VIEW: FM DETECTOR FOR PM



- PM and FM signals can be considered identical with only one exception: in FM, the analog modulating signal is inherently de-emphasized by $1/F$
- Consequences of this realization:
 - the same types of circuitry can be used to generate and detect both analog PM or FM, determined by filtering the modulating signal at baseband
 - FM has poorer signal-to-noise ratio than PM at high modulating frequencies. Therefore, pre-emphasis and de-emphasis are often used in FM systems

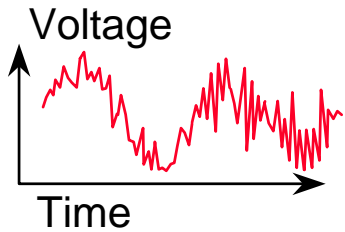
The *phase* of a PM signal is proportional to the amplitude of the modulating signal.

The *phase* of an FM signal is proportional to the integral of the amplitude of the modulating signal.

Modulation and Occupied Bandwidth

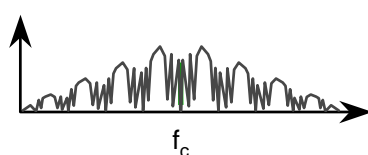
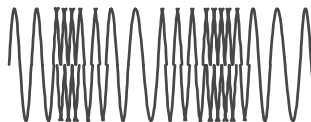
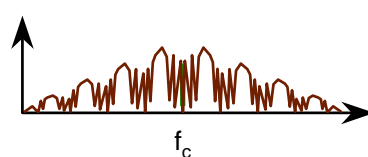
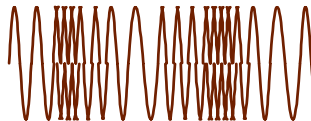
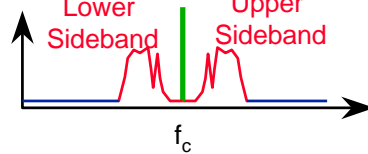
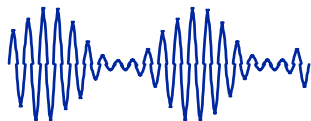
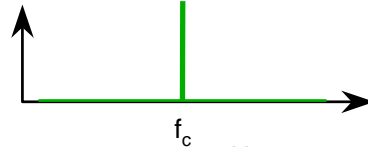
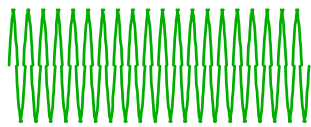
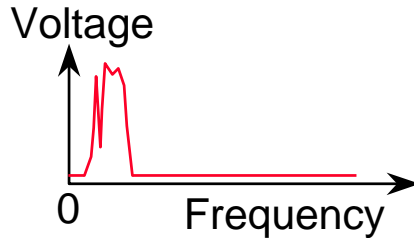
Time-Domain

(as viewed on an Oscilloscope)



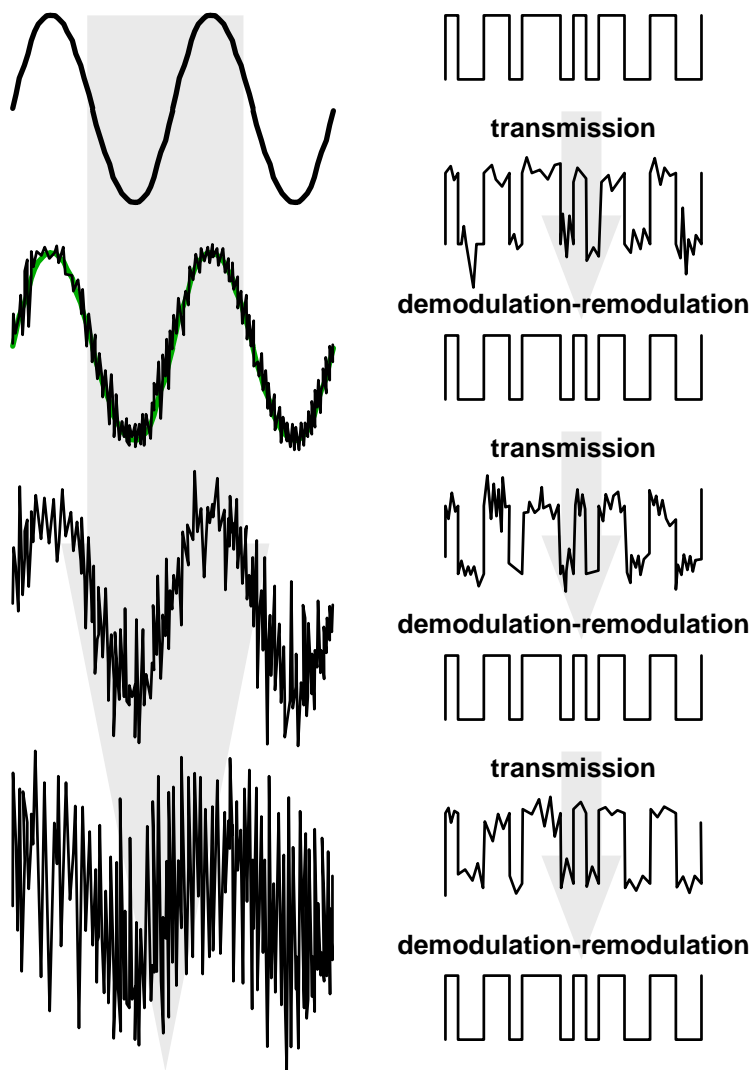
Frequency-Domain

(as viewed on a Spectrum Analyzer)



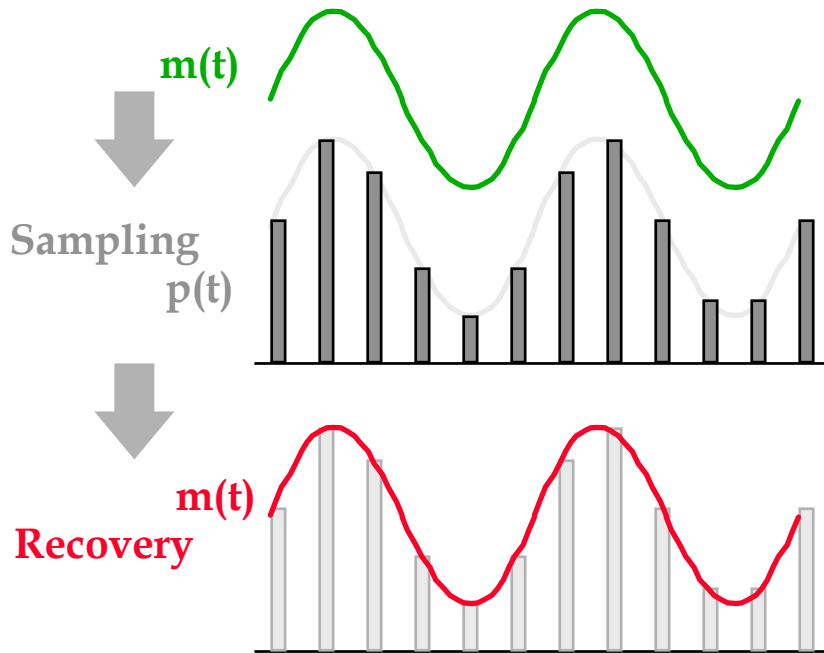
- The bandwidth occupied by a signal depends on:
 - input information bandwidth
 - modulation method
- Information to be transmitted, called “input” or “baseband”
 - bandwidth usually is small, much lower than frequency of carrier
- Unmodulated carrier
 - the carrier itself has **Zero** bandwidth!!
- AM-modulated carrier
 - Notice the upper & lower sidebands
 - total bandwidth = 2 x baseband
- FM-modulated carrier
 - Many sidebands! bandwidth is a complex mathematical function
- PM-modulated carrier
 - Many sidebands! bandwidth is a complex mathematical function

Introduction to Digital Modulation



- The modulating signals shown in previous slides were all analog. It is also possible to quantize modulating signals, restricting them to discrete values, and use such signals to perform digital modulation. Digital modulation has several advantages over analog modulation:
- Digital signals can be more easily regenerated than analog
 - in **analog** systems, the effects of noise and distortion are *cumulative*: each demodulation and remodulation introduces new noise and distortion, added to the noise and distortion from previous demodulations/remodulations.
 - in **digital** systems, each demodulation and remodulation produces a *clean* output signal free of past noise and distortion
- Digital bit streams are ideally suited to many flexible multiplexing schemes

Theory of Digital Modulation: Sampling



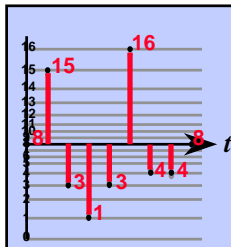
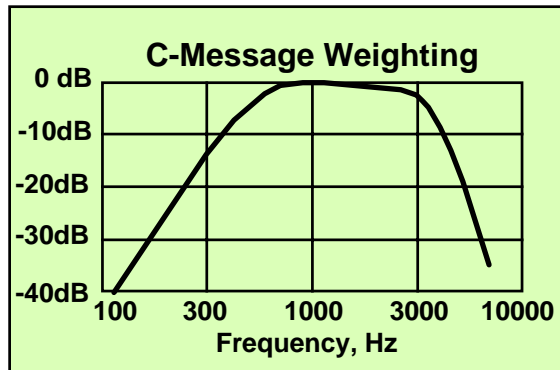
- Voice and other analog signals first must be sampled (converted to digital form) for digital modulation and transmission
- The **sampling theorem** gives the criteria necessary for successful sampling, digital modulation, and demodulation
 - The analog signal must be band-limited (low-pass filtered) to contain no frequencies higher than f_M
 - Sampling must occur at least twice as fast as f_M in the analog signal. This is called the **Nyquist Rate**
- Required Bandwidth for $p(t)$
 - If each sample $p(t)$ is expressed as an n -bit binary number, the bandwidth required to convey $p(t)$ as a digital signal is at least $N \cdot 2 \cdot f_M$
 - this follows **Shannon's Theorem**: at least one Hertz of bandwidth is required to convey one bit per second of data

The Sampling Theorem: Two Parts

- If the signal contains no frequency higher than f_M Hz., it is completely described by specifying its samples taken at instants of time spaced $1/2 f_M$ s.
- The signal can be completely recovered from its samples taken at the rate of $2 f_M$ samples per second or higher.

Sampling Example: the 64 kb/s DS-0

Band-Limiting



Companding

μ -Law

$$y = \text{sgn}(x) \frac{\ln(1 + \mu|x|)}{\ln(1 + \mu)}$$

(where $\mu = 255$)

A-LAW

$$y = \text{sgn}(x) \frac{A|x|}{\ln(1+A)} \quad \text{for } 0 \leq x \leq \frac{1}{A}$$

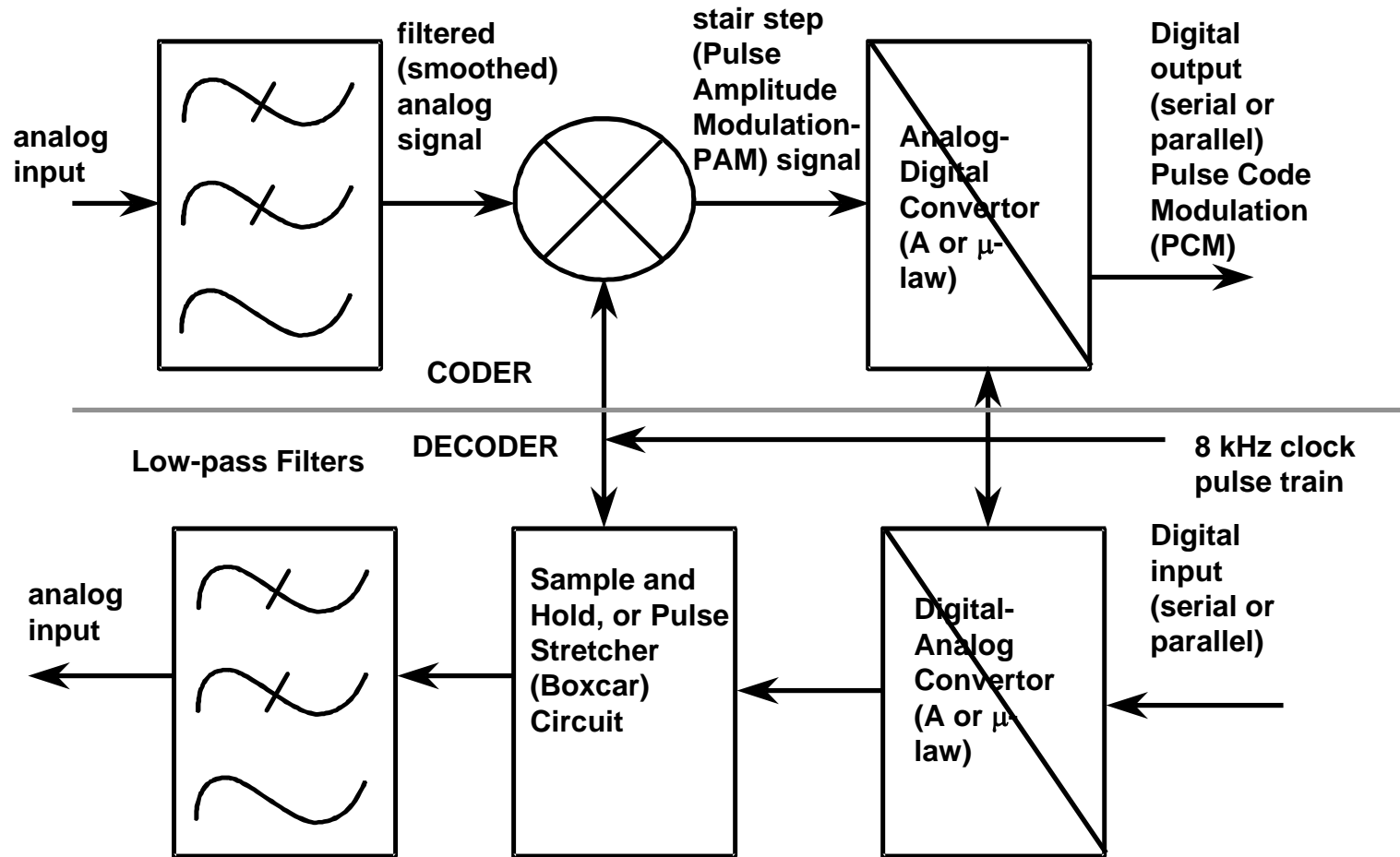
$$y = \text{sgn}(x) \frac{\ln(1+A|x|)}{\ln(1+A)} \quad \text{for } \frac{1}{A} < x \leq 1$$

(where $A = 87.6$)

x = analog audio voltage
y = quantized level (digital)

- Telephony has adopted a world-wide PCM standard digital signal employing a 64 kb/s stream derived from sampled voice data
- Voice waveforms are band-limited
 - upper cutoff between 3500-4000 Hz. to avoid aliasing
 - rolloff below 300 Hz. to minimize vulnerability to “hum” from AC power mains
- Voice waveforms sampled at 8000/second rate
 - 8000 samples x 1 byte = 64,000 bits/second
 - A>D conversion is non-linear, one byte per sample, thus 256 quantized levels are possible
 - Levels are defined logarithmically rather than linearly to accommodate a wider range of audio levels with minimum distortion
 - μ -law companding (popular in North America & Japan)
 - A-law companding (used in most other countries)
- A>D and D>A functions are performed in a CODEC (coder-decoder) (see following figure)

CODEC Block Diagram



Digital Signals: the Bandwidth Penalty

- Although digital modulation has many advantages, it requires substantially more bandwidth than corresponding analog methods
- Various techniques are used to minimize and compensate for the bandwidth-appetite of digital
 - Advanced modulation techniques: maximizing the number of bits carried per hertz of bandwidth
 - QPSK, DQPSK, GMSK, and other advanced forms
 - Compression of the content of digital signals: reducing the number of bits required to carry the message
 - for voice information content: Vocoding techniques (VSELP, RLP-LTP, CELP, etc.)
 - for data content: various compression techniques

Vocoders: Compression vs. Distortion

- Objective: to significantly reduce the number of bits which must be transmitted, but without creating objectionable levels of distortion
- We are concerned mainly with telephone applications, with voice signal already band-limited to 4 kHz. max. and sampled at 8 kHz.
- The objective is *toll-quality* voice reproduction
- General Categories of Speech Coders
 - Waveform Coders
 - attempt to re-create the input waveform
 - good speech quality but at relatively high bit rates
 - Vocoders
 - attempt to re-create the sound as perceived by humans
 - quantize and mimic speech-parameter-defined properties
 - lower bit rates but at some penalty in speech quality
 - Hybrid Coders
 - mixed approach, using elements of Waveform Coders & Vocoders
 - use vector quantization against a codebook reference
 - low bit rates and good quality speech

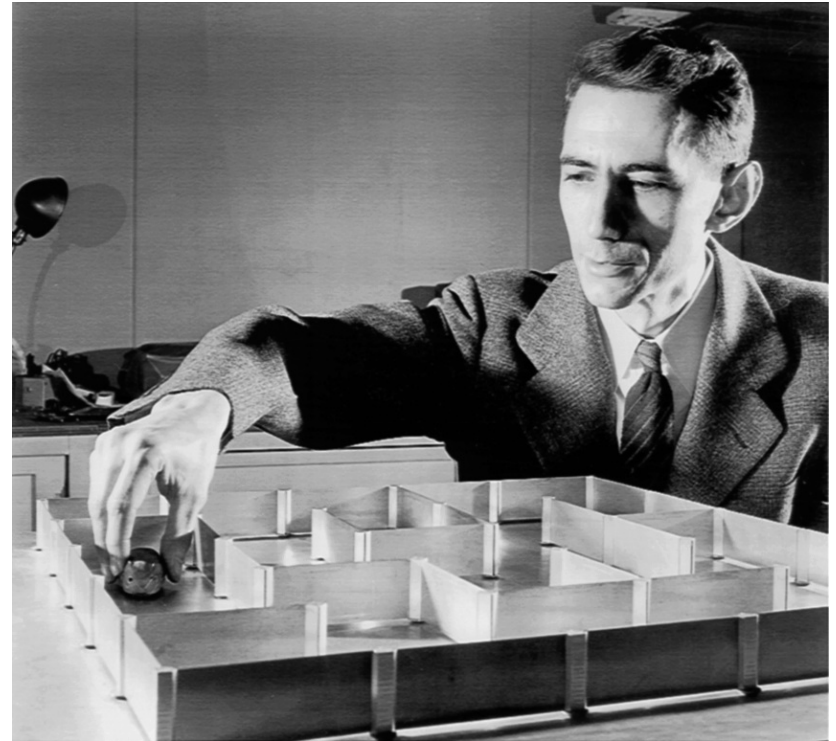
Symbol Rate, bit/s/Hz and Constant Envelope PM

- Bit rate= (symbols/sec)*(bits/symbol)
- Use of a rapid symbol rate requires increased bandwidth in a non-bandlimited channel
 - Unless phase transitions are synchronized with carrier zero voltage crossings, the resulting waveform discontinuities will require large bandwidth
- Using a rapid symbol rate together with narrow band channel filtering causes the envelope of the resulting signal to fluctuate
 - Envelope oscillation occurs when symbol rate exceeds channel bandwidth
 - Such a non-constant envelope requires use of a linear RF power amplifier, which is more complex and less efficient than constant envelope waveform with a “Class C” power amplifier

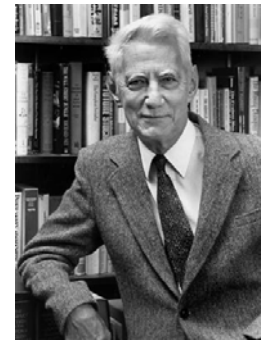
Digital Modulation

Claude Shannon: The “Einstein” of Information Theory and Signal Science

- The core idea that makes CDMA possible was first explained by Claude Shannon, a Bell Labs research mathematician
- Shannon's work relates amount of information carried, channel bandwidth, signal-to-noise-ratio, and detection error probability
 - It shows the theoretical upper limit attainable



In 1948 **Claude Shannon** published his landmark paper on information theory, ***A Mathematical Theory of Communication***. He observed that "the fundamental problem of communication is that of reproducing at one point either exactly or approximately a message selected at another point." His paper so clearly established the foundations of information theory that his framework and terminology are standard today. Shannon died Feb. 24, 2001, at age 84.



Digital Modulation Systems

- Each symbol of a digitally modulated RF signal conveys a number of bits of information
 - determined by the number of degrees of modulation freedom
- More complex modulation schemes can carry more bits per symbol in a given bandwidth, but require better signal-to-noise ratios
- The actual number of bits per second which can be conveyed in a given bandwidth under given signal-to-noise conditions is described by Shannon's equations

Modulation Scheme	Shannon Limit, Bits/Hz
BPSK	1 b/s/Hz
QPSK	2 b/s/Hz
8PSK	3 b/s/Hz
16 QAM	4 b/s/Hz
32 QAM	5 b/s/Hz
64 QAM	6 b/s/Hz
256 QAM	8 b/s/Hz

SHANNON'S CAPACITY EQUATION

$$C = B_{\omega} \log_2 \left[1 + \frac{S}{N} \right]$$

B_{ω} = bandwidth in Hertz

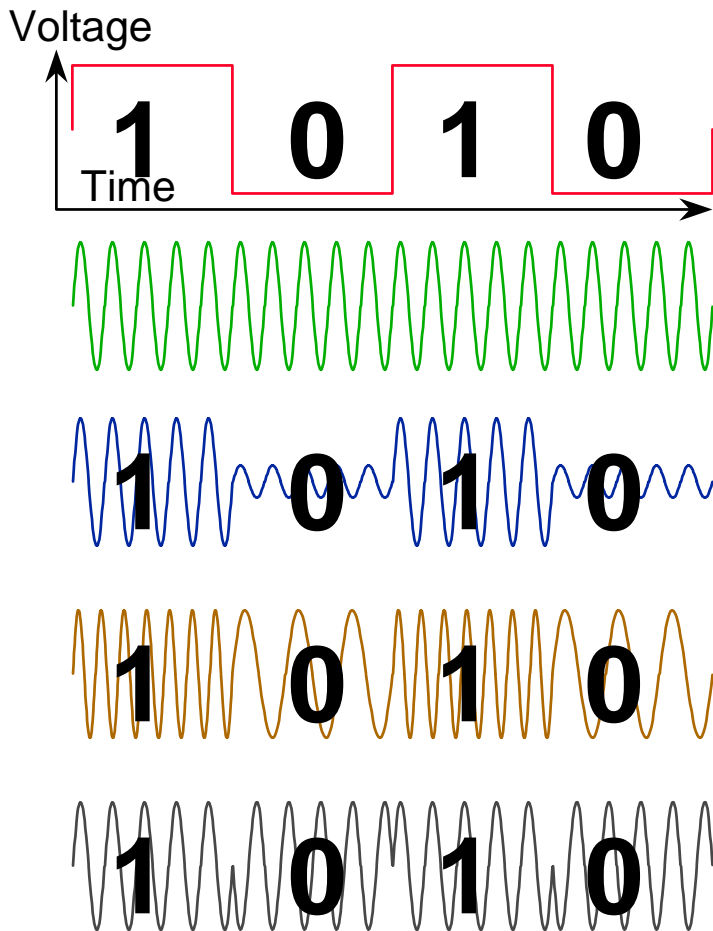
C = channel capacity in bits/second

S = signal power

N = noise power

Modulation by Digital Inputs

Our previous modulation examples used continuously-variable analog inputs. If we quantize the inputs, restricting them to digital values, we will produce digital modulation.

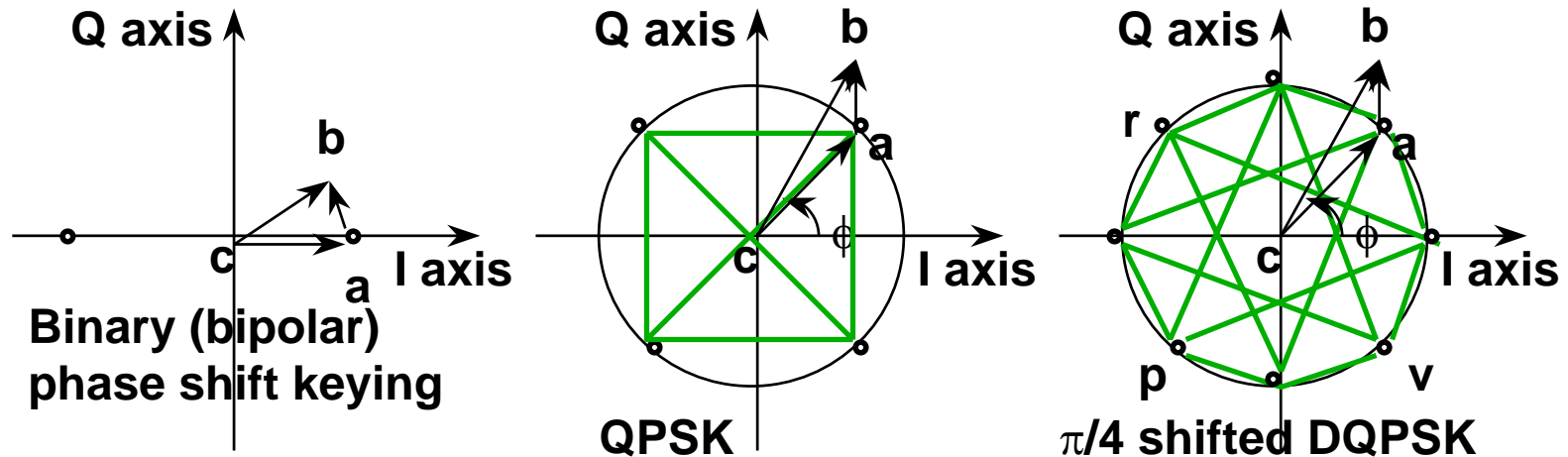


- For example, modulate a signal with this **digital** waveform. No more continuous analog variations, now we're "shifting" between discrete levels. We call this "shift keying".
 - The user gets to decide what levels mean "0" and "1" -- there are no inherent values
- **Steady Carrier** without modulation
- **Amplitude Shift Keying**
ASK applications: digital microwave
- **Frequency Shift Keying**
FSK applications: control messages in AMPS cellular; TDMA cellular
- **Phase Shift Keying**
PSK applications: TDMA cellular, GSM & PCS-1900

Digital Modulation Schemes

- There are many different schemes for digital modulation, each a compromise between complexity, immunity to errors in transmission, required channel bandwidth, and possible requirement for linear amplifiers
- Linear Modulation Techniques
 - BPSK Binary Phase Shift Keying
 - DPSK Differential Phase Shift Keying
 - **QPSK** Quadrature Phase Shift Keying ***IS-95 CDMA forward link***
 - Offset QPSK ***IS-95 CDMA reverse link***
 - Pi/4 DQPSK ***IS-54, IS-136 control and traffic channels***
- Constant Envelope Modulation Schemes
 - BFSK Binary Frequency Shift Keying ***AMPS control channels***
 - MSK Minimum Shift Keying
 - GMSK Gaussian Minimum Shift Keying ***GSM systems, CDPD***
- Hybrid Combinations of Linear and Constant Envelope Modulation
 - MPSK M-ary Phase Shift Keying
 - QAM M-ary Quadrature Amplitude Modulation
 - MFSK M-ary Frequency Shift Keying ***FLEX paging protocol***
- Spread Spectrum Multiple Access Techniques
 - DSSS Direct-Sequence Spread Spectrum ***IS-95 CDMA***
 - FHSS Frequency-Hopping Spread Spectrum

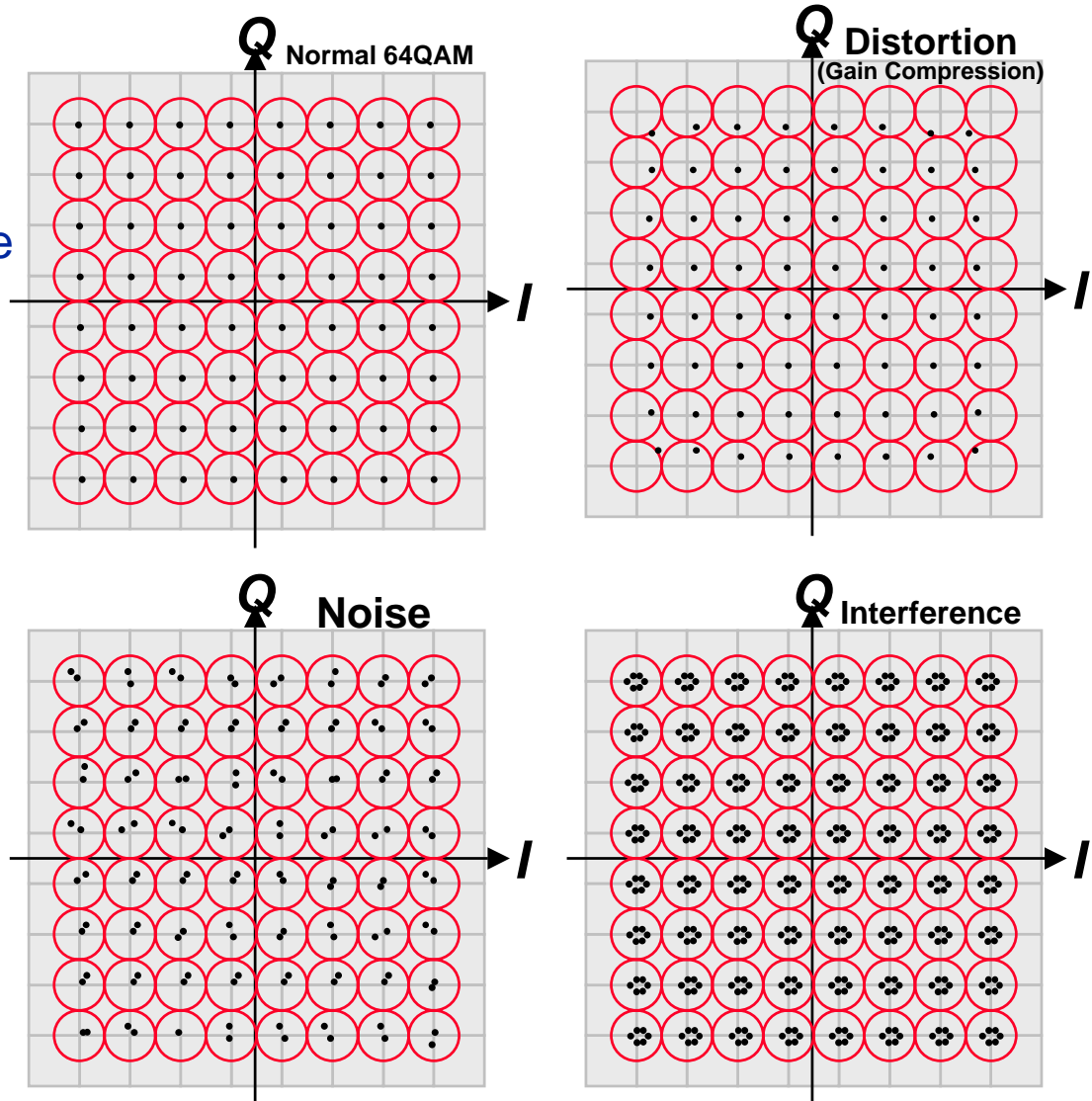
Phase-Plane (Argand) Diagrams for BPSK, QPSK, $\pi/4$ DQPSK



The I axis is in-phase with a carrier reference signal. Each dot represents a digital code value. The decision area is bounded by a sector (180 or 90 deg) around the point. QPSK may use absolute or differential coding. Phase change sequences shown by green lines may occur. Transitions from a to p,r, or v are permitted, others are not. Phasor ab represents additive interference, ϕ the resulting phase angle.

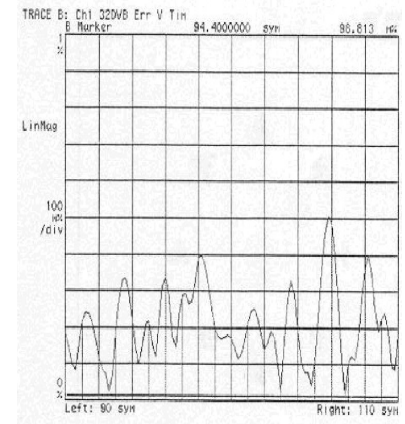
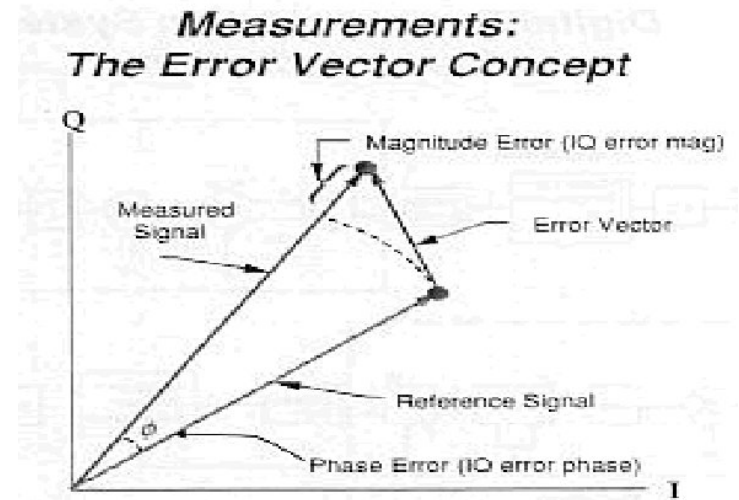
Error Vulnerabilities of Higher-Order Modulation Schemes

- Higher-Order Modulation Schemes (16PSK, 32QAM, 64QAM...) are more vulnerable to transmission errors than the simpler, more rugged schemes (BPSK, QPSK)
 - Closely-packed constellations leave little room for vector error
- Non-linearities (gain compression, clipping, reflections within antenna system) “warp” the constellation
- Noise and long-delayed echoes cause “scatter” around constellation points
- Interference blurs constellation points into “rings” of error



Error Vector Magnitude and ρ (“Rho”)

- A common measurement of overall error is Error Vector Magnitude “EVM”
 - usually a small fraction of total vector amplitude, ~ 0.1
- EVM is usually averaged over a large number of symbols
 - Root-mean-square (RMS)
- Commercial test equipment for BTS maintenance measures EVM
- Signal quality is often expressed as 1-EVM
 - normally called ρ (“Rho”)
 - typically 0.89-0.96



Digital Modulation Schemes: Binary FSK

- Binary Frequency Shift Keying is the modulation scheme used to carry digital information on the AMPS analog cellular control channel
- The constant-amplitude carrier signal is switched between two frequencies according to the binary value of the message bits
- In AMPS control channels, the two FSK frequencies are 8 kHz. above and below the channel center frequency and the bit rate is 10 kb/s.
- Required bandwidth: Carson's Rule gives the bandwidth required:
 $BT = 2\Delta f + 2B$, where:
 BT = total bandwidth of BFSK signal
 Δf = difference between the two frequencies employed
 B = bandwidth of the digital baseband signal
- Binary FSK signals can be detected non-coherently or coherently

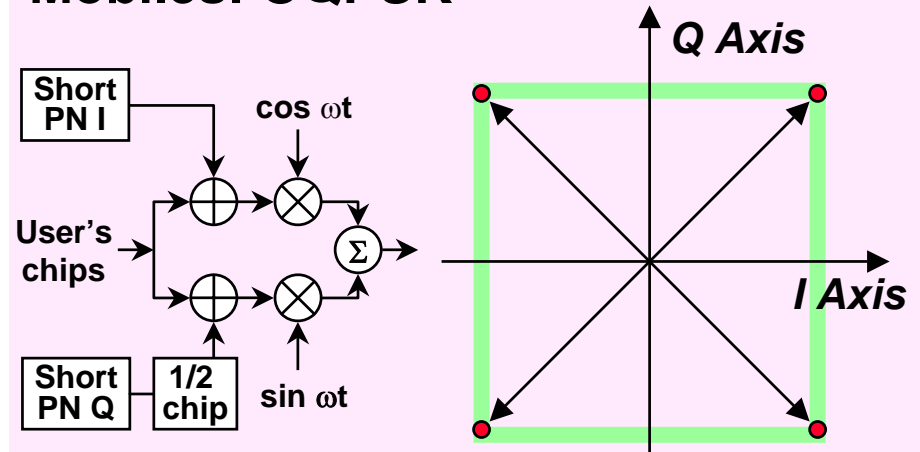
Digital Modulation: GMSK for GSM and CDPD

- MSK (Minimum Shift Keying) is a version of FSK in which the peak frequency deviation is set equal to half the bit rate. This is the minimum frequency separation that allows orthogonal detection of the two binary states
- Advantages of MSK:
 - constant envelope, spectral efficiency, good BER performance, self-synchronizing capabilities
- GMSK is a derivative of MSK
 - before modulation, the message waveform (in NRZ format) is fed through a Gaussian filter to accomplish pulse shaping
 - this greatly reduces the sidelobes in the signal's spectrum
 - this introduces a small penalty in BER performance, but it has been shown that the mobile channel introduces an irreducible error rate larger than the GMSK penalty anyway. Thus there is no effective penalty for using GMSK

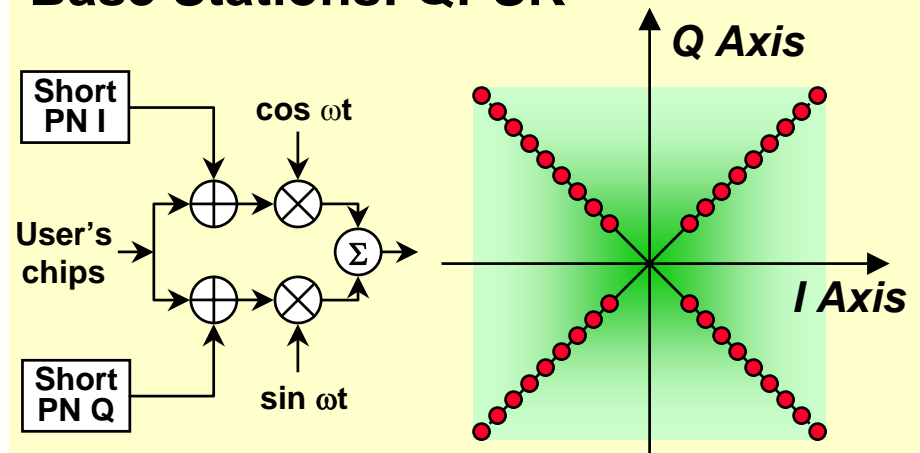
Modulation used in IS-95 CDMA Systems

- CDMA mobiles use offset QPSK modulation
 - the Q-sequence is delayed half a chip, so that I and Q never change simultaneously and the mobile TX never passes through (0,0)
- CDMA base stations use QPSK modulation
 - every signal (voice, pilot, sync, paging) has its own amplitude, so the transmitter is unavoidably going through (0,0) sometimes; no reason to include 1/2 chip delay

Mobiles: OQPSK

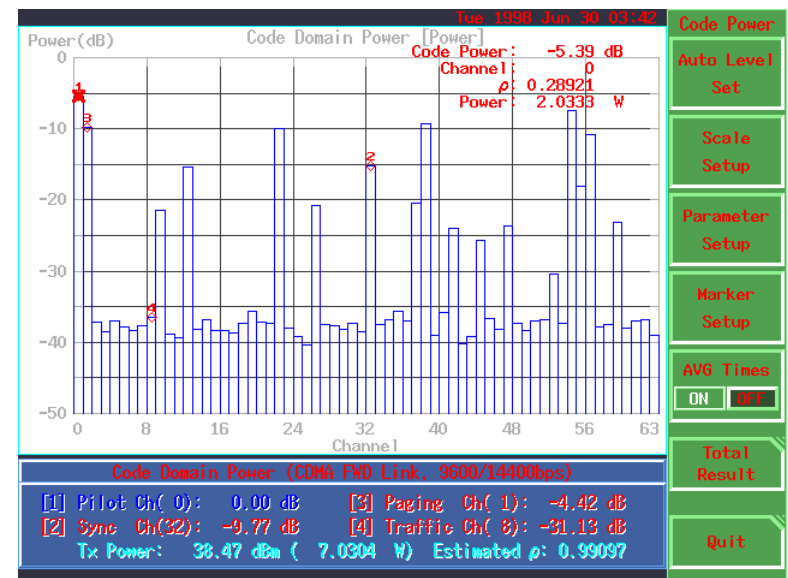
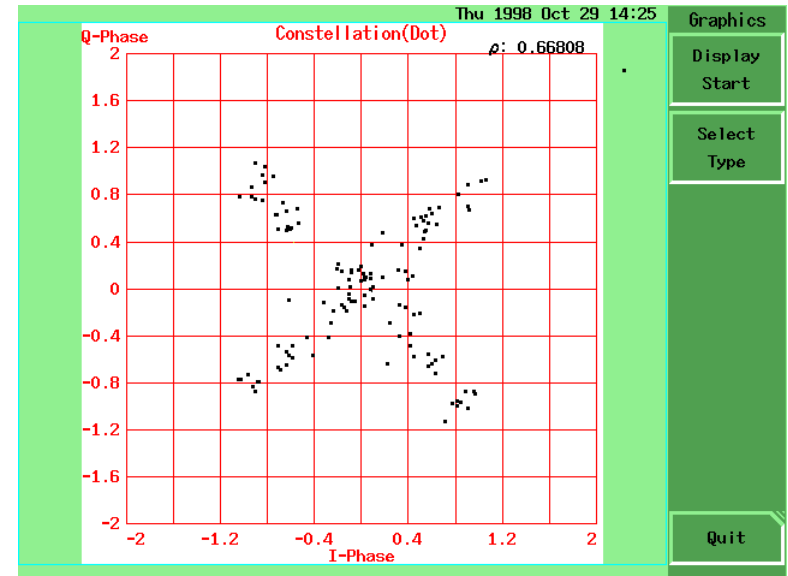


Base Stations: QPSK



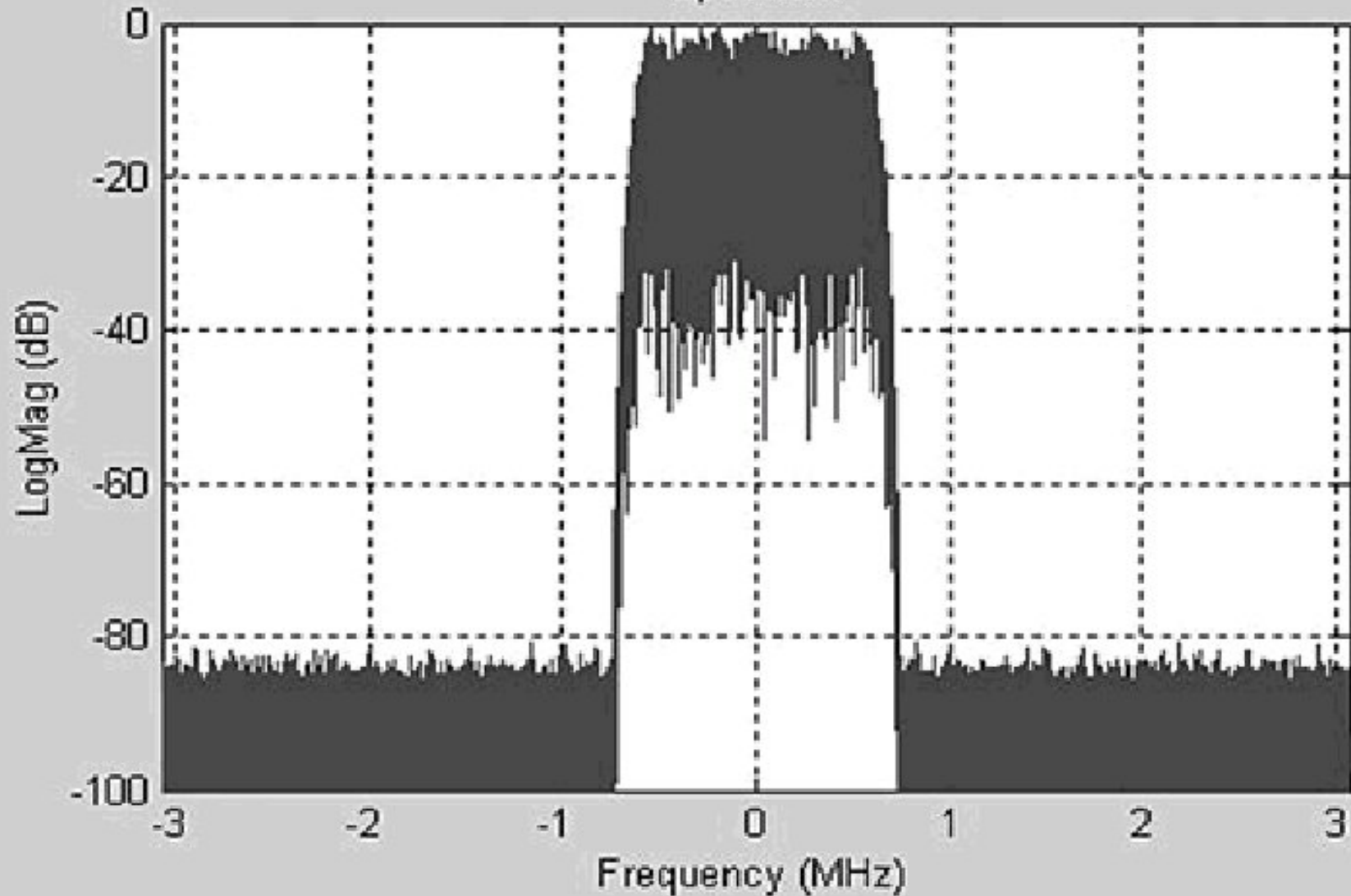
CDMA Base Station Modulation Views

- The view at top right shows the actual measured QPSK phase constellation of a CDMA base station in normal service
- The view at bottom right shows the measured power in the code domain for each walsh code on a CDMA BTS in actual service
 - Notice that not all walsh codes are active
 - Pilot, Sync, Paging, and certain traffic channels are in use



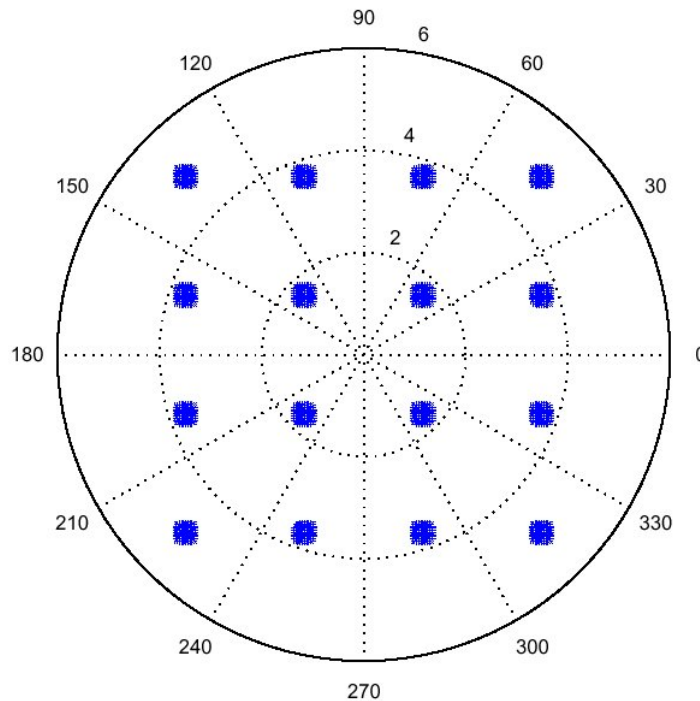
File Edit Tools

Spectrum

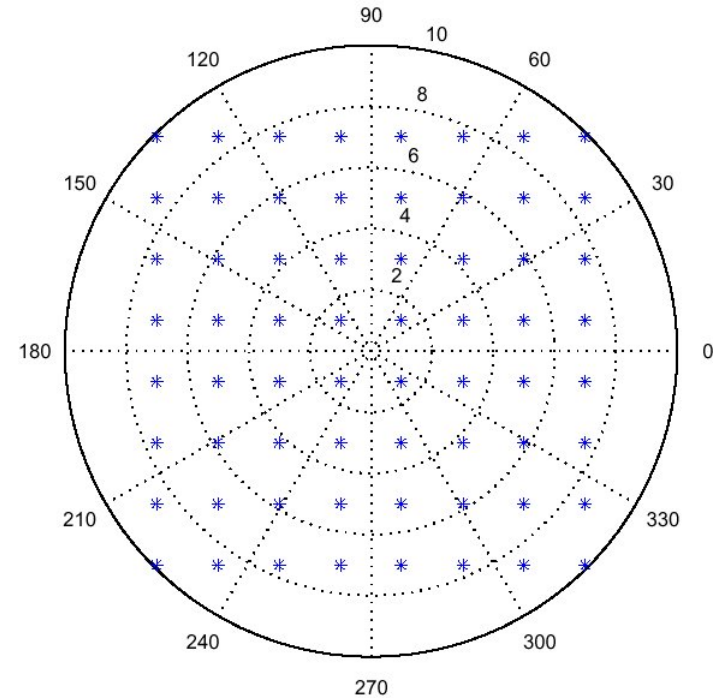


1xEV DO and 1xEV DV Constellations

16-QAM



64-QAM



- Dynamic selection of modulation type, coding scheme, and data rate squeeze the best performance out of each moment
- Although complex modulation schemes pack large amounts of data into a relatively small bandwidth, they are very vulnerable to noise and distortion during transmission