Digital Communication System Overview

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Contents

1. Nyquist Filter

- Communication system: sending as many bits as possible, reliably, using a band-limited noisy channel.
- Baud rate is related to the bandwidth
  - Baud rate: Symbols per second (1/Ts)
- Bits per symbol is determined by the SNR. (N)
- Bit rate R = N/Ts, BW efficiency = R/B

Can we send arbitrary amount of data for a given channel and power?

- Shannon's channel capacity
  - The capacity is proportional to the bandwidth assigned
  - The capacity is also dependent on the power vs noise per bandwidth

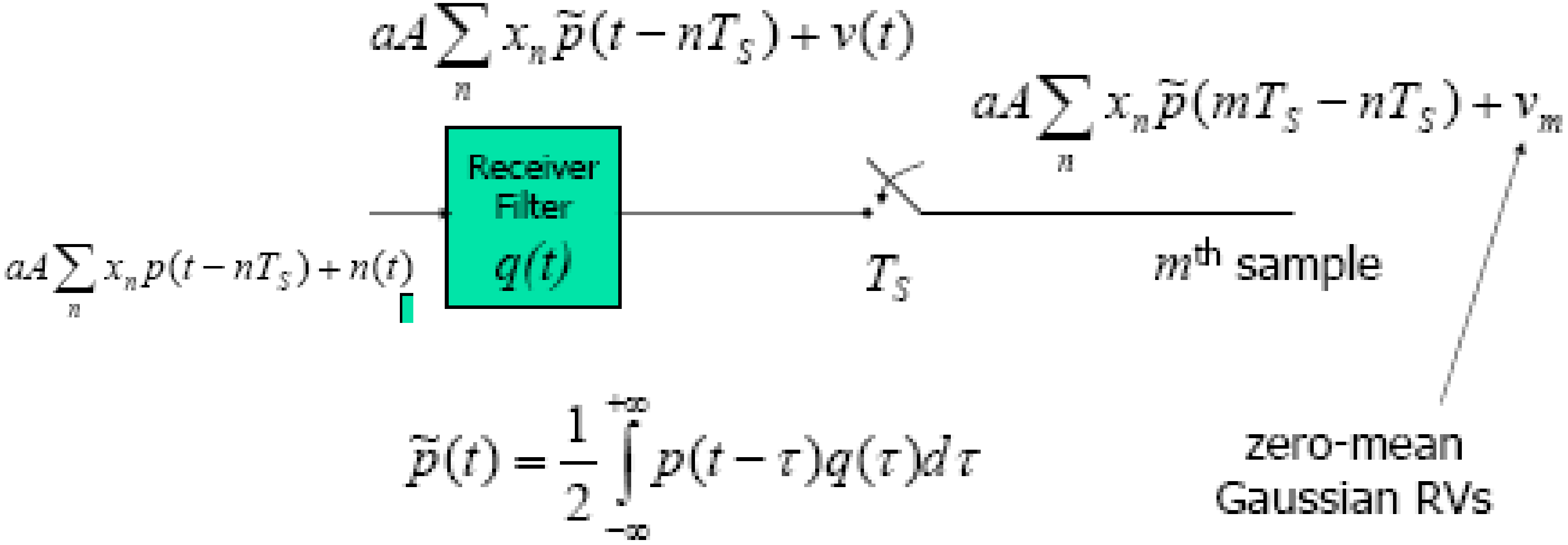
$$C = W \log\left(1 + \frac{P}{N_0 W}\right)$$

-large W, low P

-Small W, large P: wired channel?

Sampling in the receiver

- The baseband representation of the receiver in additive Gaussian noise



$$aA\sum_n x_n p(t-nT_S)+n(t)$$

$$aA\sum_n x_n \tilde{p}(t-nT_S)+v(t)$$

Receiver Filter $q(t)$

T_S

$$aA\sum_n x_n \tilde{p}(mT_S-nT_S)+v_m$$

m^{th} sample

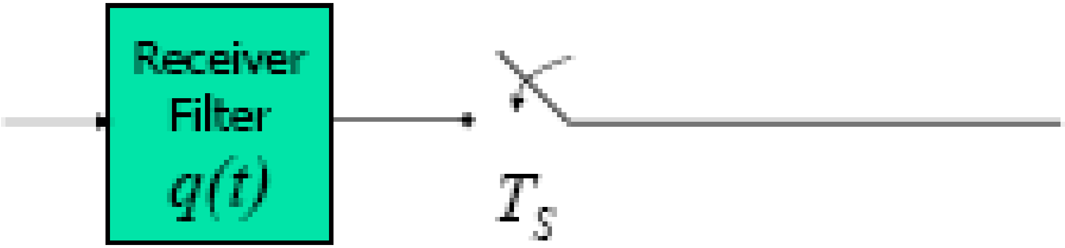
zero-mean Gaussian RVs

$$\tilde{p}(t)=\frac{1}{2}\int_{-\infty}^{+\infty} p(t-\tau)q(\tau)d\tau$$

Ideal situation

- The mth sample depends on only the mth symbol and the noise.

$$aA\sum_{n} x_n \tilde{p}(mT_S - nT_S) + v_m = aAx_m + v_m$$



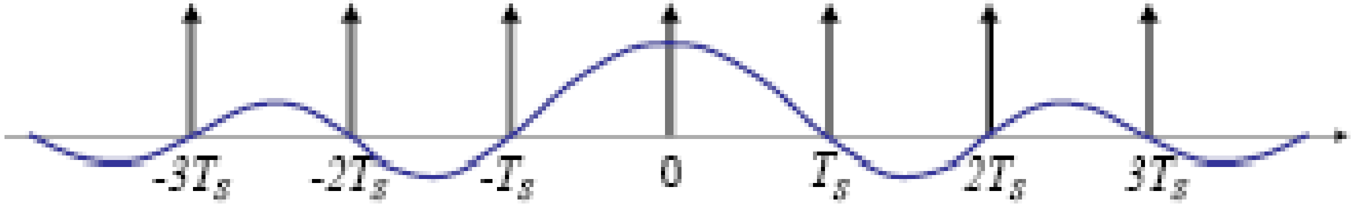
Necessary condition of non-overlapping

- To have this ideal situation, we must have

$$\tilde{p}(mT_S - nT_S) = \begin{cases} 1 & m = n \\ 0 & m \neq n \end{cases}$$

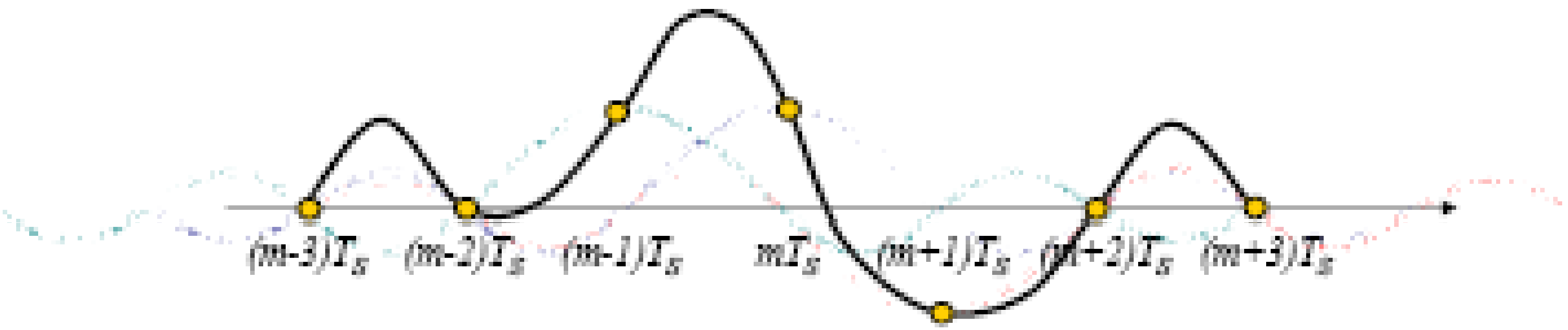
  - or, alternatively,

$$\tilde{p}(t) \sum_{n=-\infty}^{+\infty} \delta(t - nT_S) = \delta(t)$$



Received signal example

- Suppose the m-1st, mth, and m+1st symbols were +1, +1, -1, respectively

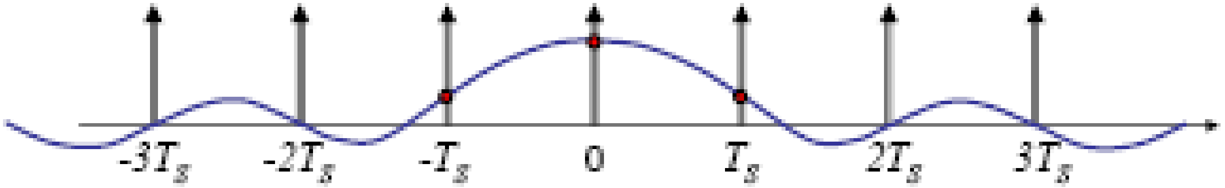


The signal is constrained only at the sample points

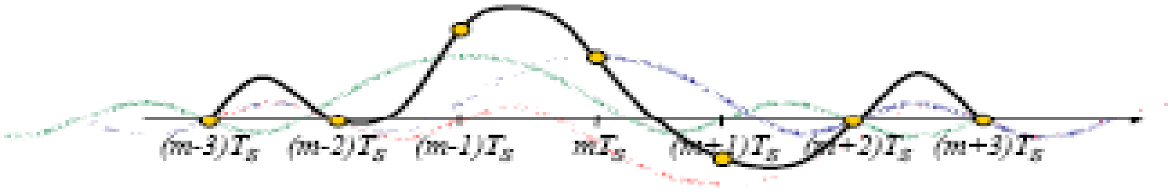
Non-ideal situation: Inter Symbol Interference (ISI)

- Suppose the received pulse did not satisfy the condition, but did this instead:

$$\tilde{p}(t)\sum_{n=-\infty}^{+\infty}\delta(t-nT_S)=\delta(t)+0.3\delta(t-T_S)+0.3\delta(t+T_S)$$



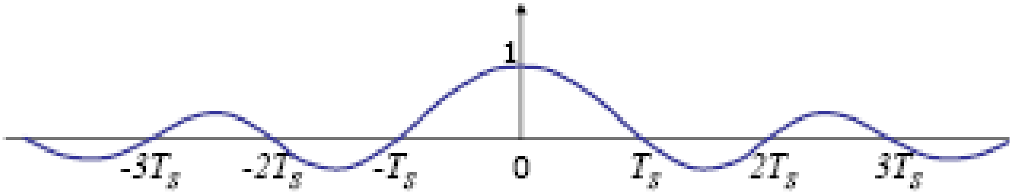
$$1.3+v_{m-1},\ 1+v_m,\ -0.7+v_{m+1}$$

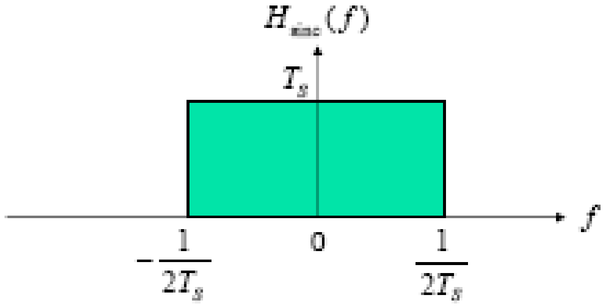


Nyquist pulses

- Pulses that satisfy the condition for no ISI are called Nyquist pulses
- One example: Sinc Pulse
  - Minimum bandwidth, infinite length filter - non-causal

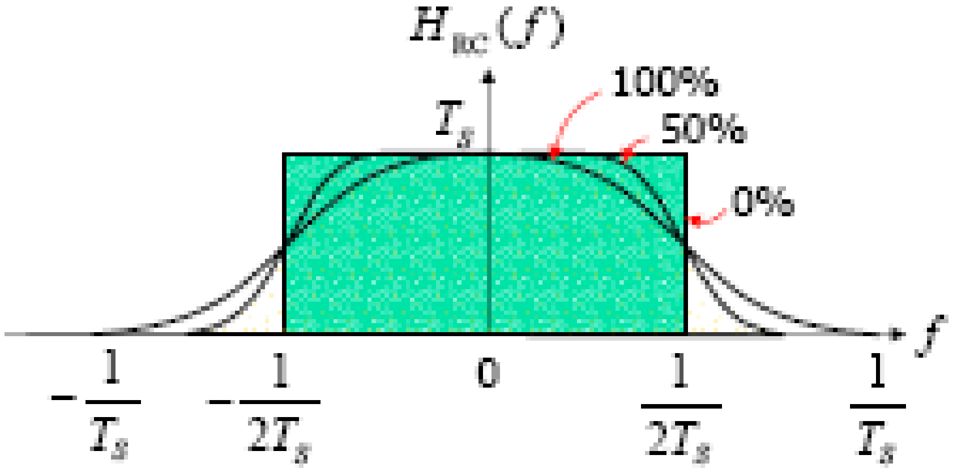
$$\mathrm{sinc}(t/T_S) = \frac{\sin(\pi t/T_S)}{\pi t/T_S}$$





Practical filters

- Gives some excess bandwidth, (alpha)
  - Filter length is inversely proportional to the transition banc



$$H_{RC}(f)=\begin{cases} T_s & \\ \frac{T_s}{2}\left[1-\sin\frac{\pi T_s}{\alpha}\left(|f|-\frac{1}{2T_s}\right)\right] & (1-\alpha)/2T_s \le |f| \le (1+\alpha)/2T_s \\ 0 & (1+\alpha)/2T_s \le |f| \end{cases}$$

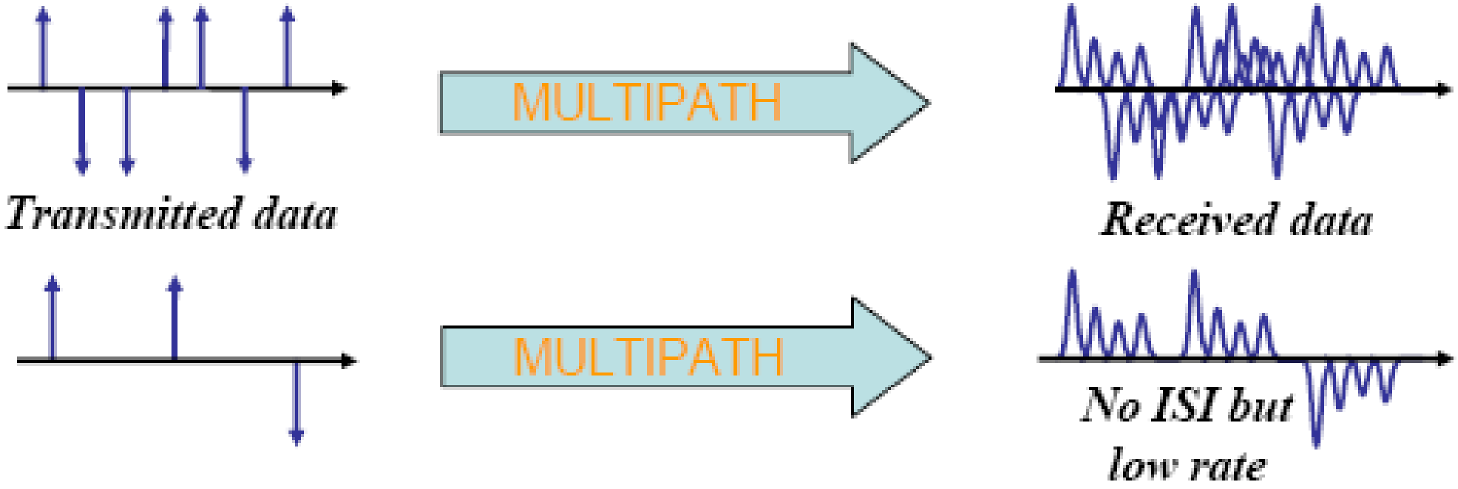
$$p_{RC}(t)=F^{-1}\{H_{RC}(f)\}=\frac{\sin \pi t/T_s}{\pi t/T_s}\frac{\cos \alpha\pi t/T_s}{1-4\alpha^2t^2/T_s^2}$$

Bandwidth Requirements

- **The minimum bandwidth required for a lowpass filter to satisfy Nyquist constraints is 1/2T Hz**
  - such a brickwall filter would require infinite order thus all practical Nyquist filters havesome "excess" bandwidth (20-100%)
  - The baseband spectrum actually extends from -1/2T to +1/2T Hz thus the minimum bandwidth of a modulated signal is 1/T Hz
  - Ex. Using Nyquist filters with 20% excess bandwidth, a 5 Mbaud signal will occupy 6 MHz of RF bandwidth

- **The transmitter requires filters to satisfy FCC masks and the receiver requires filters to reject out-of-band noise**
  - The optimum partitioning of the Nyquist filtering in an additive white Gaussian noise (AWGN) channel is *equally* between the transmitter and receiver resulting in the so-called square-root Nyquist matched filter pair

Inter-Symbol Interference (ISI)

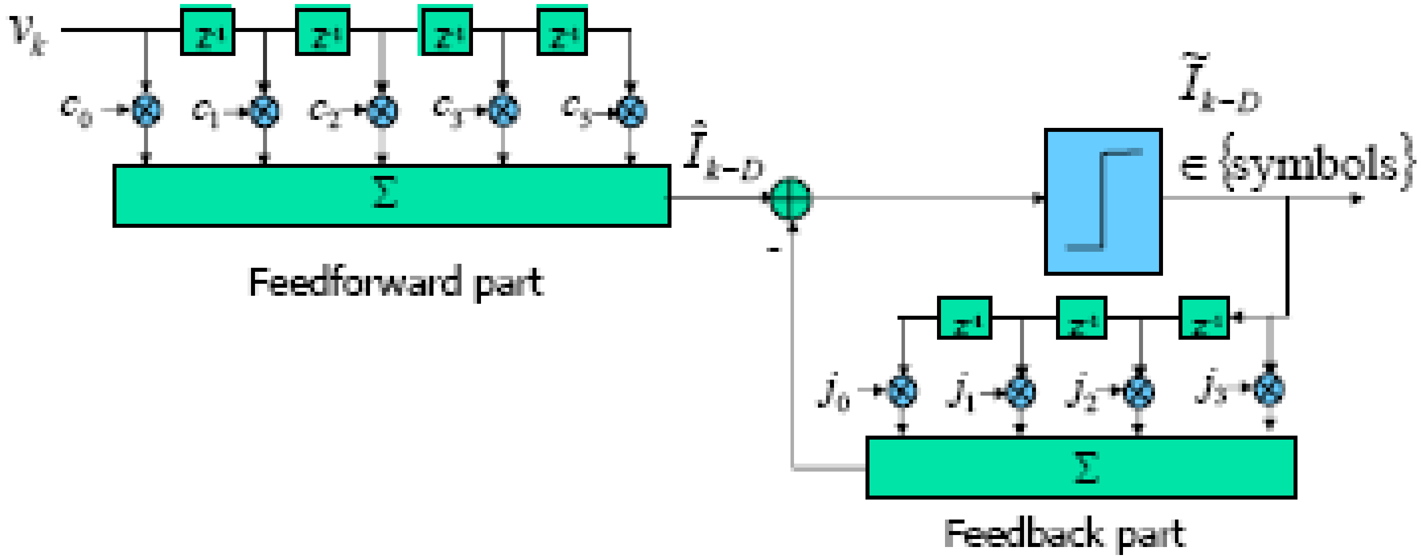


Solutions

- Lower data rate
- Equalization or combining
  - TDMA and CDMA
- Code as multiple low data rate streams
  - Each stream at different frequency – OFDM

Decision Feedback Equalizer

- The DFE has a feedforward part and a feedback part
- The feedback part feeds back detected symbols (the detector is the nonlinearity)



- Problem: the filter length needs to be long
  - Slow convergence, … several problems

Summary

- Intersymbol interference (ISI) can dominate BER (bit error rate) in bandlimited channels.
- Nyquist pulses do not cause ISI in an AWGN channel
- Raised cosine is a popular Nyquist pulse, this is a finite length filter, but needs more bandwidth.
- ISI due to multipath channels can be removed by adaptive channel equalization.

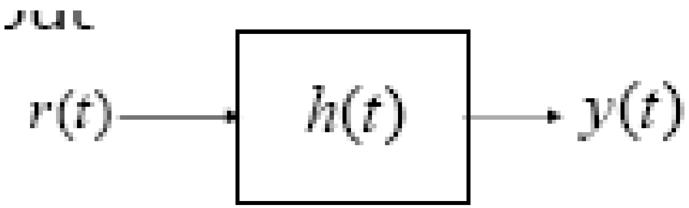
2. Matched Filter

- Assume AWGN
- We want the filter that yields the highest SNR (Signal to Noise Ratio).
- The optimum filter is called the matched filter.
- 예: 잡음이 심하다.
  - A 가 큰소리로 말하는데 1이라고 하는 것 같다.
  - B 가 작은소리로 말하는데 0이라고 하는 것 같다.
  - 당신의 추측은: 1 or 0?

  - 만약 c, d도 작은 소리로 말하는데 모두 0이라고 하는 것 같다.

Set-up

- Suppose r(t) = s(t) + n(t), where n(t) is WGN with spectral height No/2, and s(t) is a signal with a duration T.



$$y_s(t) = \int_0^t s(u)h(t-u)du \qquad y_n(t) = \int_0^t n(u)h(t-u)du$$

Signal to Noise Ratio

$$\mathrm{SNR} = \frac{y_s^2(t)}{E\left[y_n^2(t)\right]}$$

$$= \frac{\left[\int_0^t s(u)h(t-u)du\right]^2}{E\left[\int_0^t n(u)h(t-u)du\right]^2}$$

$$E\left[y_n^2(t)\right]$$

$$= \int_0^t \int_0^t \frac{N_0}{2} \delta(u-v)h(t-u)h(t-v)dudv$$

$$= \frac{N_0}{2} \int_0^t h^2(t-u)du$$

SNR maximization

- To optimize SNR, choose h(u) to maximize the numerator

$$\mathrm{SNR} = \frac{\left[\int_0^t s(u)h(t-u)du\right]^2}{\frac{N_0}{2}\int_0^t h^2(t-u)du}$$

Cauchy-Schwarz Inequality

- Let S and Q be points in the Hilbert space of square-integrable functions
- Then,

$$\left[\int_0^t s(u)q(u)du\right]^2 \leq \int_0^t s^2(u)du \int_0^t q^2(u)du$$

- Equality is reached when $cs(u) = q(u)$

Simplify optimal SNR

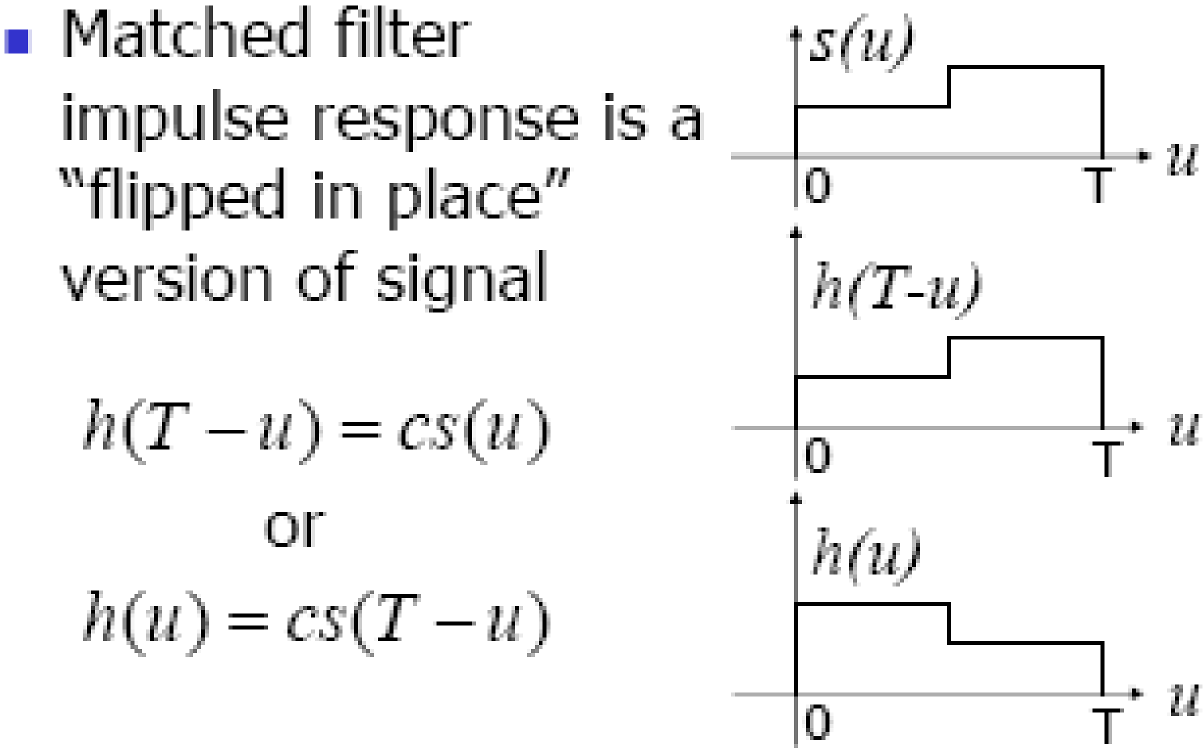
- Substitute $h(t-u)=cs(u)$

$$\mathrm{SNR}^{\mathrm{opt}}(t)=\frac{\left[c\int_0^t s^2(u)du\right]^2}{\frac{N_0c^2}{2}\int_0^t s^2(u)du}=\frac{\int_0^t s^2(u)du}{\frac{N_0}{2}}$$

$$=\frac{\int_0^T s^2(u)du}{\frac{N_0}{2}}=\frac{2\mathcal{E}_s}{N_0}$$

Max SNR filter=Matched filter

- Gives more weight when the signal is strong!



- Matched filter impulse response is a "flipped in place" version of signal

$$h(T-u) = cs(u)$$

or

$$h(u) = cs(T-u)$$

Matched filter for pulses

- Integrate and dump corresponds to the matched filter with ideal rectangular pulses.

Summary

- When the input is signal plus WGN, then the filter that maximizes the SNR is the matched filter
- The filter shape is the flipped one of the signal in the time-domain
- In the frequency domain, the same magnitude with the signal.

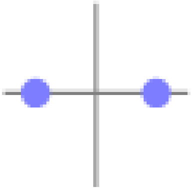
3. Modulation Schemes

- A channel with lowpass frequency characteristics is called *baseband*. Digital information is transmitted directly
  - Ex. Pulse Amplitude Modulation (PAM)
- A channel far removed from DC (like optical) is called a *bandpass* channel
- Transmission on a bandpass channel requires modulation of a *carrier*
  - Amplitude Shift Keying (ASK)
  - Phase Shift Keying (PSK)
  - Frequency Shift Keying (FSK
  - Quadrature Amplitude Modulation (QAM)
- Tolerant to amplitude distortion or not?

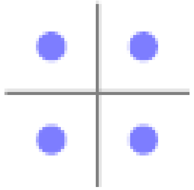
Digital Modulation Techniques

- **Frequency Shift Keying(FSK)**
  - Relatively simple to implement
  - Constant envelope signal is tolerant of nonlinearities
  - Ex. 300 bps voiceband modems, radio pagers, digital cellular
- **Phase Shift Keying(PSK, QPSK)**
  - Improved Bit Error Rate vs. Signal-to-Noise Ration performance
  - Constant envelope signal is tolerant of nonlinearities
  - Ex. 1200 bps modems, satellite modems, digital cellular
- **Amplitude Shift Keying(ASK, PAM, QAM)**
  - High degree of spectral efficiency (bps/Hz)
  - Requires highly linear analog front-end
  - Ex. 9600 bps modems, microwave radio, future digital television

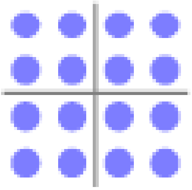
Data Encoding



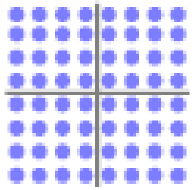
BPSK



QPSK



16QAM



64QAM

- Data subcarrier encoding
  - BPSK, QPSK, 16QAM, 64QAM
  - 1, 2, 4, 6 bits/subcarrier
- Error correction coding
  - 1/2, 2/3, or 3/4 rate convolutional code
  - Increased robustness
- Overall data rates:
  - 6, 9, 12, 18, 24, 36, 48, 54 Mbps
  - Lowest: 48 * 1 * 1/2 * 250K = 6 Mbps
  - Highest: 48 * 6 * 3/4 * 250K = 54 Mbps

Frequency Shift Keying

- **Data bits are mapped into different tones on the RF channel**
  - Minimum tone spacing = 1/T with noncoherent detection (T=symbol duration in sec)
  - Minimum tone spacing = 1/2T with coherent detection
    EX. Minimum Shift Keying (MSK)
- **Improved SNR performance can be achieved by using more than two tones to increase the symbol duration**
  - For R bps using M-ary FSK : T = $\log_2(M) / R$
    Ex. R=100 kbps : T=100 $\mu\sec$ for M=2, T=200 $\mu\sec$ for M=4
  - Receiver bandwidth (and noise power) is proportional to 1/T , thus SNR increases as T increases
  - 4-ary FSK has 3db SNR advantage over binary FSK
  - Channel bandwidth increases as M increases

BFSK orthogonal waveforms

- BFSK has the following waveforms:

$$s_1(t) = \sqrt{\frac{2\mathcal{E}_b}{T_s}} \cos(2\pi[f_c + f_d]t) \qquad 0 < t < T_s$$

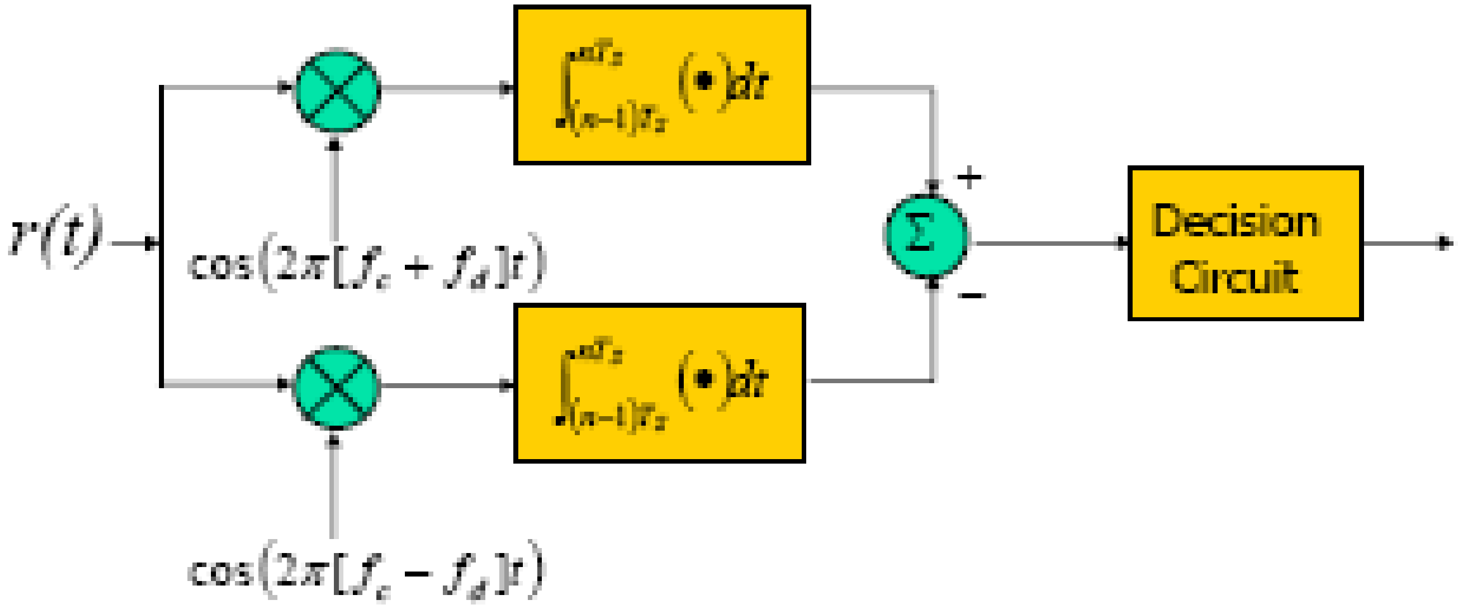
$$s_2(t) = \sqrt{\frac{2\mathcal{E}_b}{T_s}} \cos(2\pi[f_c - f_d]t) \qquad 0 < t < T_s$$

- If $f_d = n/4T_s$, for n a positive integer, these waveforms will be orthogonal

BFSK coherent detection

- Need frequency synchronization

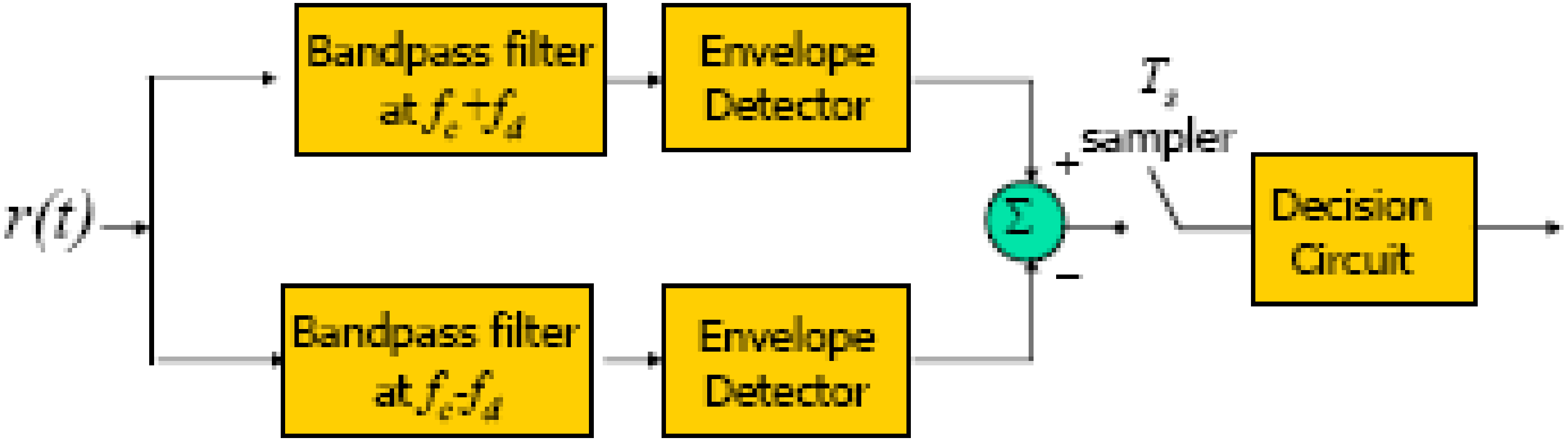
$$P_{CBFSK}(error) = Q\left(\sqrt{\frac{\mathcal{E}_b}{N_0}}\right)$$



BFSK non-coherent detection

- Use bandpass filters

$$P_{NBFSK}(error) = \frac{1}{2}\exp\left(-\frac{\mathcal{E}_b}{2N_0}\right)$$



$r(t)$

Bandpass filter at f_c+f_d

Envelope Detector

Bandpass filter at f_c-f_d

Envelope Detector

Σ

+

−

T_s sampler

Decision Circuit

Phase Shift Keying

- **Data bits are mapped into different phases of an RF carrier**
  - BPSK : each bit maps into 0 or 180 degrees
  - QPSK : each pair of bits maps into 45, 135, 225, or 315 degrees
  - For a given bit rate, QPSK requires half the bandwidth of BPSK yet achieves the same BER vs. SNR performance
  - Spectrum has sin(x)/x shape with nulls at 1/T Hz from carrier

- **PSK is constant envelope modulation if no baseband filtering is used**
  - Highly desirable feature for systems with nonlinear power amplifiers such as satellites (QPSK is most common modulation technique used in digital satellite links)
  - With baseband filtering, Offset-QPSK is often used to partially compensate for the envelope fluctuations

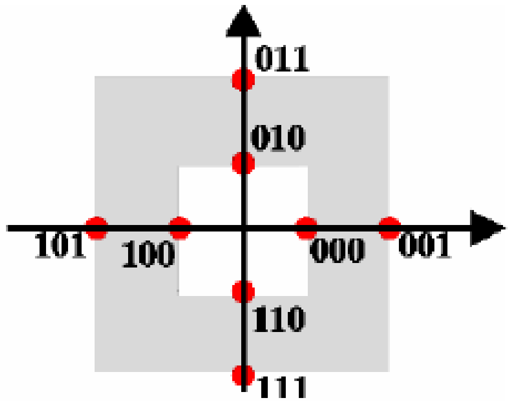
Amplitude Shift Keying

- **M-ary Pulse Amplitude Modulation (PAM)**
  - $\log_2 M$ bits are mapped into M amplitude levels
  - Commonly used in baseband twisted-pair transmission systems (ISDN and HDSL use 4-PAM or 2B1Q)
  - Redundant sideband wastes spectrum when PAM is modulated onto an RF carrier
  - Single Sideband(SSB) or Vestigial Sideband(VSB) are often used in analog AM systems to improve spectral efficiency(TV).

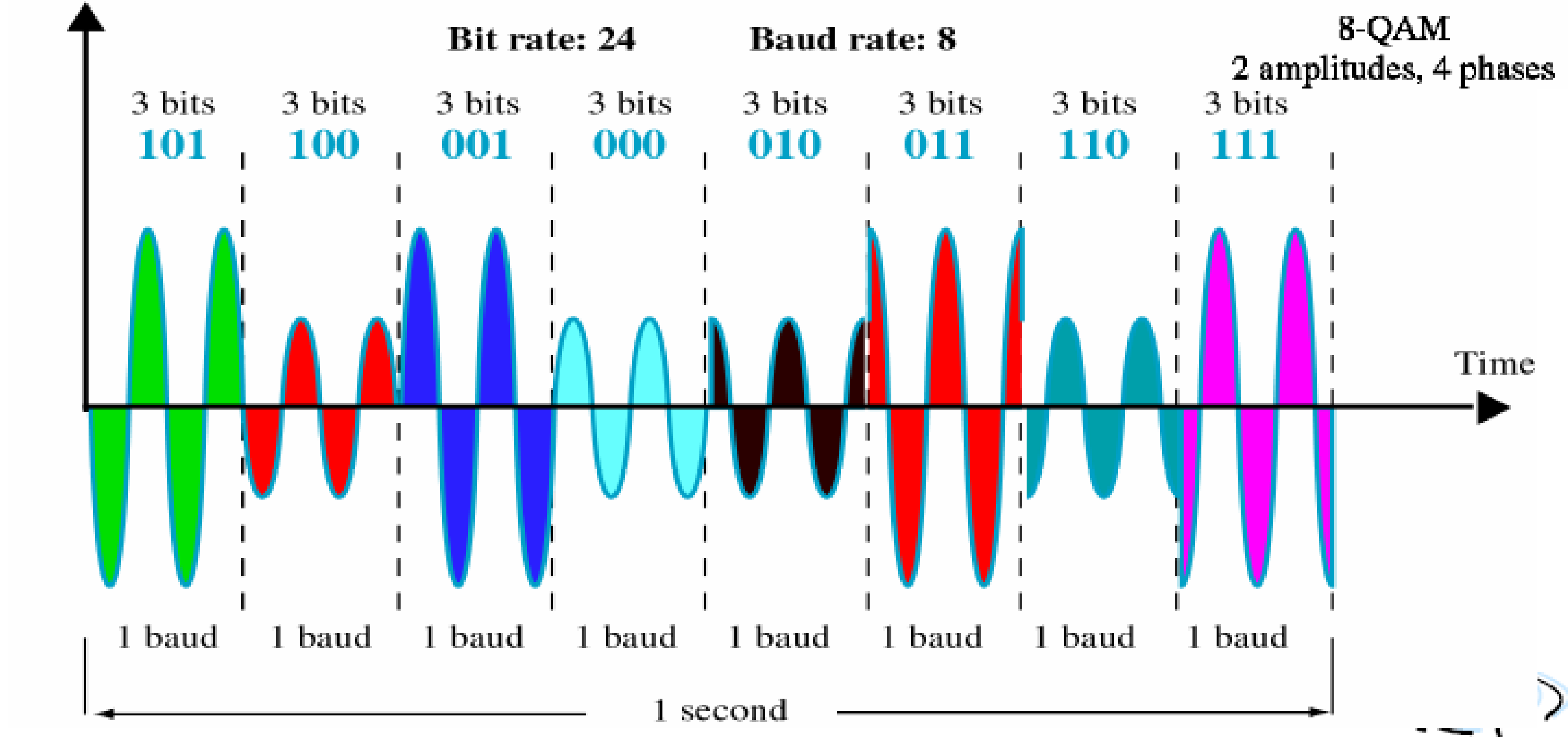
- **M-ary Quadrature Amplitude Modulation (QAM)**
  - $\log_2 M$ bits are mapped into sqrt(M) amplitude levels on each of two quadrature carriers
  - The most spectrally efficient modulation technique in use today
  - Requires large SNR to maintain low BER

구상진폭변조(QAM)

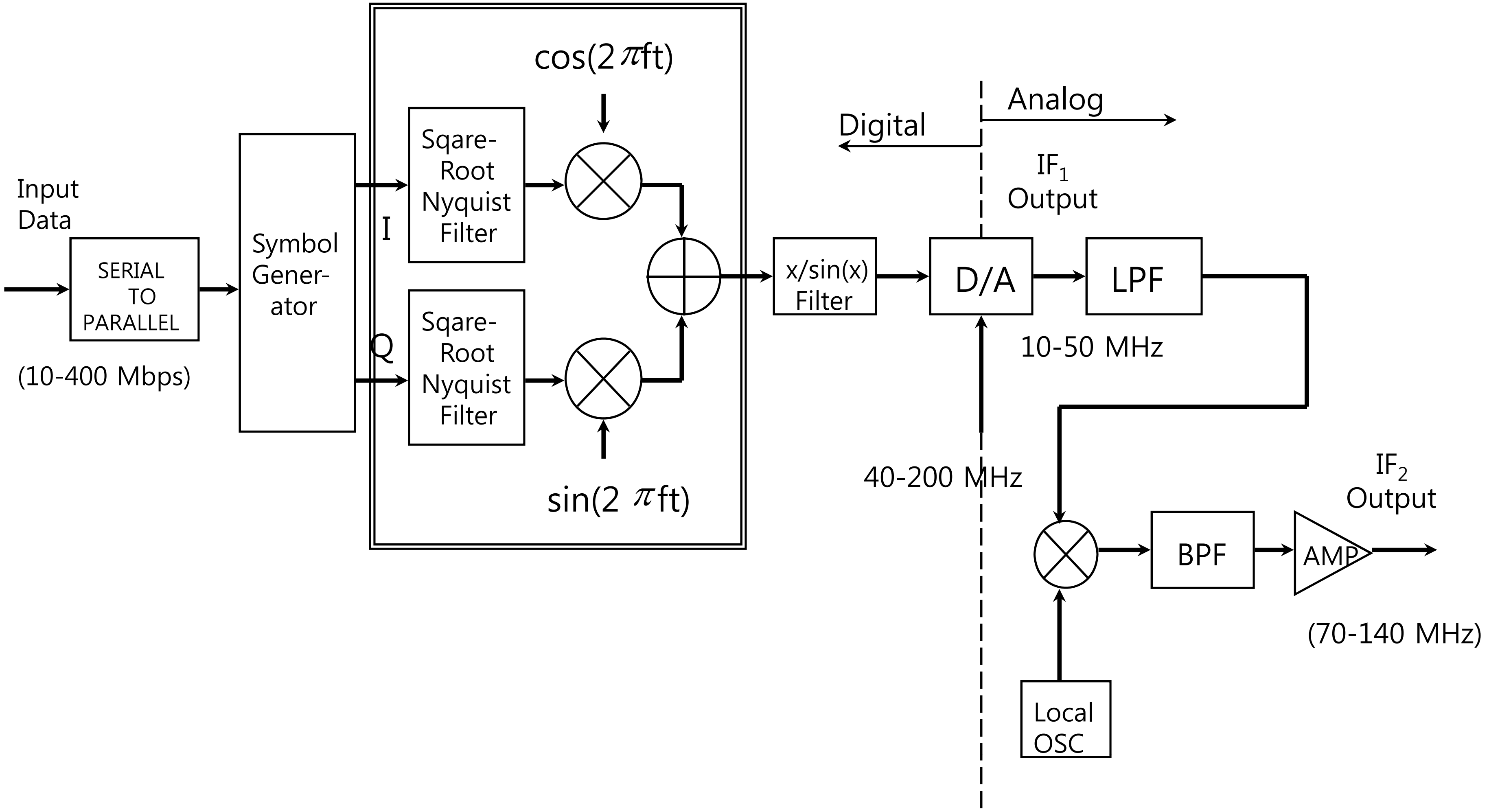
8-QAM의 시간영역 표시



8-QAM
2 amplitudes, 4 phases



All-Digital QAM Modulator Architecture

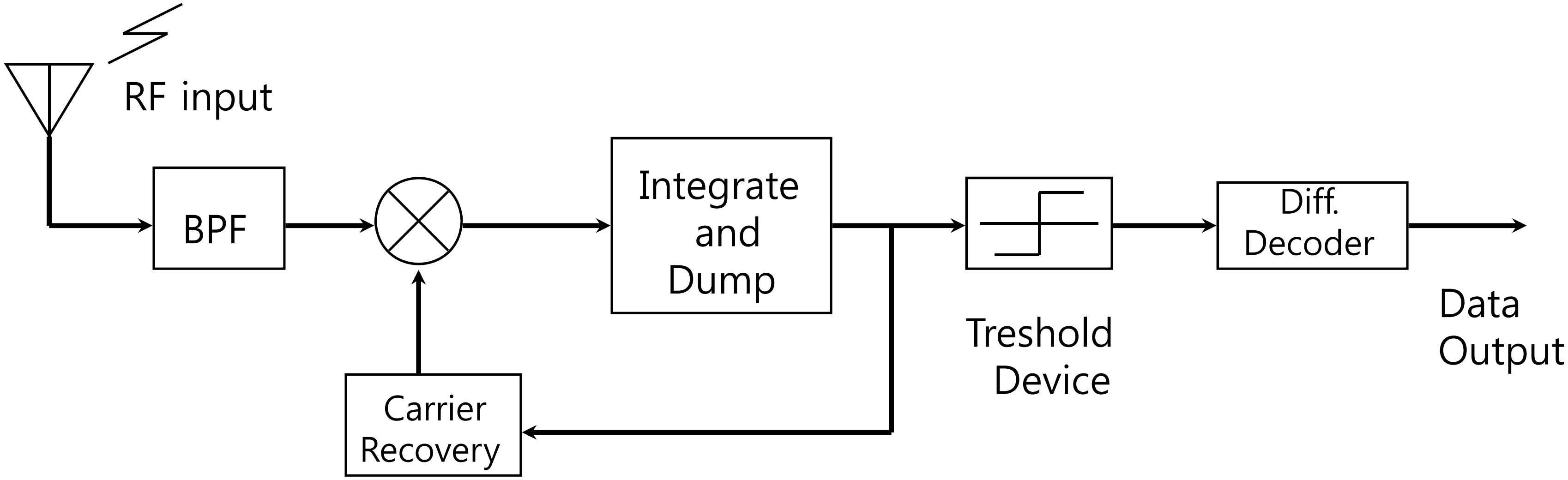


Coherent vs. Noncoherent Detection

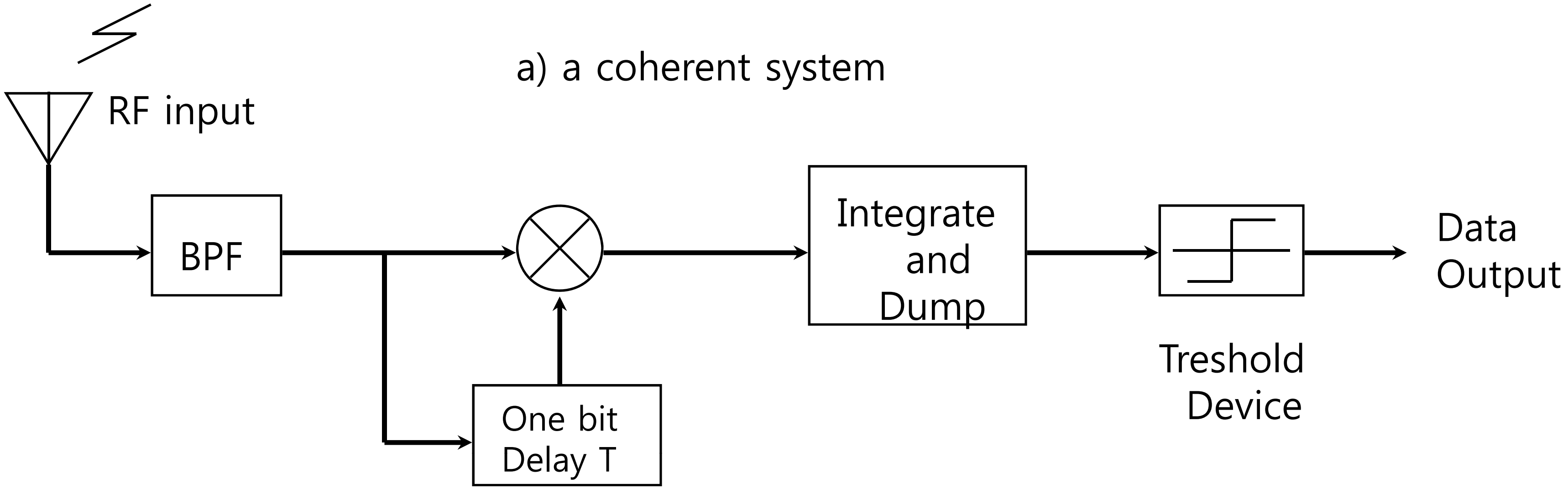
- **In a coherent receiver the local oscillator is synchronized in both frequency and phase to the incoming carrier**
  - The incoming carrier can then be exactly downconverted to baseband without any residual frequency or phase error
  - A phase-locked loop(PLL) is most commonly used for this task
  - Coherent detection is mandatory in QAM systems and optional in PSK and FSK systems. The performance gain is typically 3dB

- **A free-running oscillator which closely approximates the incoming carrier frequency is used in a noncoherent receiver**
  - In an FSK system one can simply use bandpass filters to detect the power in the received signal at the appropriate frequencies
  - In a PSK system “different detection” can be used to detect phase *changes* from symbol to symbol

Coherent and Noncoherent System

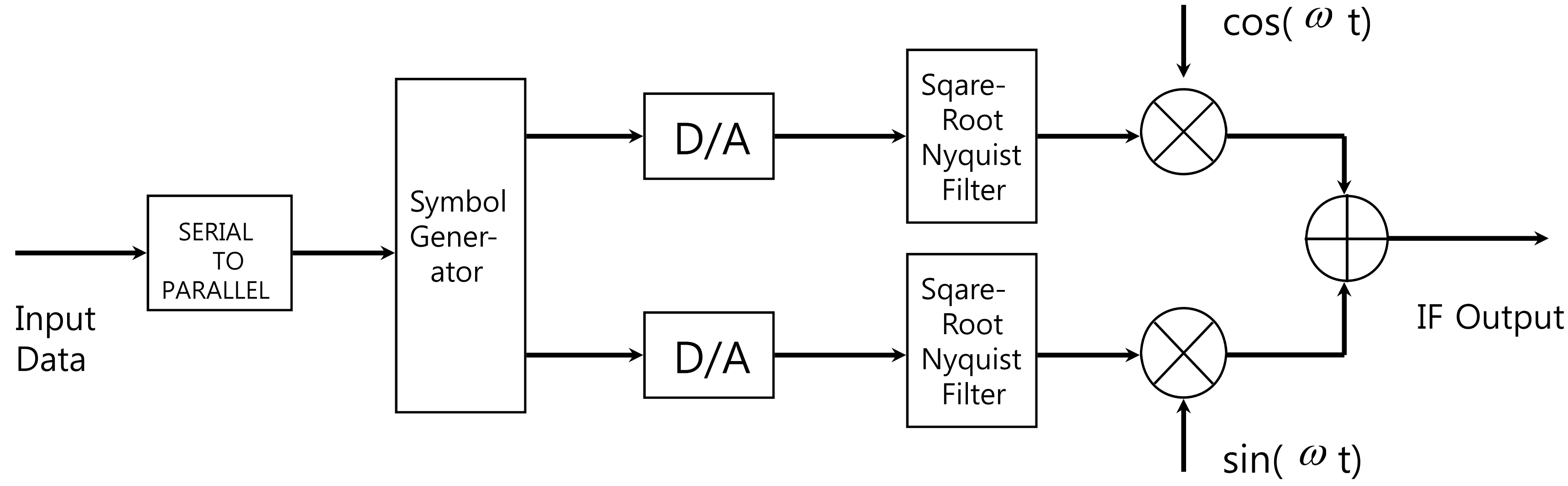


a) a coherent system



b) a noncoherent system

Basic Analog and Digital Architectures for Modulators



Input Data

SERIAL TO PARALLEL

Symbol Generator

D/A

D/A

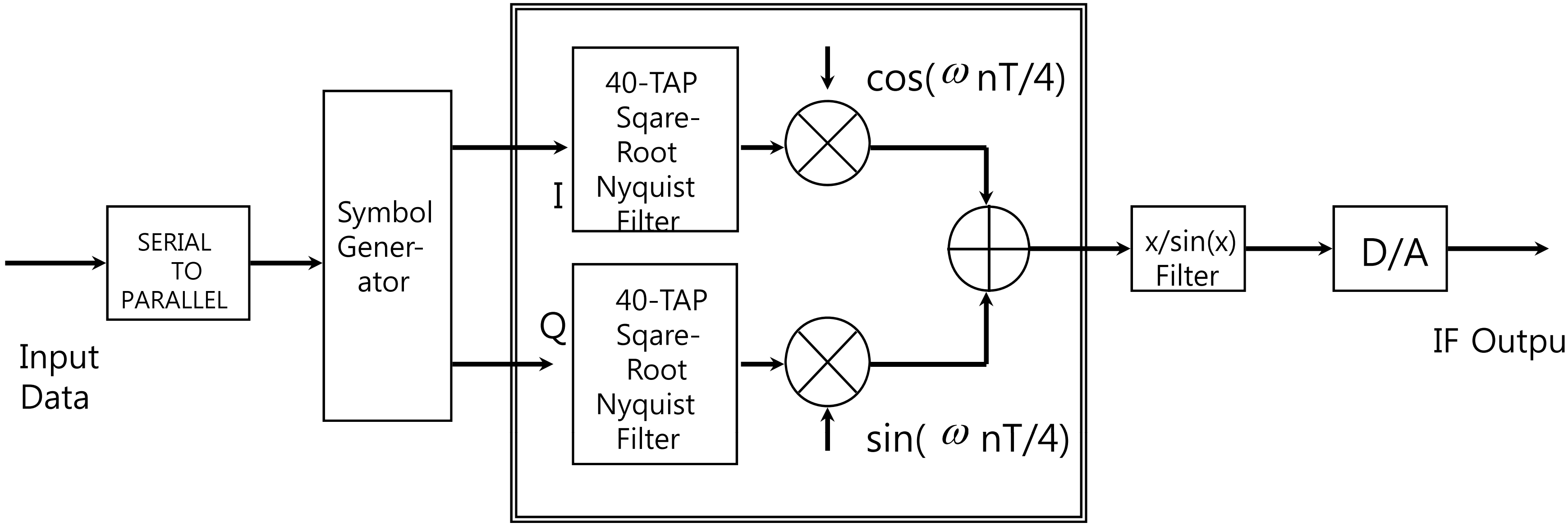
Sqare-Root Nyquist Filter

Sqare-Root Nyquist Filter

cos(ω t)

sin(ω t)

IF Output



Input Data

SERIAL TO PARALLEL

Symbol Generator

I

Q

40-TAP Sqare-Root Nyquist Filter

40-TAP Sqare-Root Nyquist Filter

cos(ω nT/4)

sin(ω nT/4)

x/sin(x) Filter

D/A

IF Output

4. Multiple Access Technologies

- FDM (Frequency Division Multiplexing)
  - Uses different frequency bands for each user
  - Needs band-pass filters
- TDM (Time Division Multiplexing)
  - Uses different time slots for each user
- CDMA (Code Division Multiplexing)
  - Uses Pseudo Noise-like code based correlation for selecting a specific user's data

FDM (Frequency Division Multiplexing)

- Most familiar with us because analog communications widely have used this scheme. AM, FM radios
- Can be implemented in analog (LC, crystal, ceramic, saw) filters, but can also be implemented with digital filters (down-sampled filtering needed). Digital filters are getting more cheap.
- Multiple channels can be more efficiently implemented with FFT (digital hardware)

Spread-Spectrum Techniques

- **The military pioneered the use of spread-spectrum techniques primarily for achieving anti-jam protection and low probability of detection**
  - The primary benefit for commercial application is to combat multipath distortion through the use of frequency diversity
  - The information is spread over a large bandwidth and can still be recovered even if part of the spectrum experiences a deep fade
  - Spread spectrum is also well-suited for unlicensed use since interference from other users appears like broadband noise rather than norrowband jammers (more graceful degradation)

- **Direct-Sequence and Frequency-Hopping are the two most popular spread-spectrum techniques**
  - The FCC has allocated 3 bands in the U.S. for unlicensed spread spectrum use (902-928 MHz, 2.4-2.4835 GHz, 5.8-5.9125 GHz)

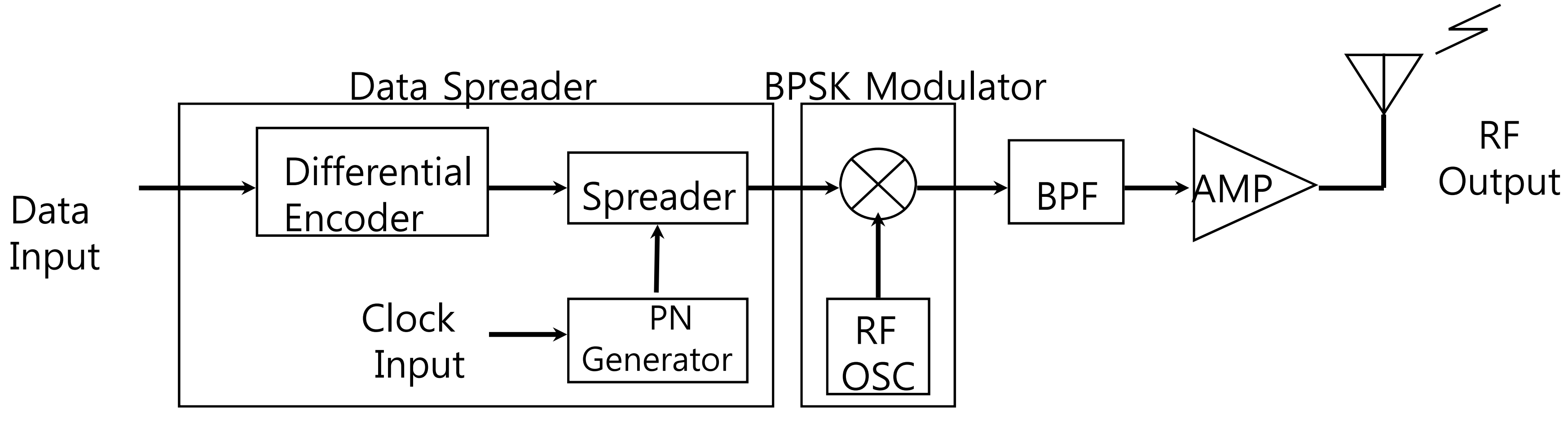
Direct-Sequence Spread Spectrum

- **The binary information stream is mixed with a much higher data rate binary pseudo-random noise (PN) sequence which is then modulated onto an RF carrier**
  - Binary-PSK (BPSK) or QPSK modulation is most commonly used for DS/SS systems
  - The ratio of the PN chip rate to the information bit rate is defined as the processing gain (PG) which is a measure of the interference rejection capabilities of the system

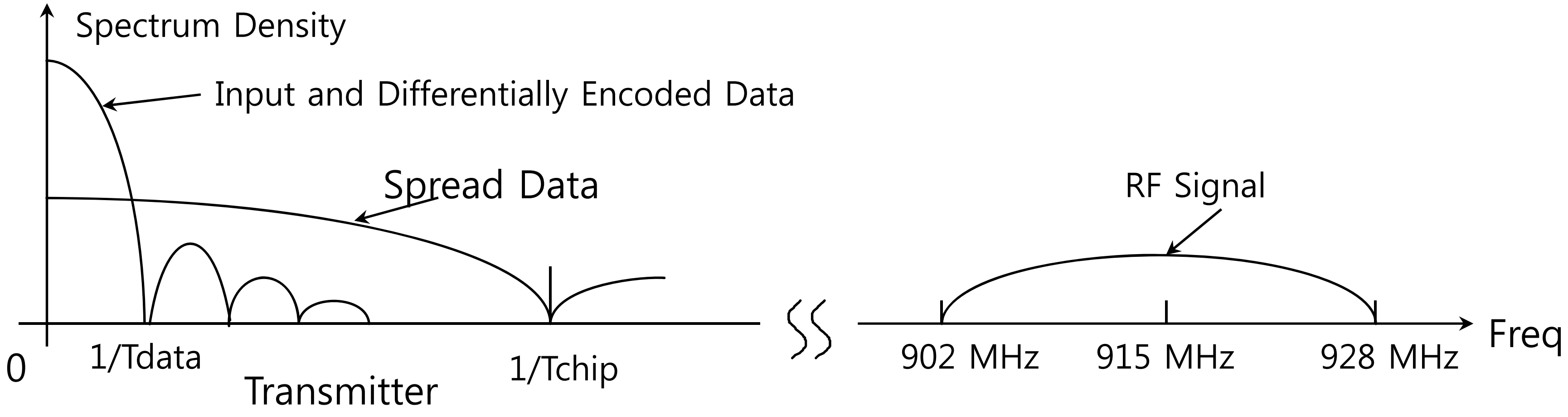
    Ex. Bit rate=10 kbps, Chip rate=10 Mchip/s --> PG=30 dB
  - Narrowband interferers are suppressed in the receiver by an amount equal to the processing gain

- **Each user is assigned a unique PN code wihch ideally is orthogonal to the other users PN codes, i.e., zero cross correlation and all users share the same bandwidth**

DS Spread Spectrum Transmitter


Data Input
Data Spreader
Differential Encoder
Spreader
Clock Input
PN Generator
BPSK Modulator
RF OSC
BPF
AMP
RF Output

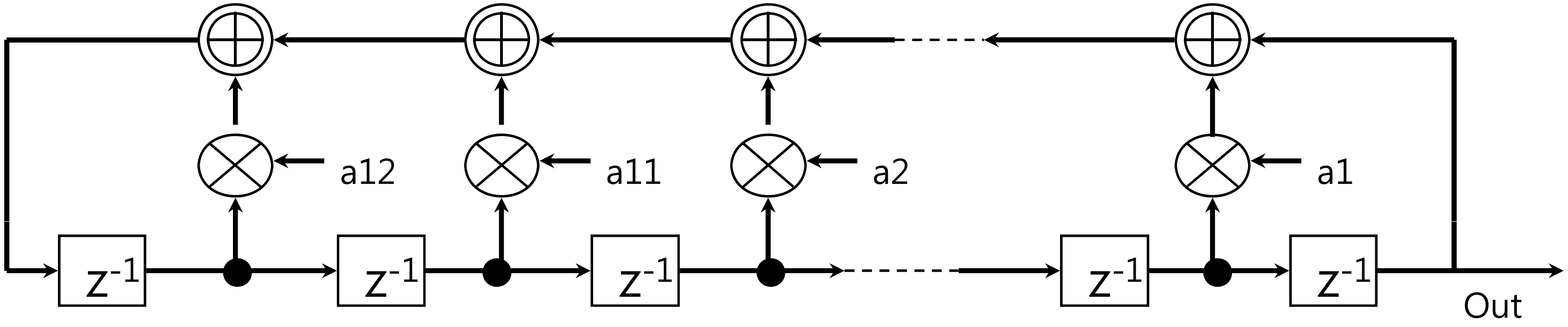
a) Transmitter Block Diagram


Spectrum Density
Input and Differentially Encoded Data
Spread Data
RF Signal
0
1/Tdata
Transmitter
1/Tchip
902 MHz
915 MHz
928 MHz
Freq

b) Spectrum Density

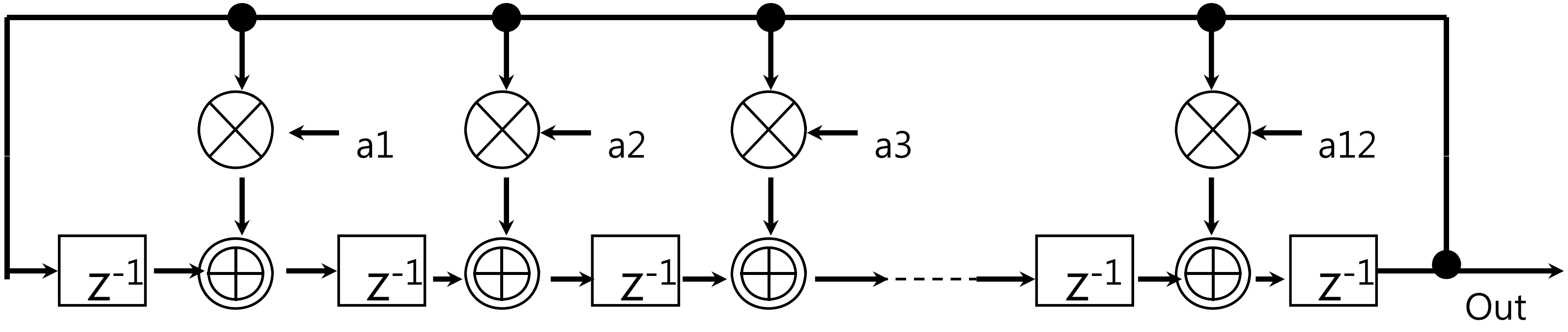
Circuit Diagram of a M-Sequence Generator (Degree 13)

$$f(x) = a_0 + a_1 x^1 + a_2 x^2 + \cdots + a_{11} x^{11} + a_{12} x^{12} + x^{13}$$



1) Linear Feedback Shift Register (LFSR) Implementation

$$f(x) = a_0 + a_1 x^1 + a_2 x^2 + \cdots + a_{11} x^{11} + a_{12} x^{12} + x^{13}$$

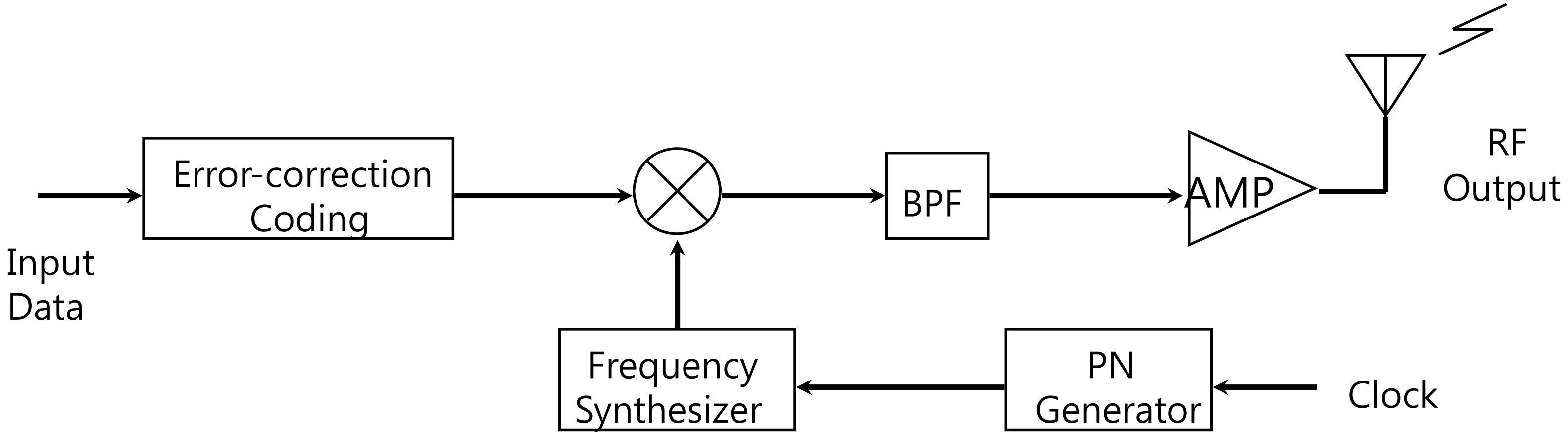


2) Modular Shift Register(MSR) Implementation

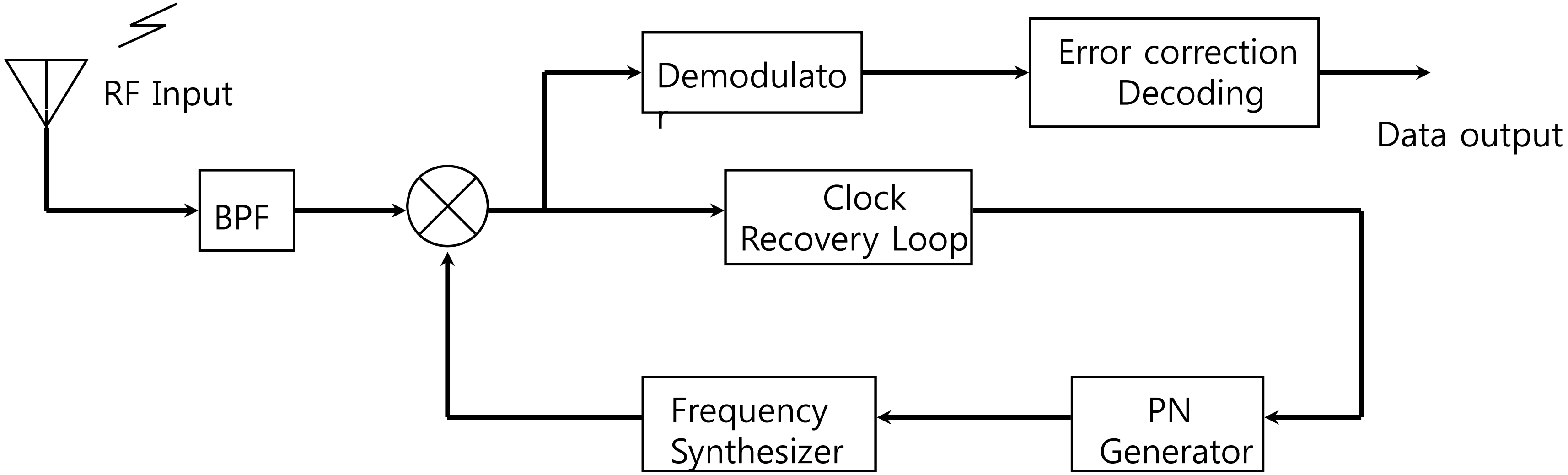
Frequency-Hopped Spread Spectrum

- **The carrier frequency of the modulated waveform is hopped over a wide bandwidth under the control under the control of a PN sequence**
  - Slow-Hopped FH/SS -- the hopping rate is slower than the information symbol rate (multiple symbols per hop)
  - Fast-Hopped FH/SS -- the hopping rate is faster than the informa-tion symbol rate (multiple hops per symbol)
  - the faster the hopping, the more robust is the system to multipath fading and jamming
- **FSK with noncoherent detection is the most commonly used modulation format of FH/SS**
  - Coherent demodulation is difficult since the phase of the carrier changes on every hop
  - Slow hopping with coherent detection within the hop is feasible

Frequency Hopping SS System


Input Data
Error-correction Coding
BPF
AMP
RF Output
Frequency Synthesizer
PN Generator
Clock

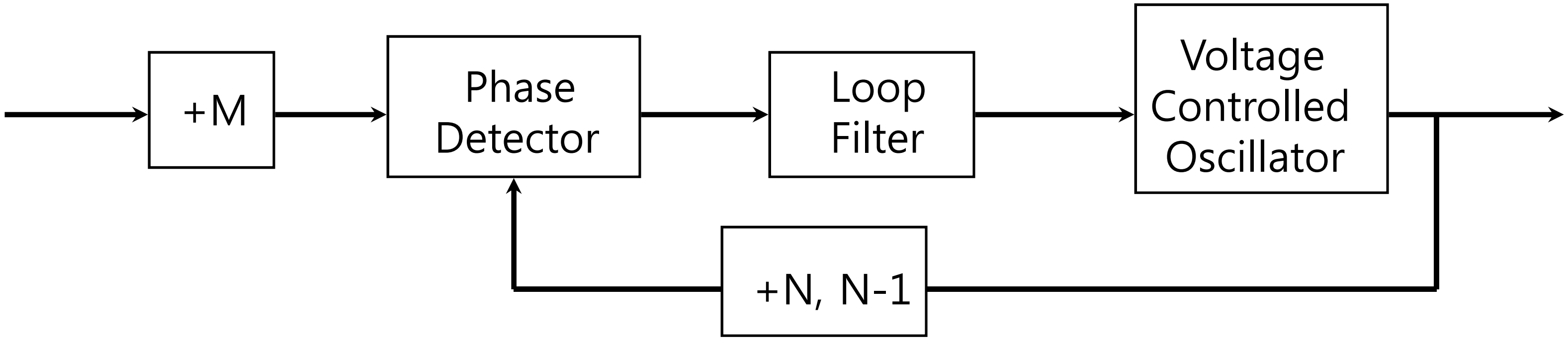
a) Transmitter


RF Input
BPF
Demodulator
Error correction Decoding
Data output
Clock Recovery Loop
Frequency Synthesizer
PN Generator

b) Receiver

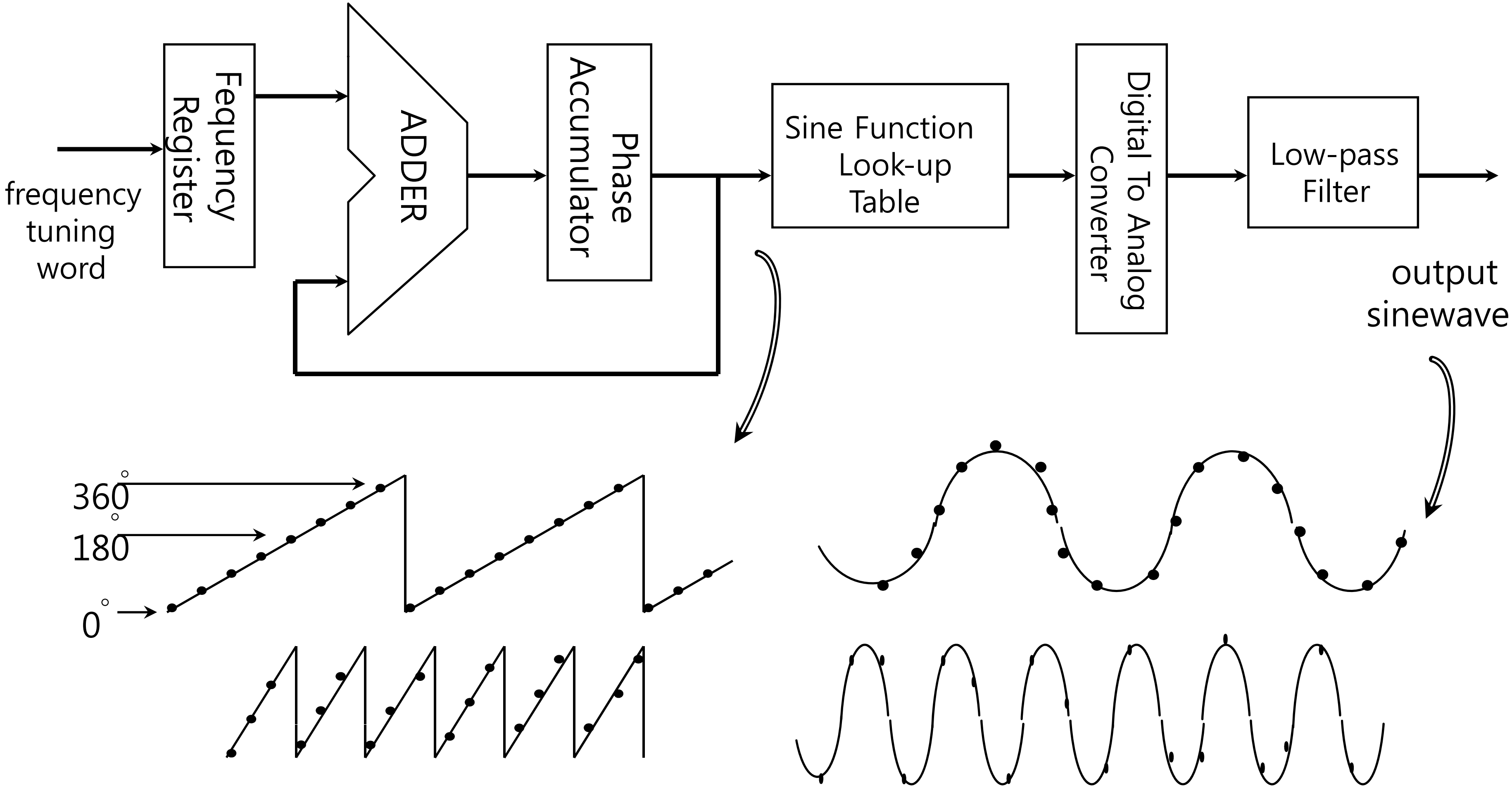
Frequency Synthesizers for FH/SS systems

- **PLL-based frequency synthesizers are most often used**
  - Low complexity, good spectral purity, and wide tuning range
  - Poor settling time limits their use to slow-hopped systems



- **Direct digital frequency synthesizers (DDFS's) are beginning to gain popularity**
  - Moderate complexity, good spectral purity, moderate tuning range
  - Fast settling time can accommodate fast-hopped systems

Basic direct Digital Frequency Synthesis Technique



frequency tuning word

Fequency Register

ADDER

Phase Accumulator

Sine Function Look-up Table

Digital To Analog Converter

Low-pass Filter

output sinewave

Output of periodically Overflowing Phase Accumulator Indexes Look-up Table

Sine Wave Samples are Output Every Clock Cycle With Analog Interpolation provided By DAC and Low pass Filter

Diversity Techniques for Spread Spectrum

- **The diversity order of a system is defined as the number of different (and independent) replicas of the transmitted data bits that are processed in the receiver**
- **Diversity can be provided in many domains**
  - Space diversity through the use of multiple receive antennas
  - Frequency diversity by sending the signal at multiple frequencies
  - Time diversity by sending the data at different times

- **In a severely fading channel diversity techniques are crucial for achieving reliable transmission**
  - Ex. SNR required for BPSK modulation at a BER of 10^{-5}

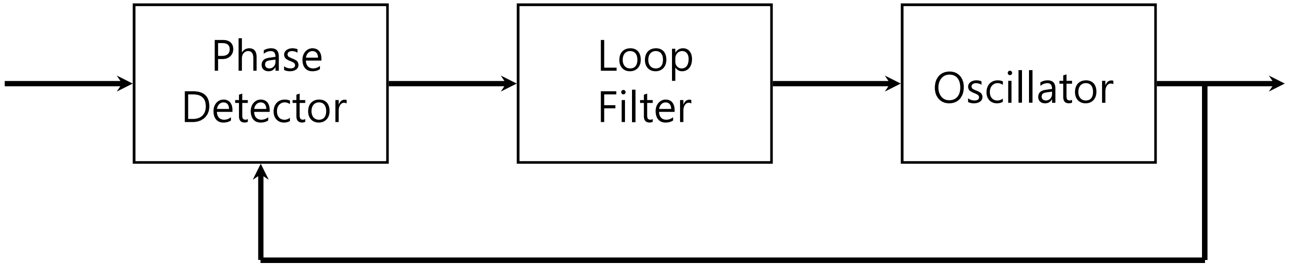
    Additive white Gaussian noise channel : 10 dB
    Rayleigh fading channel with no diversity : 45 dB
    Rayleigh fading channel with 4th-order diversity : 15 dB

5. Synchronization

- Carrier synchronization for heterodyne/direct conversion receivers
- PLL
- Symbol synchronization

Phase Locked Loops

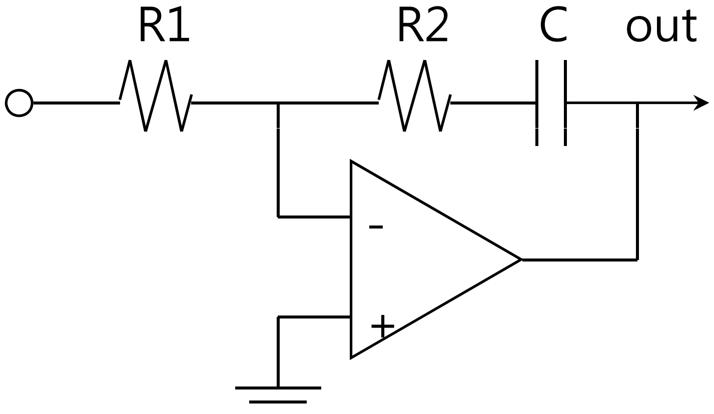
- **Phase locked loops (PLL's) are fundamental building blocks in communication systems**
  - Used for frequency synthesis, carrier recovery and clock recovery



  - Feedback loop tries to force the frequency and phase of output oscillator to match frequency and phase of input signal
  - Phase detector is most critical element in determining overal l performance of PLL
  - Both analog an digital implementations are commonly used

Circuit Diagram of a digital and an analog Loop Filter

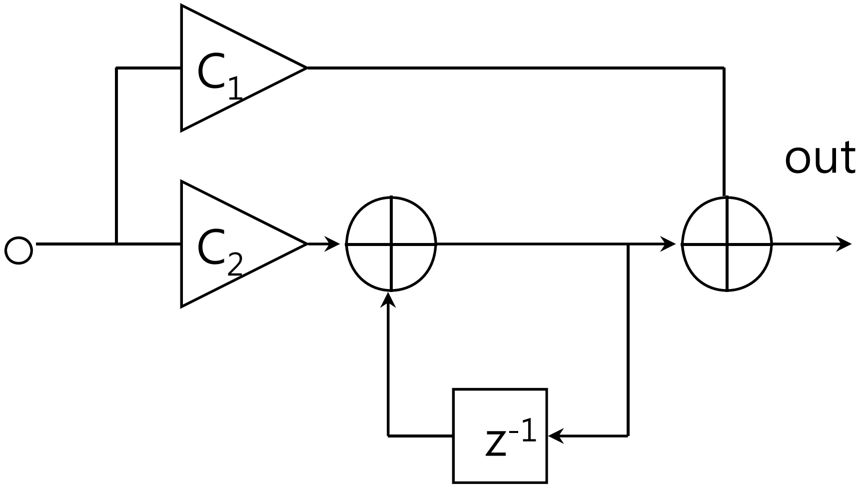
1st order Analog Loop Filter



$\tau_1 = R_1 C \qquad \tau_2 = R_2 C$

$$H(s) = \frac{1 + s\tau_2}{s\tau_1}$$

1st order Digital Loop filter



$$H(z) = \frac{(C_1 + C_2) - C_1 z^{-1}}{1 - z^{-1}}$$

Carrier Recovery Techniques

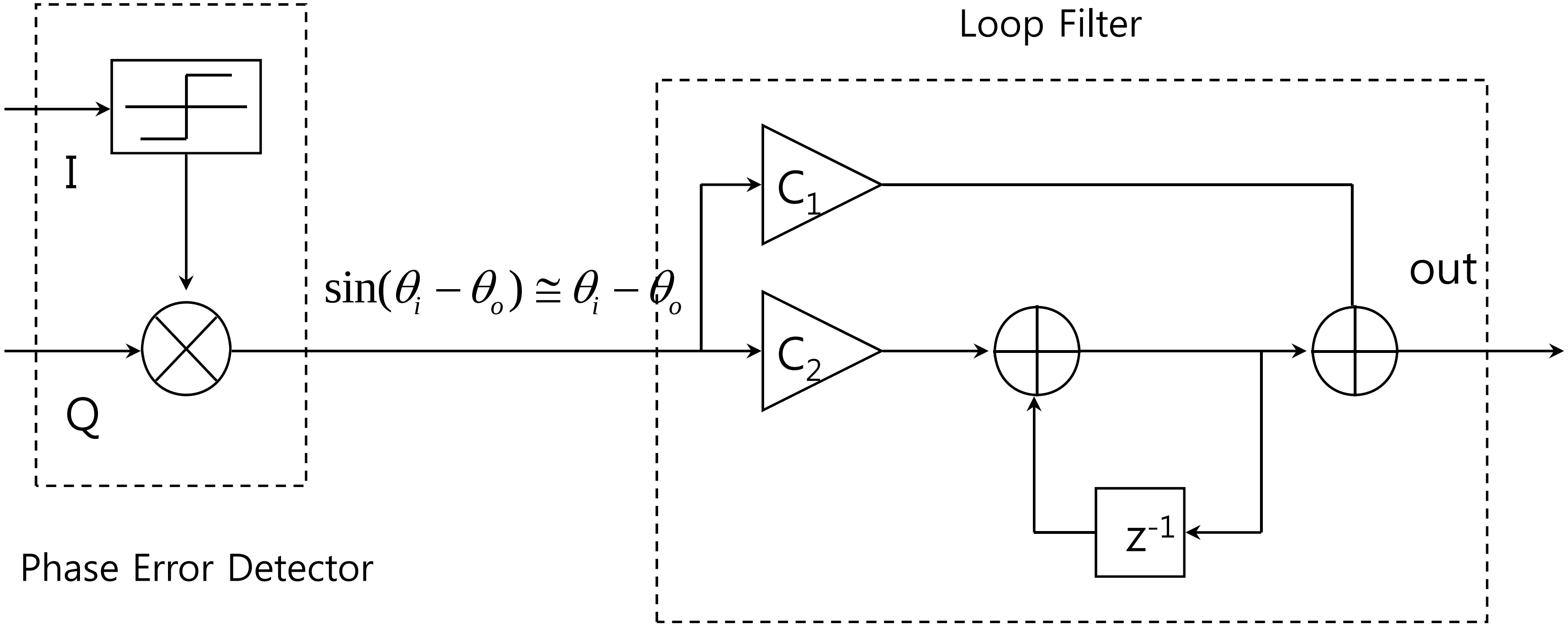
- **M-th Power Spectral Line Techniques**
  - By passing a BPSK signal through a square law device, a spectral line is created at twice the carrier frequency which can be input to a phase-locked loop (PLL) to generate the recovered carrier :

$$[\pm A\cos(\omega t)]^2 = \frac{A^2}{2}[1+\cos(2\omega t)]$$

  - Similarly, by passing a QPSK signal through a 4-th law device, a spectral line is created at 4 times the carrier frequency

- **Costas Loop**
  - The baseband inphase and quadrature signals are cross multiplied together to form the phase error signal for the PLL
  - Achieves identical performance to 4-th law technique but is gener-ally easier to implement and is therefore a very popular technique in BPSK, QPSK and QAM modems

Circuit Diagram of Phase Detector and a Loop Filter



$$H(z) = \frac{(C_1 + C_2) - C_1 z^{-1}}{1 - z^{-1}}$$

Clock Recovery Tehniques

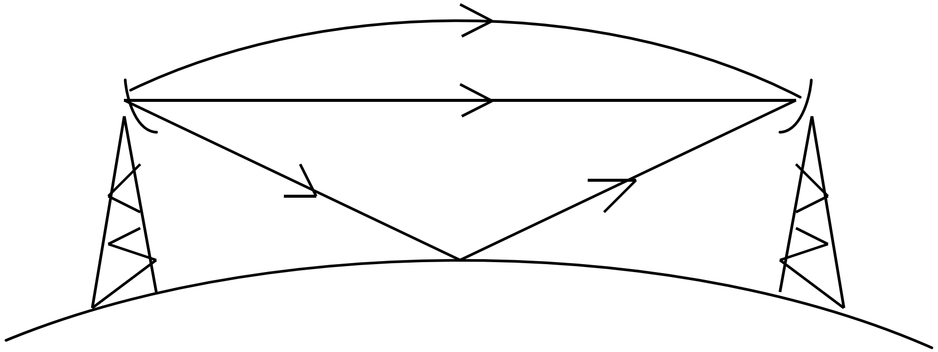
- **Spectral Line Techniques**
  - By passing the baseband inphase or quadrature signal through a square law device, a spectral line is created at the baud frequency which can be input to a PLL to derive the symbol clock
- **Early-Late Gate Synchronizers**
  - By subtracting an early sample of the baseband signal from a late sample of the baseband signal, a timing phase error signal can be derived for the PLL

- **Minimum Mean Squared Error Timing Recovery**
  - By multipying the decision error of the detected symbol by the slope of the baseband waveform, a timing phase error signal can be derived for the PLL

6. Adaptive Equalization

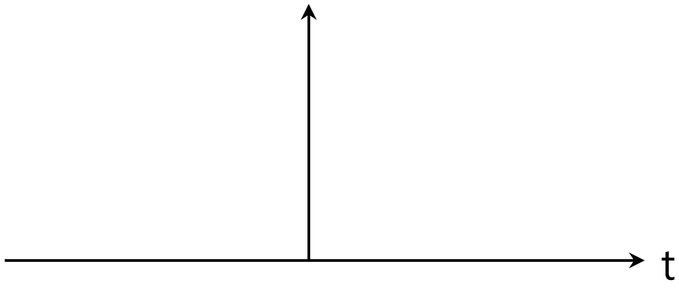
- **Multipath propagation is the most common distortion encountered in a radio channel**
  - Caused by reflections off of large objects or by propagation delay differences through varying layers in the atmosphere
  - Manifests itself as ghosts or echos in an analog transmission link and intersymbol interference (ISI) in a digital transmission link
  - In the frequency domain, multipath channels have notches of varying width and depth appearing at random frequency locations
  - For narrowband (low data rate) signals the multipath distortion results in occasional loss of signal when the notch frequency coin-cides with the carrier frequency
  - For wideband signals the notches are frequently encountered in the receivered signal and adaptive equalization must be used to achieve realiable transmission

MULTI-PATH PROPAGATION

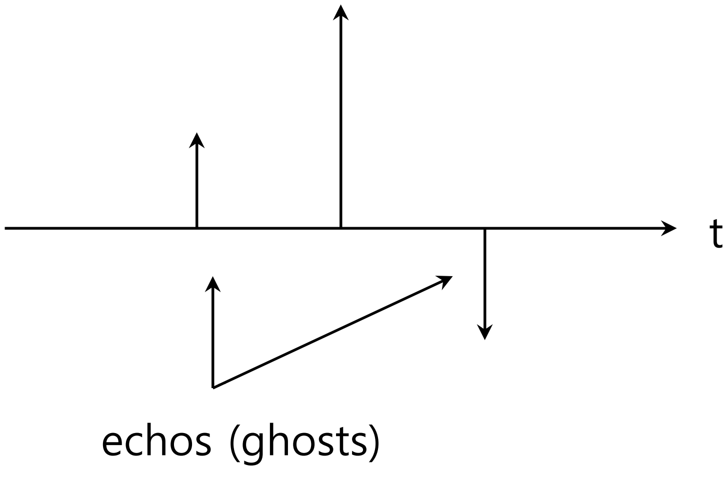


CHANNEL IMPULSE RESPONSE

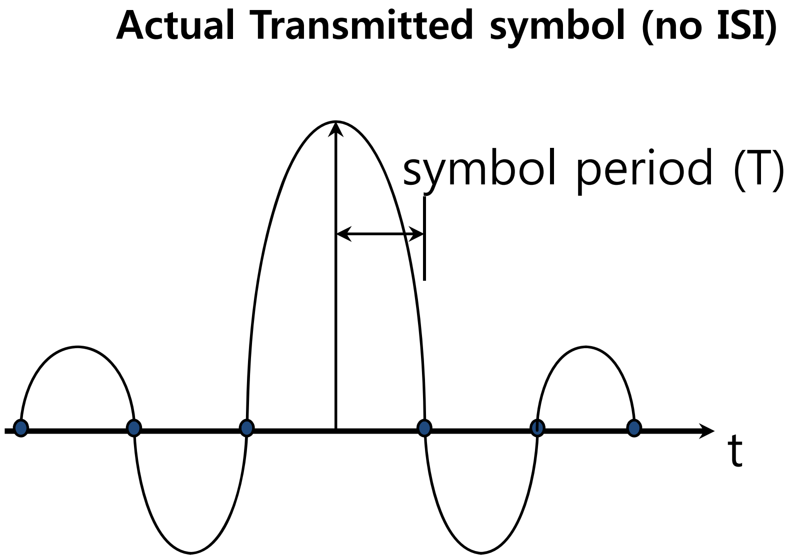
Ideal transmitted symbol :

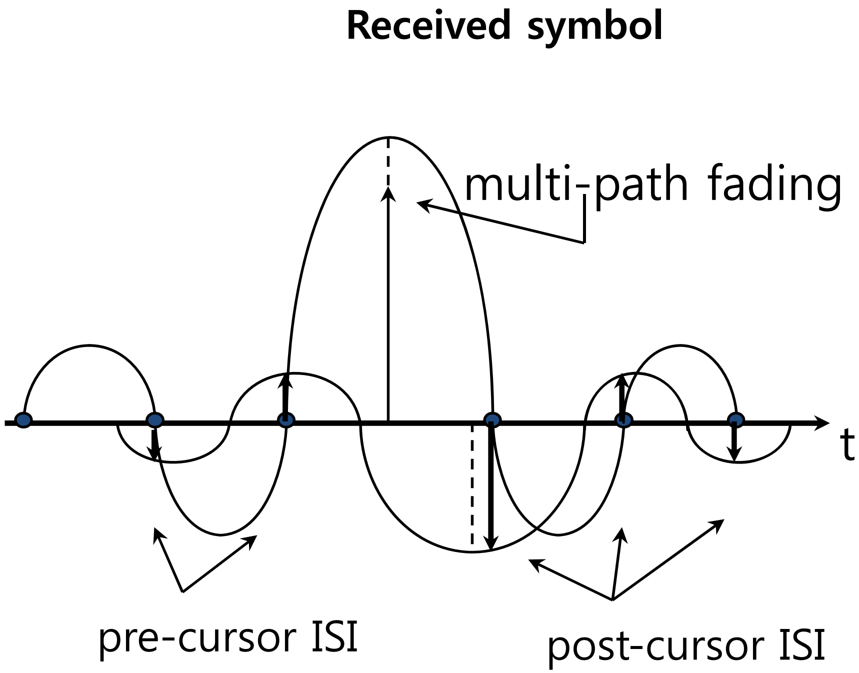


Received symbol :



Inter-symbol Interference (ISI)





- Pre-cursor ISI can be canceled by Feed-Forward Linear transversal Equalizer (FFE)
- Post-cursor ISI can be canceled by Decision-Feedback Equalizer (DFE)

Linear Transversal Equalizers

- **Most commonly used form of adaptive equalization due to straightforward and easily implementable structure**
  - Tapped delay line structure requiring 1 multiply/accumulate opera-tion per real filter tap or 4 multiply/accumulate operation per complex filter tap as required in quadrature demodulators
  - The delay between the taps is typically chosen to equal the symbol rate T or half the symbol rate T/2
  - The number of taps in the equalizer should be large enough to span the total multipath delay spread in the channel

- **Compensates for channel distortion by providing a transfer function which is the inverse of the channel frequency response**
  - Primary drawback is potential noise enhancement when large gain is introduced to compensate for a deep channel null

Decision-Feedback Equalizers

- **Provides improved performance over LTE by incorporating a feedback section whose imputs are the symbol decisions**
- **The DFE is among the most powerful structures available for channel equalization in digital transmission systems**
  - Can compensate for deep channel nulls without incurring the noise enhancement penalty of the LTE
  - The more severe the channel distortion, the more significant is the performance gain of a DFE over an LTE

- **Similar in hardware complexity to LTE except that high-speed implementations are difficult due to its recursive structure**
  - Pipelining cannot be incorporated into the feedback loop
  - The LTE can be pipelined for high-speed operation since it is non-recursive

Coefficient Updating Algorithms

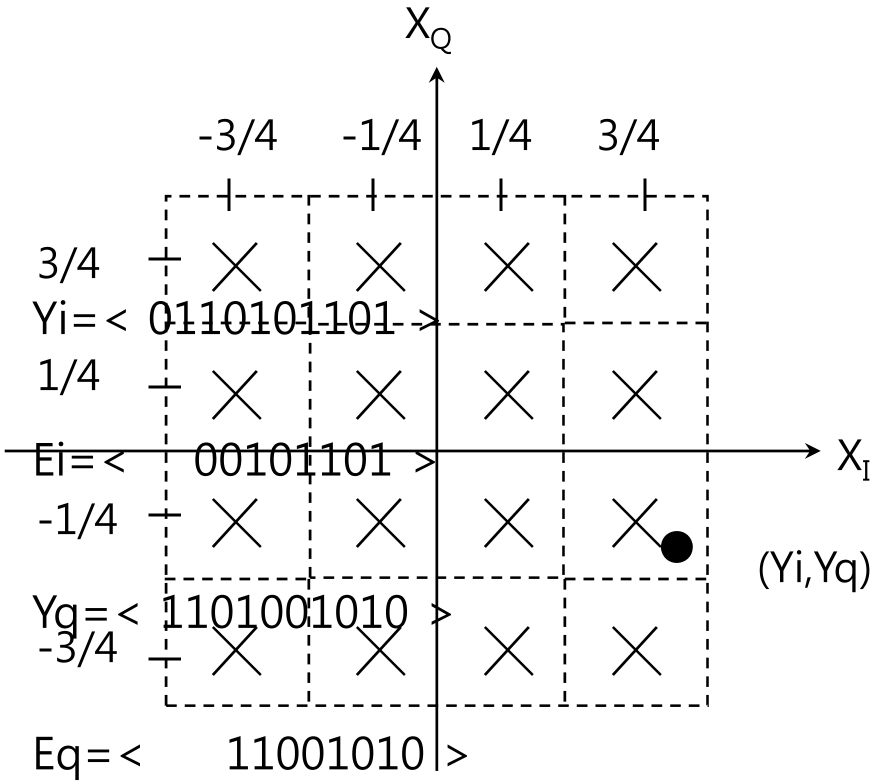
- **The Least Mean Square (LMS) algorithm is the oldest and most popular technique fo radaptive equalizer coefficient updating**
  - invented by Widrow at Stanford in the 1960's
  - Very simple to implement requiring only 1 multiply/accumulate operation per coefficient per iteration

$$c_k(n+1) = c_k(n) + \mu e(n)x(n-k)$$

  - Simplified multiplierless versions are also commonly used
  - Extremely robust and stable and relatively insensitive to finite wordlenth effects

- **More sophisticated and complex algorithms have been developed to achieve faster convergence but have not gained much popula-rity**
  - Recursive Least Squares (RLS)

An Example -- 16 QAM Constellation



Yi=< 0110101101 >

Ei=< 00101101 >

Yq=< 1101001010 >

Eq=< 11001010 >

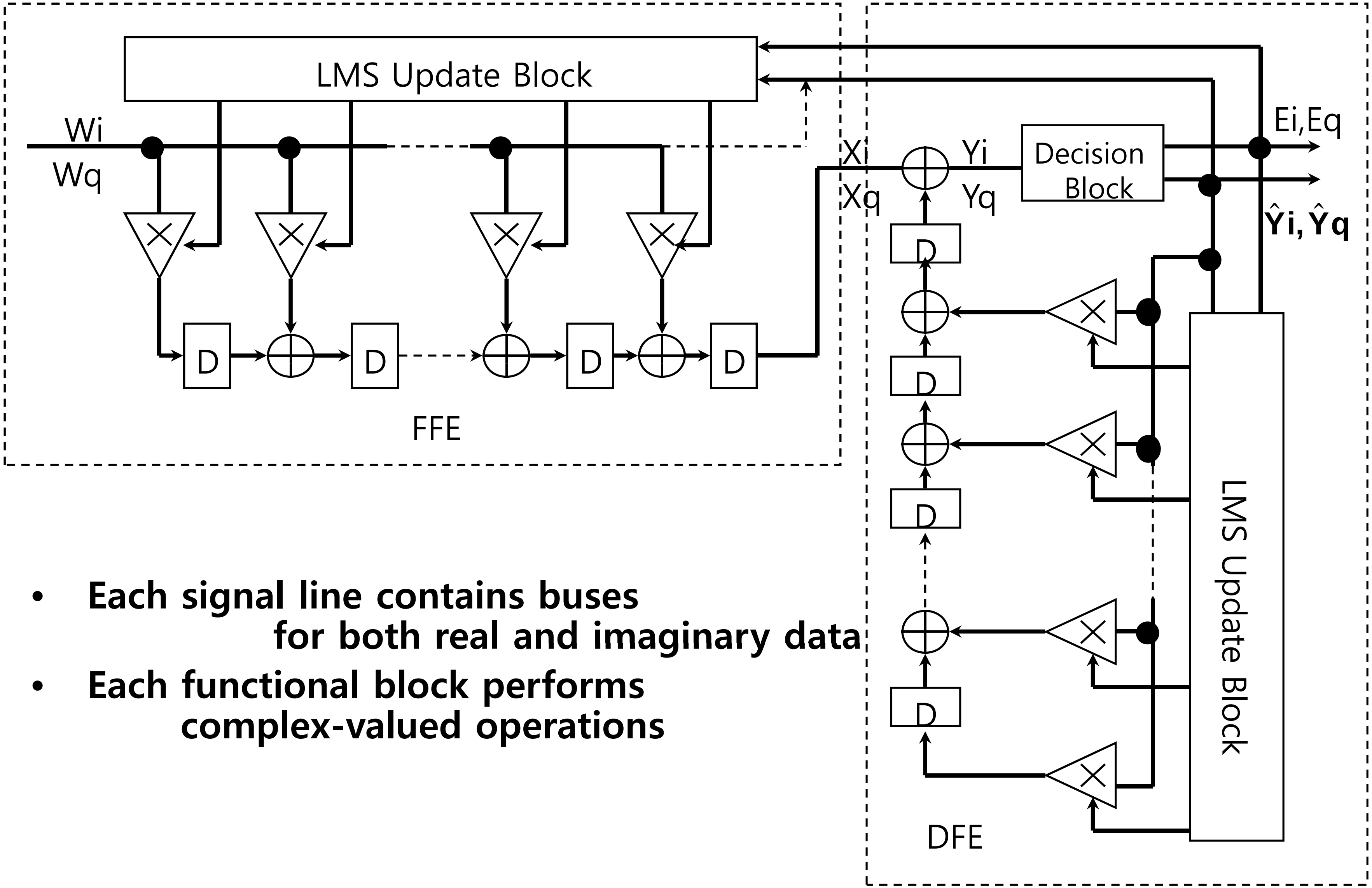
Decision/Error
Extractions :

$\hat{Y}$**i** =< **011** >

$\hat{Y}$**q = <111>**

- **all vectors have 2's complement representation**
- **Saturation is required to mininize overflow errors**

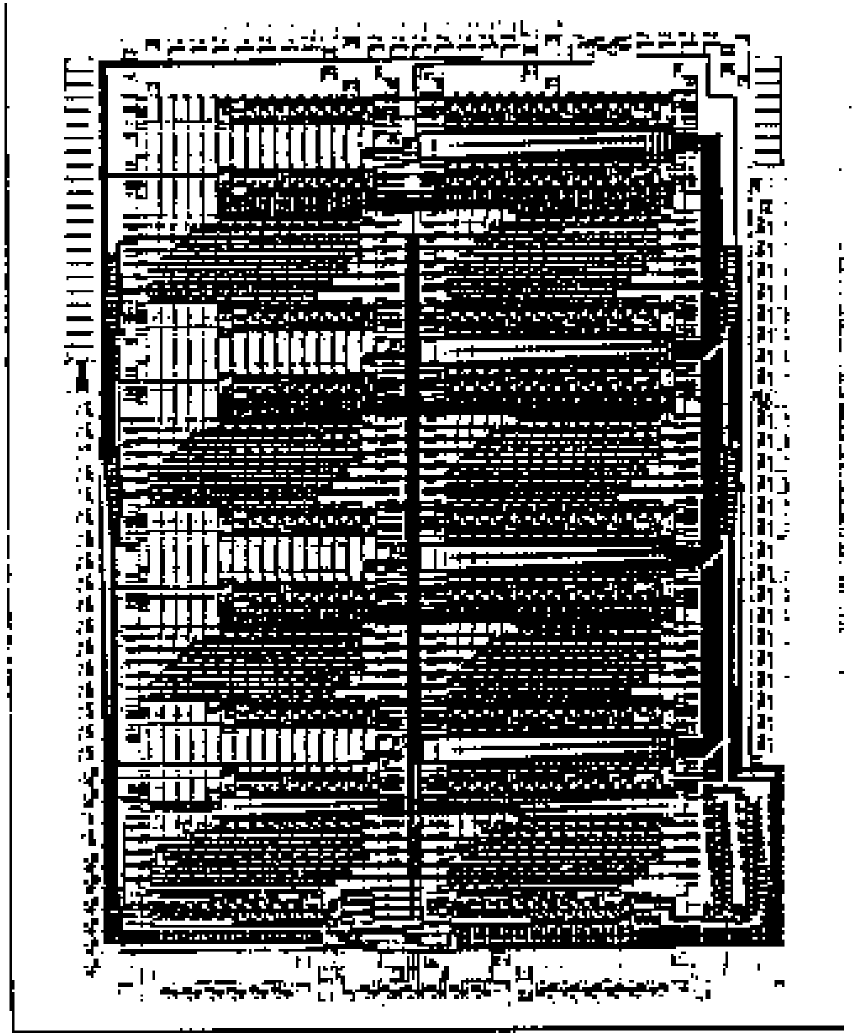
Simplified Block Diagram of the Decision-Feedback Equalizer System



- **Each signal line contains buses for both real and imaginary data**
- **Each functional block performs complex-valued operations**

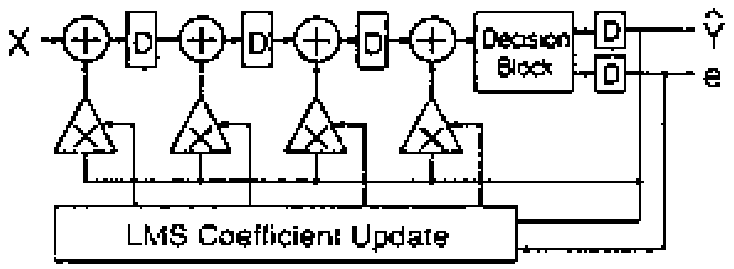
QAM DECISION-FEEDBACK EQUALIZER

UCLA Integrated Circuits
and Systems Laboratory



Chip Features

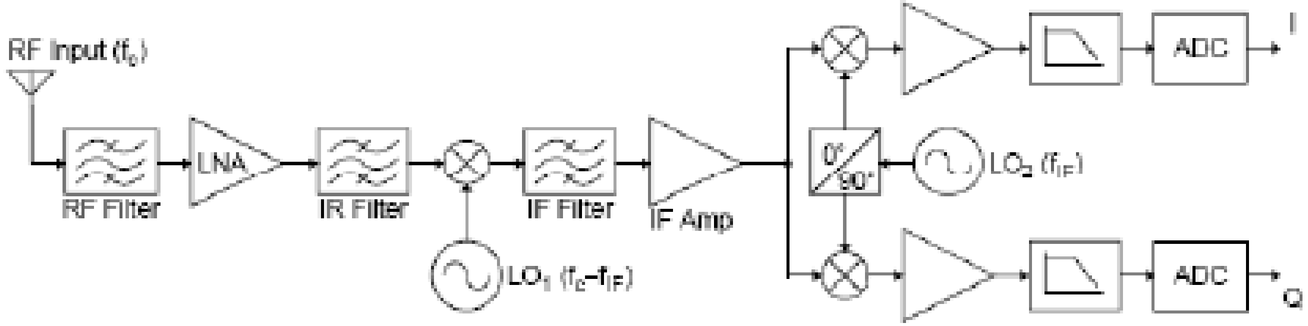
- 60 MHz Maximum Clock Rate
- 60 Mbaud Maximum Symbol Rate
- QPSK, 16-, 64-, and 256-QAM Signal Formats
- 4 Complex-Valued Filter Taps
- On-Chip LMS Coefficient Updating Circuit
- Cascadable With No Speed Degradation
- Adaptively Biased Pseudo-NMOS Logic Technique
- 1.5 W Power Dissipation @ 60 MHz
- 770 mW @ 35 MHz (Power-Saving Mode)
- 7.0 mm x 4.9 mm Die Area
- 70,000 Transistors
- 1.2-micron CMOS Technology (MOSIS)

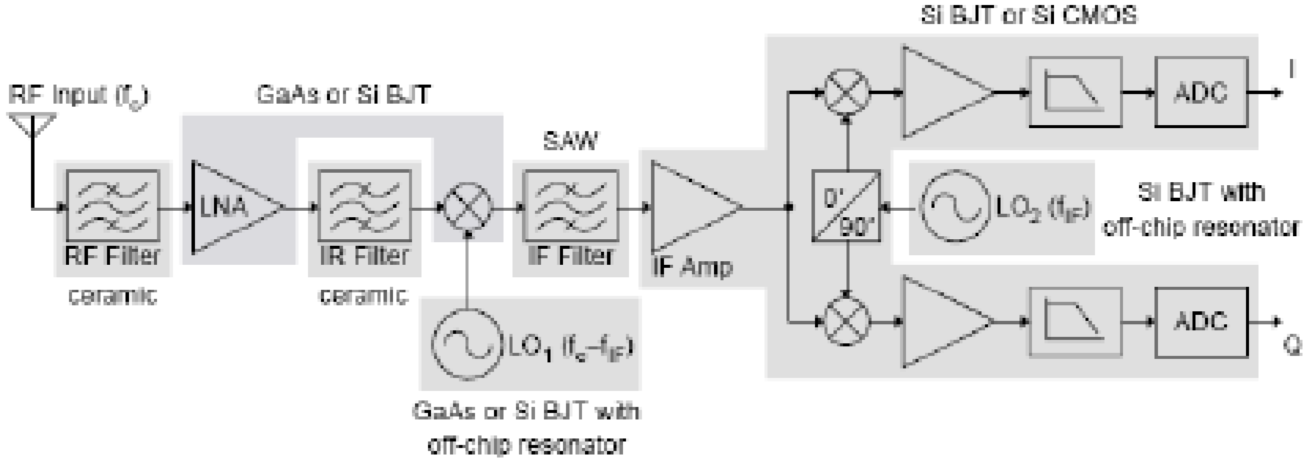


7. Communication System Architecture

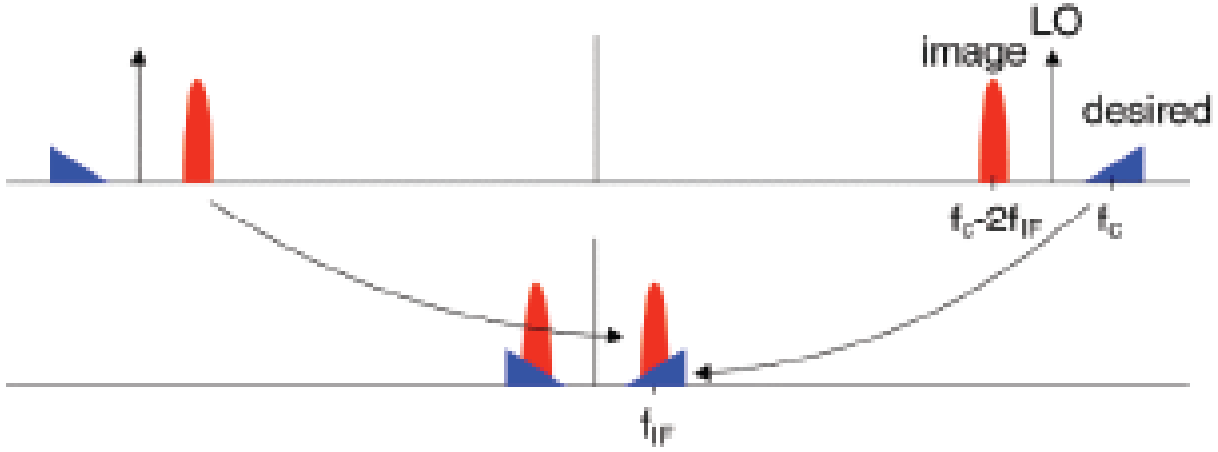
- Heterodyne architecture
  - Use multi-stage for baseband to RF or RF to baseband
  - Needs complex RF circuits but low frequency ADC/DAC needed
  - Image problem
- Direct conversion architecture
  - RF<->baseband
  - High frequency ADC/DAC needed
  - Good for single chip implementation
- Low-IF architecture

Heterodyne Architecture





The Image Problem



- The image signal can be much stronger than the desired signal and must be attenuated before downconversion.
- Trade-offs:
  - High IF relaxes RF and IR requirements
  - Low IF relaxes IF requirements

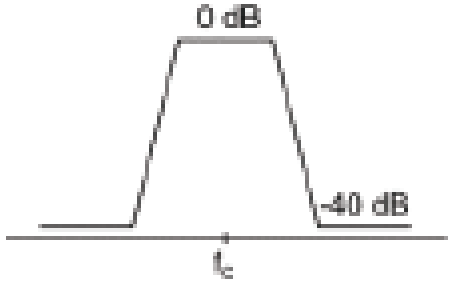

Typical BPF Attenuation

Image-Reject Mixing

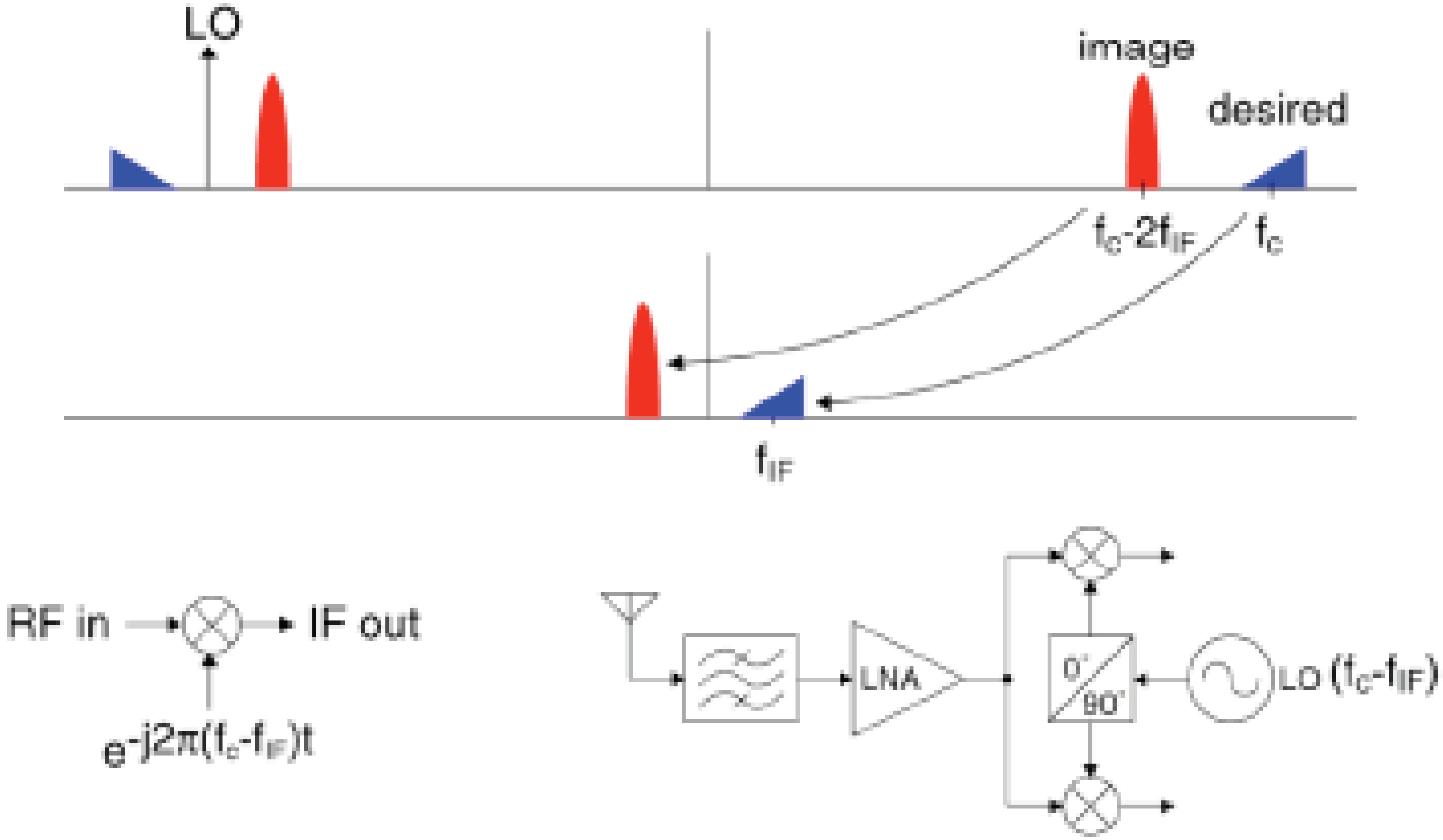
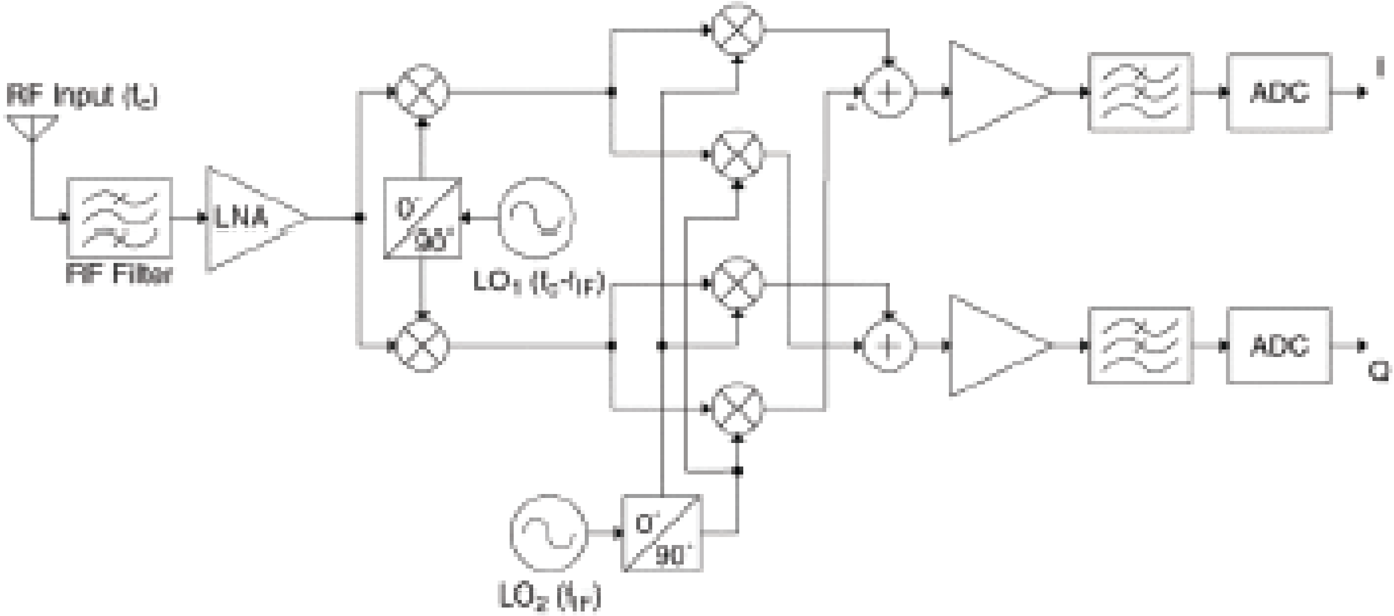
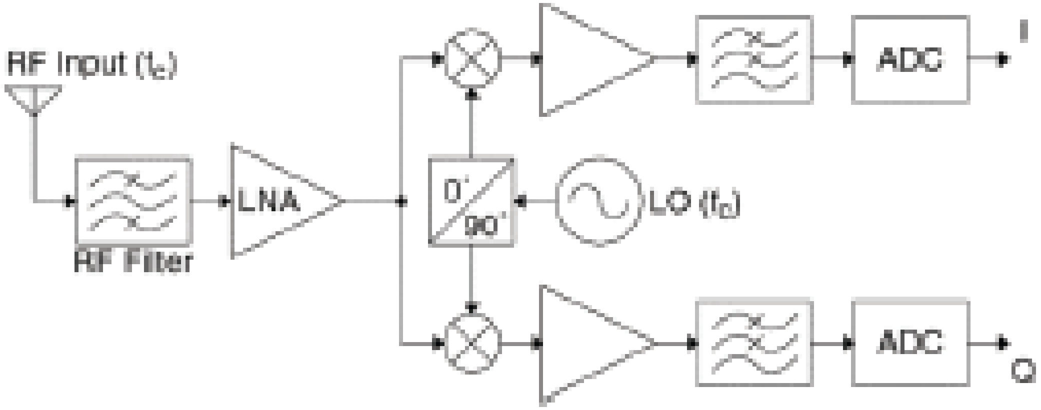


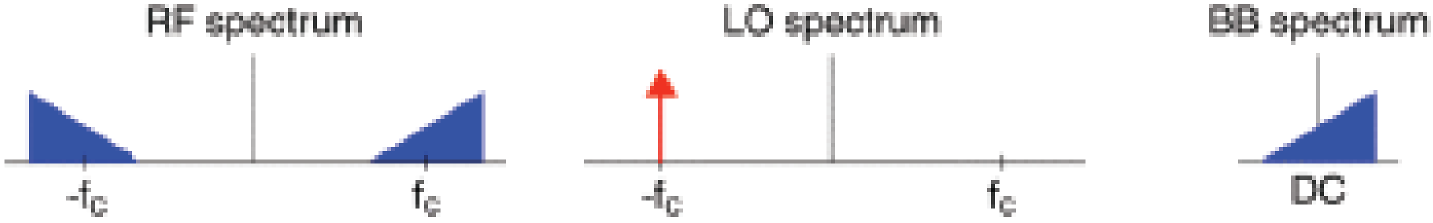
Image-Reject (Weaver) Architecture



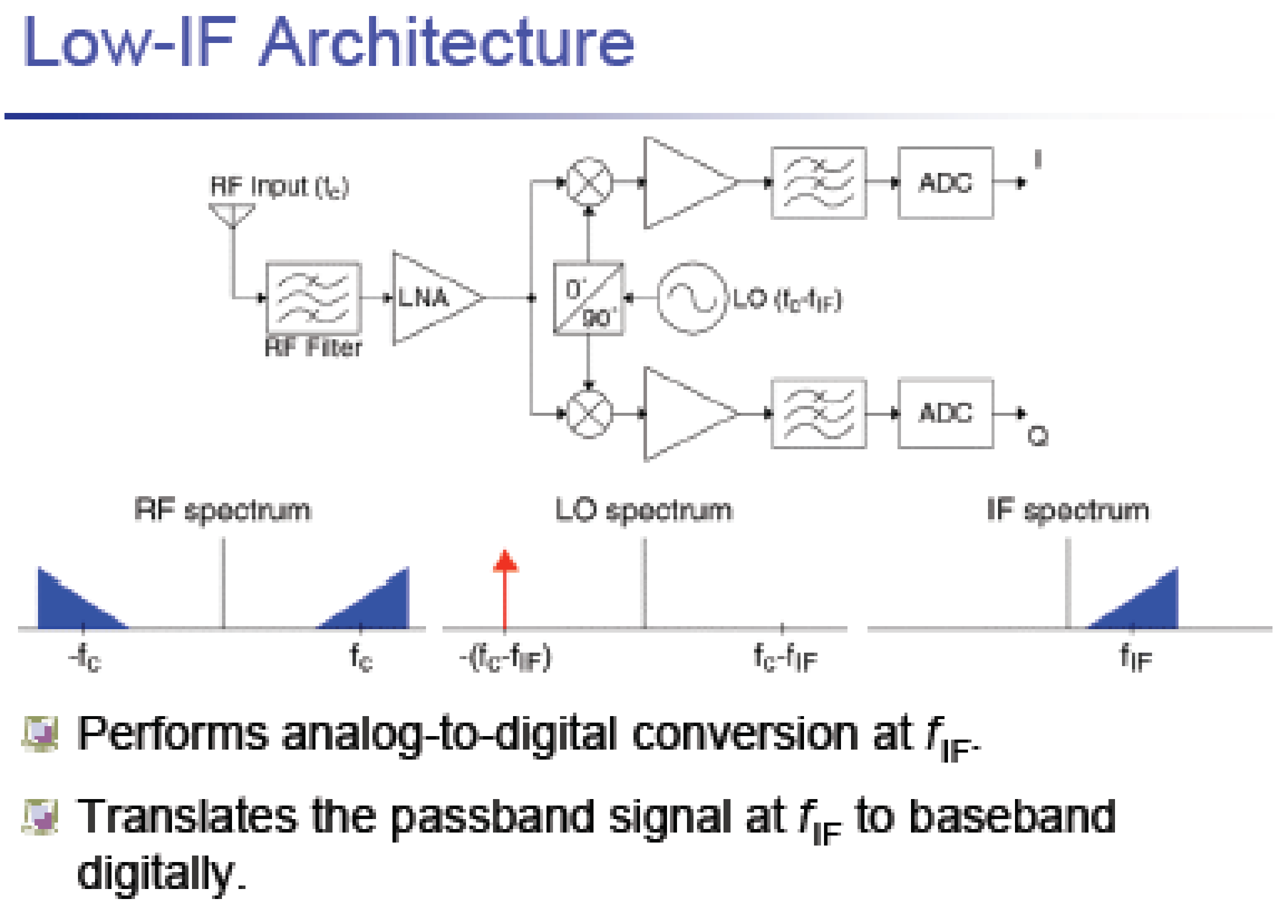
- The amount of image rejection is limited by
  - Gain mismatch between the different signal paths
  - Quadrature phase mismatch between the I and Q signals in the two local oscillators.

Direct-Conversion Architecture





- Eliminates intermediate-frequency stage(s) ⇒ no image problem.
- Pros: minimal number of RF components, high level of integration
- Cons: sensitive to DC offsets.

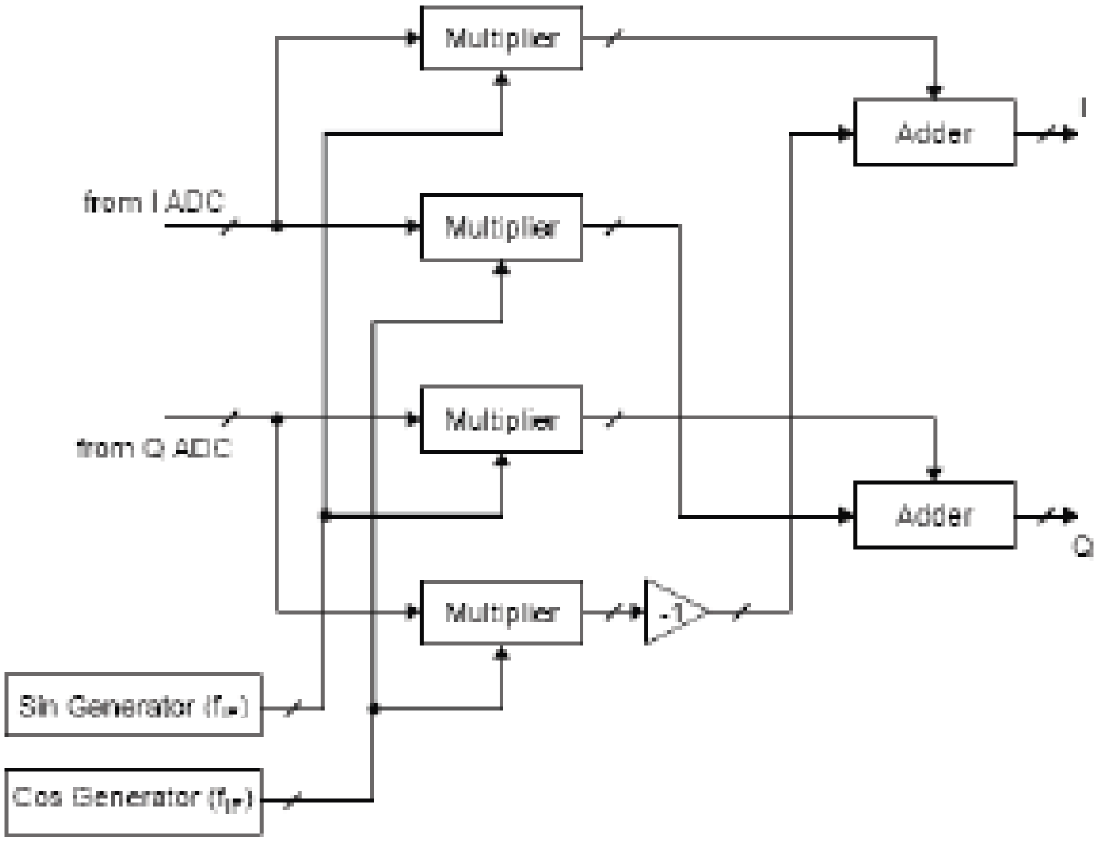


Low-IF Architecture

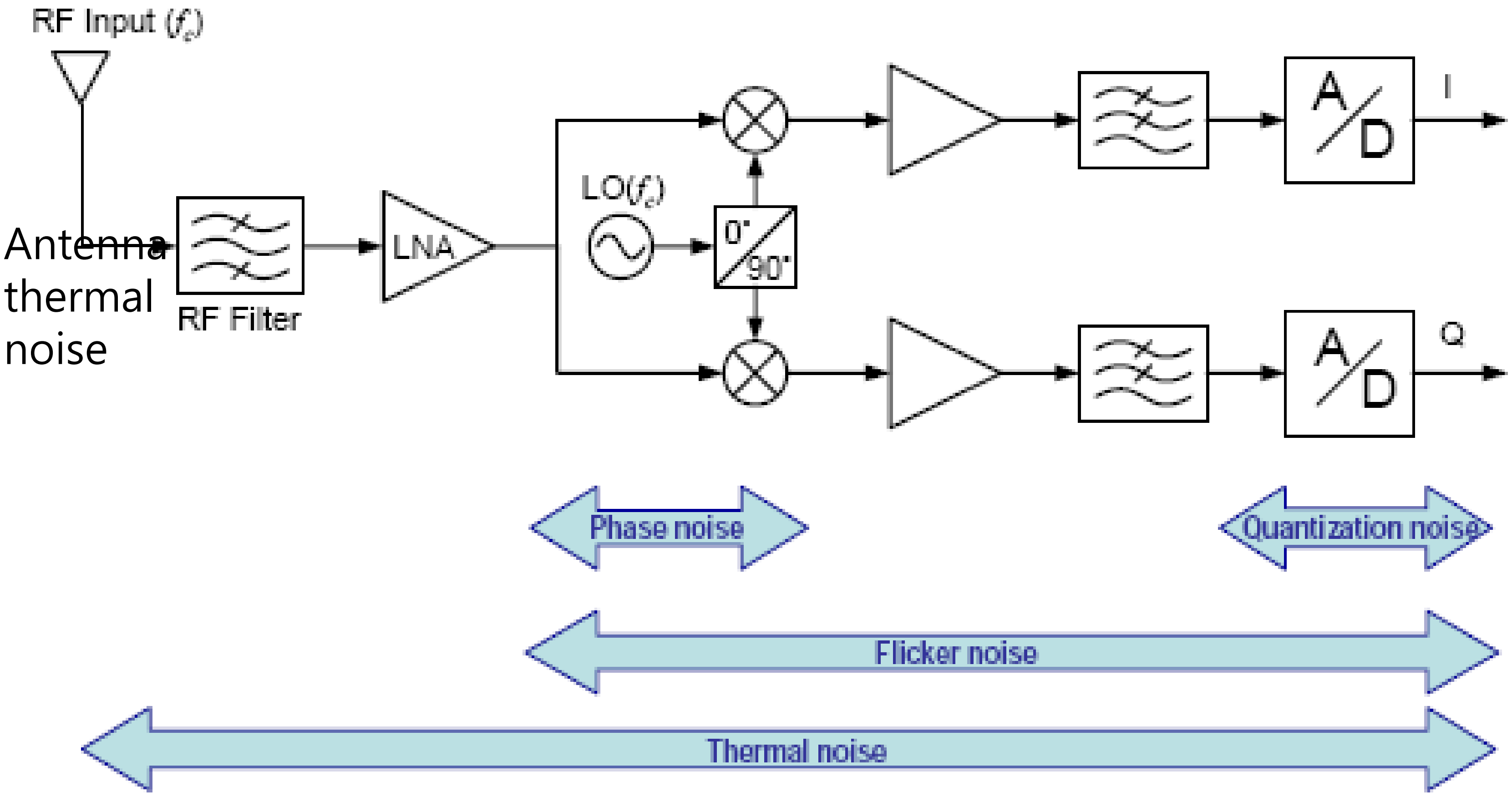
- Performs analog-to-digital conversion at f_{IF}.
- Translates the passband signal at f_{IF} to baseband digitally.

- No DC offset problem, but image problem
- Digital VCO for IF to baseband conversion

Digital Frequency Translation

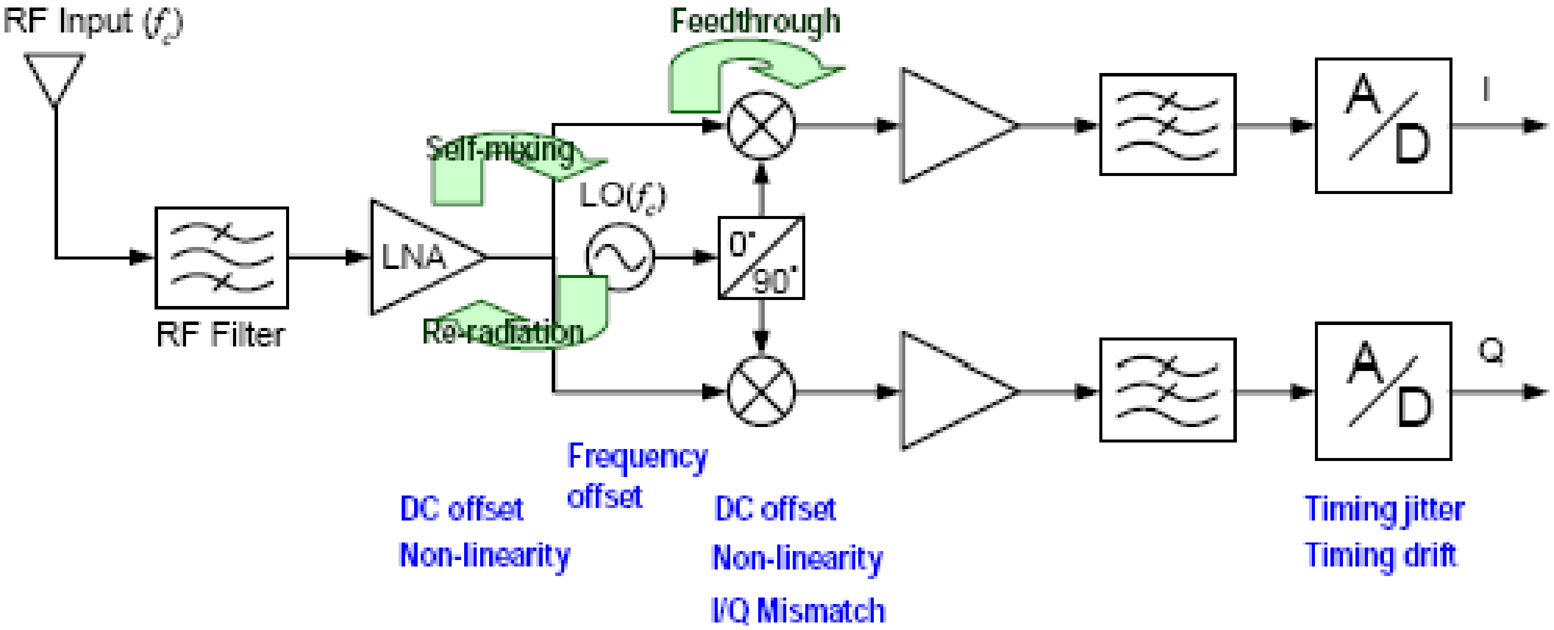


Noise Sources



Antenna thermal noise

Analog Impairments



- Distortion calculation
- Time variation

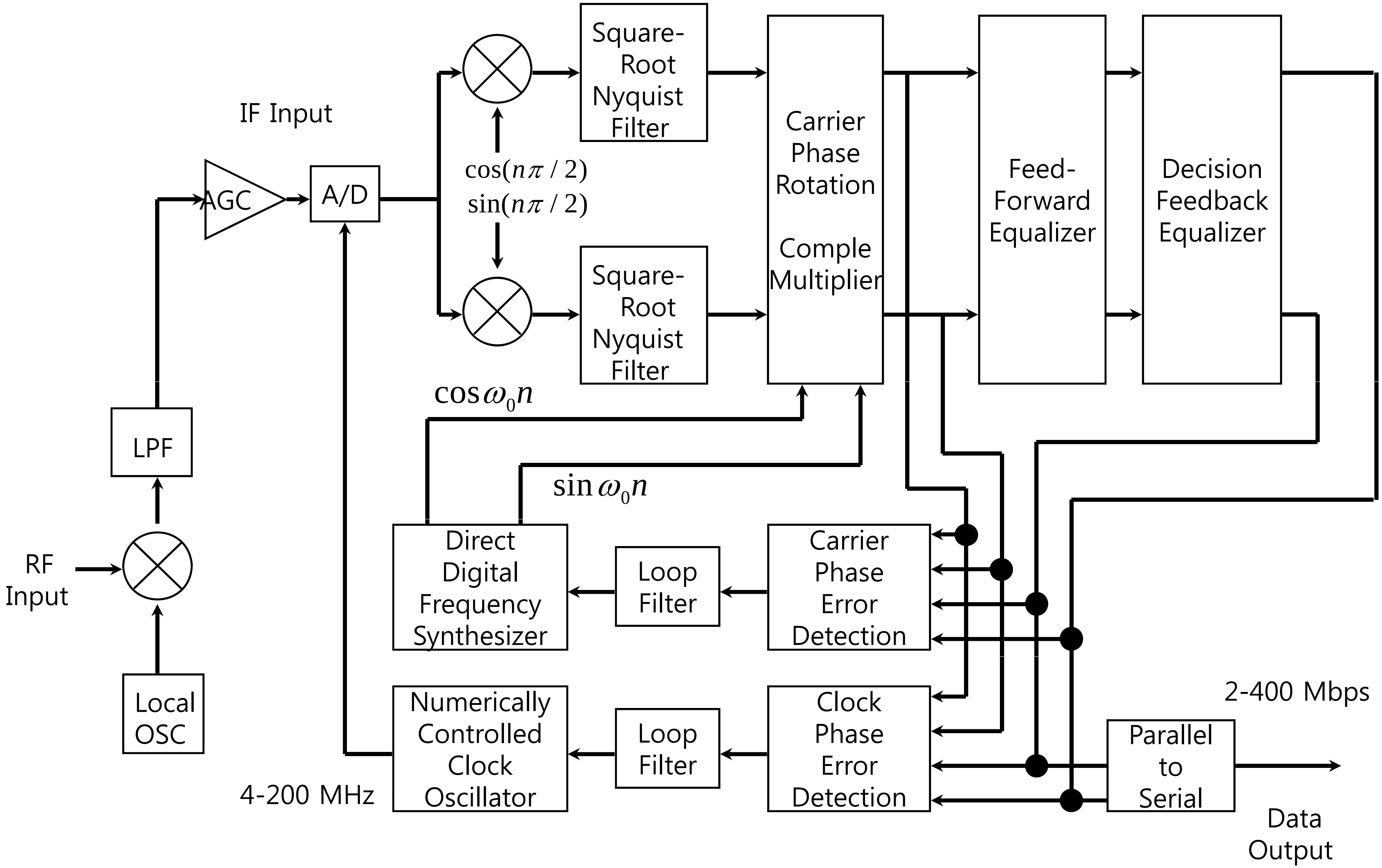
8. System Design Examples

- **High data-rate QAM modem**
  - Data rates up to 50 Mbaud
  - Modulation formats from QPSK through 256-QAM
  - Adaptive decision-feedback equalizer

- **Direct-Sequence spread spectrum transceiver**
  - 800 kbps maximum data rate
  - 12.7 Mchip/s PN spreading rate
  - Coherent BPSK receiver

- **Frequency-hopped spread spectrum transceiver**
  - 160 kbps maximum data rate
  - 26 MHz fast-hopping DDFS-based synthesizer
  - Noncoherent FSK modulation

QAM Receiver Block Diagram



IF Input

RF Input

Local OSC

LPF

AGC

A/D

$\cos(n\pi/2)$

$\sin(n\pi/2)$

Square-Root Nyquist Filter

Square-Root Nyquist Filter

Carrier Phase Rotation

Comple Multiplier

Feed-Forward Equalizer

Decision Feedback Equalizer

$\cos\omega_0 n$

$\sin\omega_0 n$

Direct Digital Frequency Synthesizer

Loop Filter

Carrier Phase Error Detection

Numerically Controlled Clock Oscillator

Loop Filter

Clock Phase Error Detection

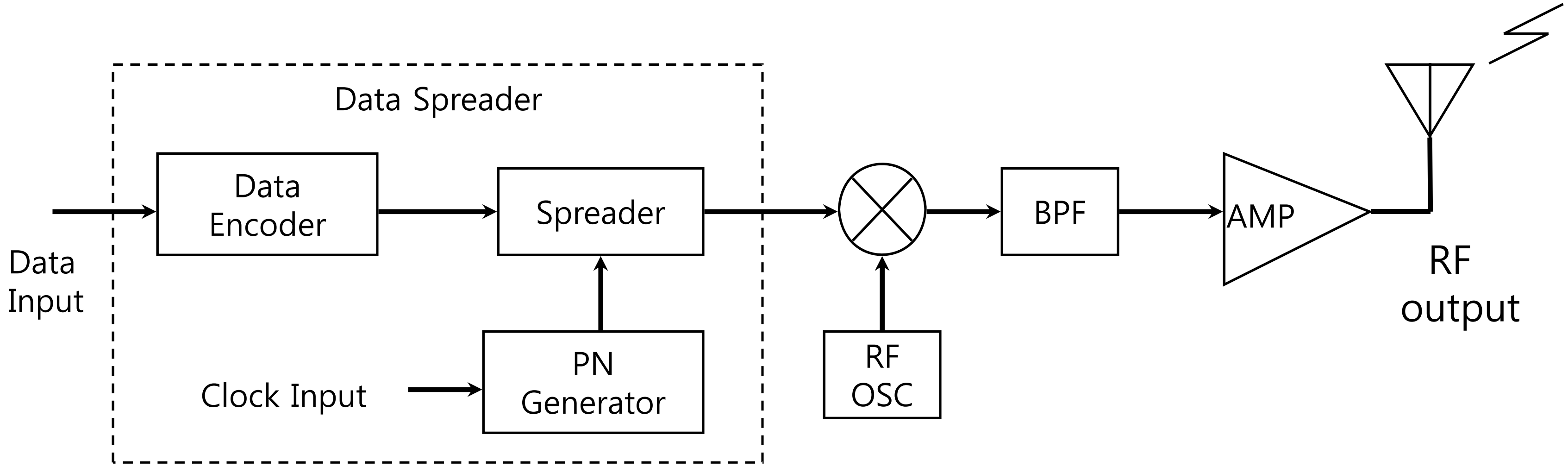
4-200 MHz

2-400 Mbps

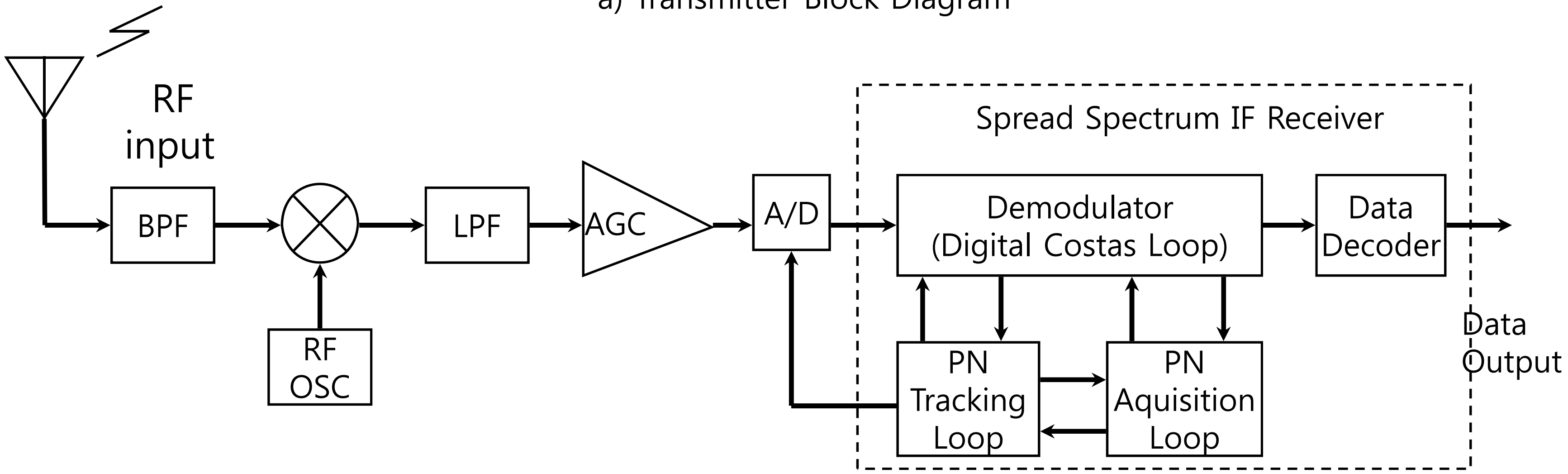
Parallel to Serial

Data Output

Direct Sequence Spread Spectrum System

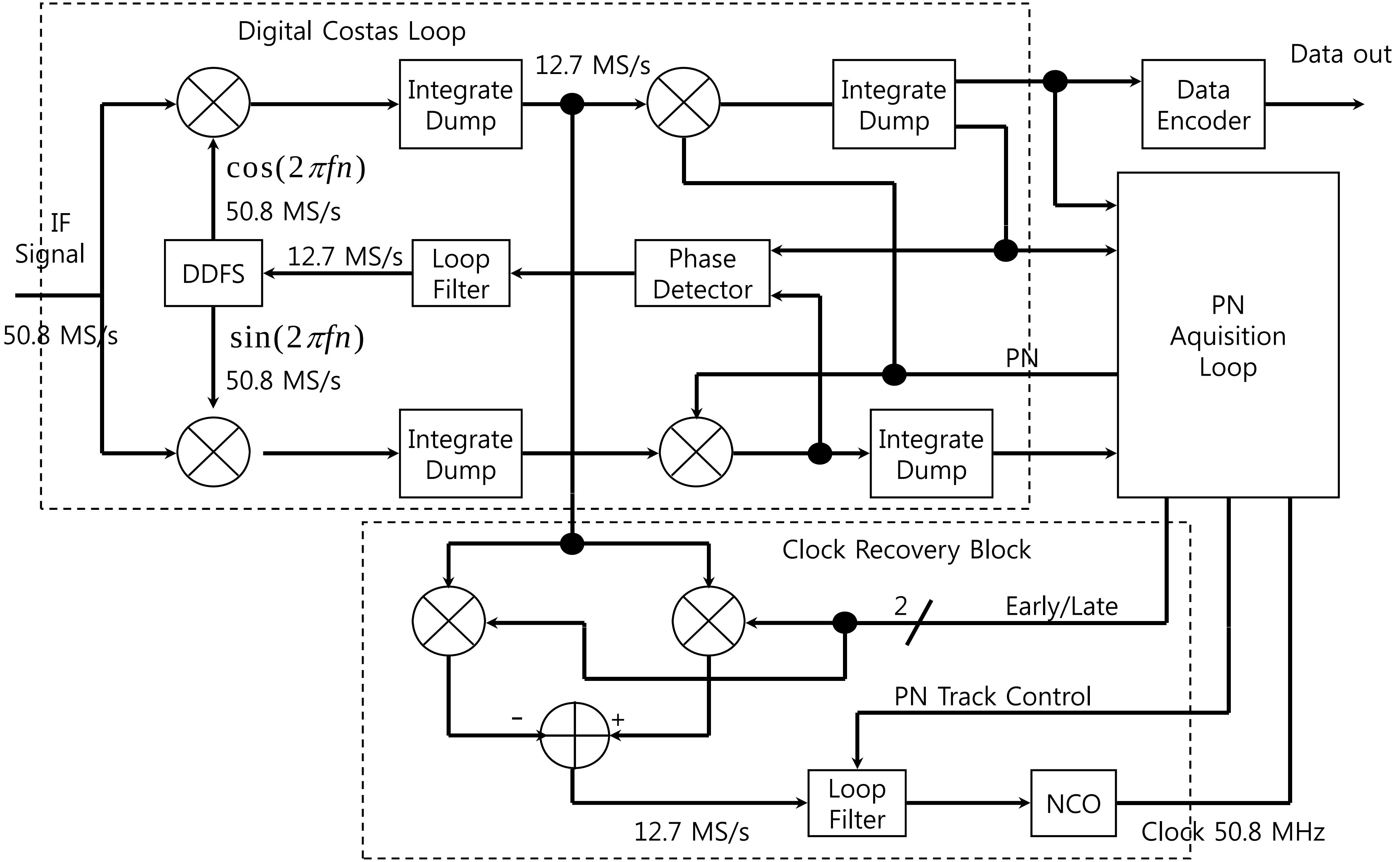


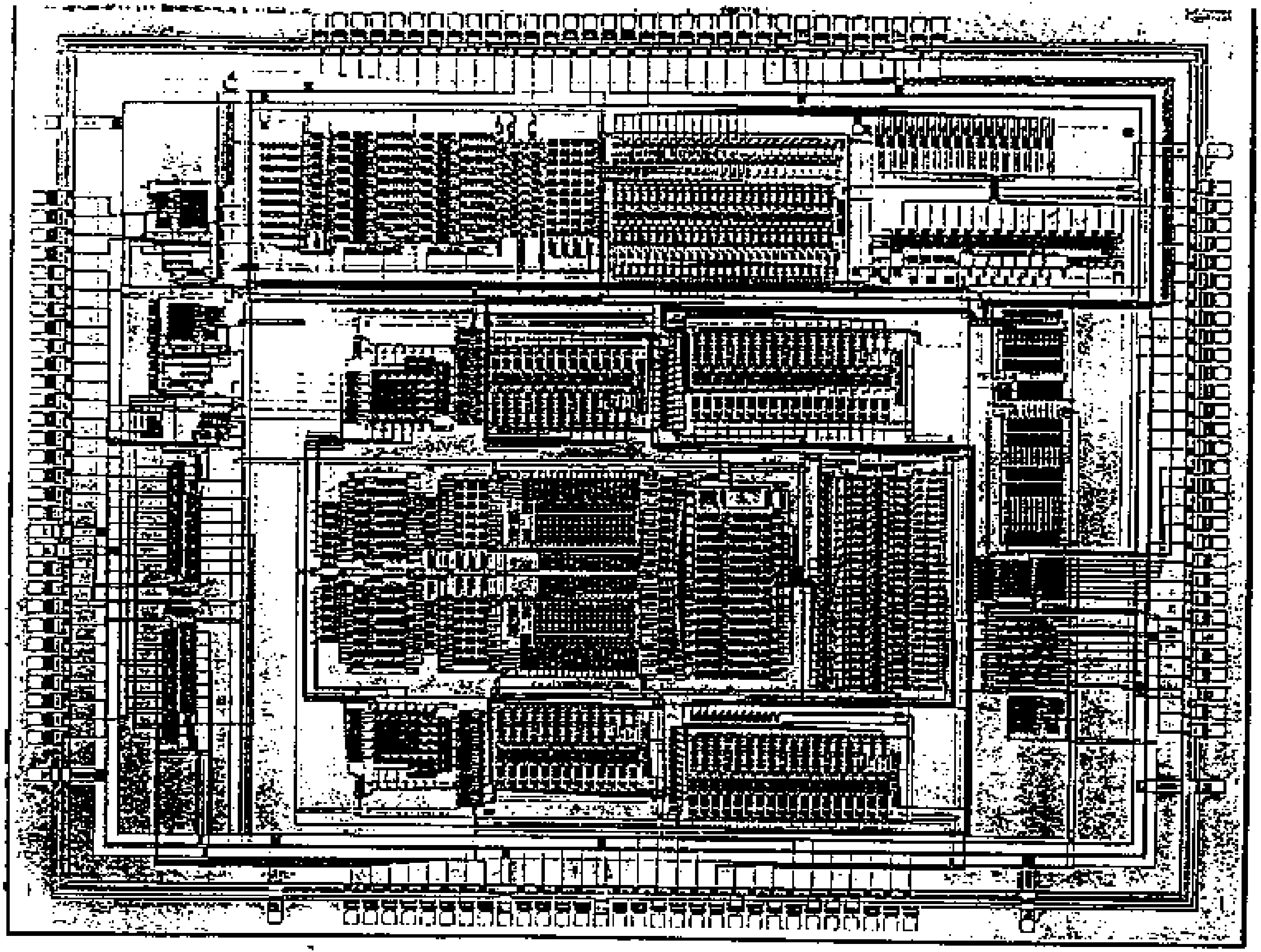
a) Transmitter Block Diagram



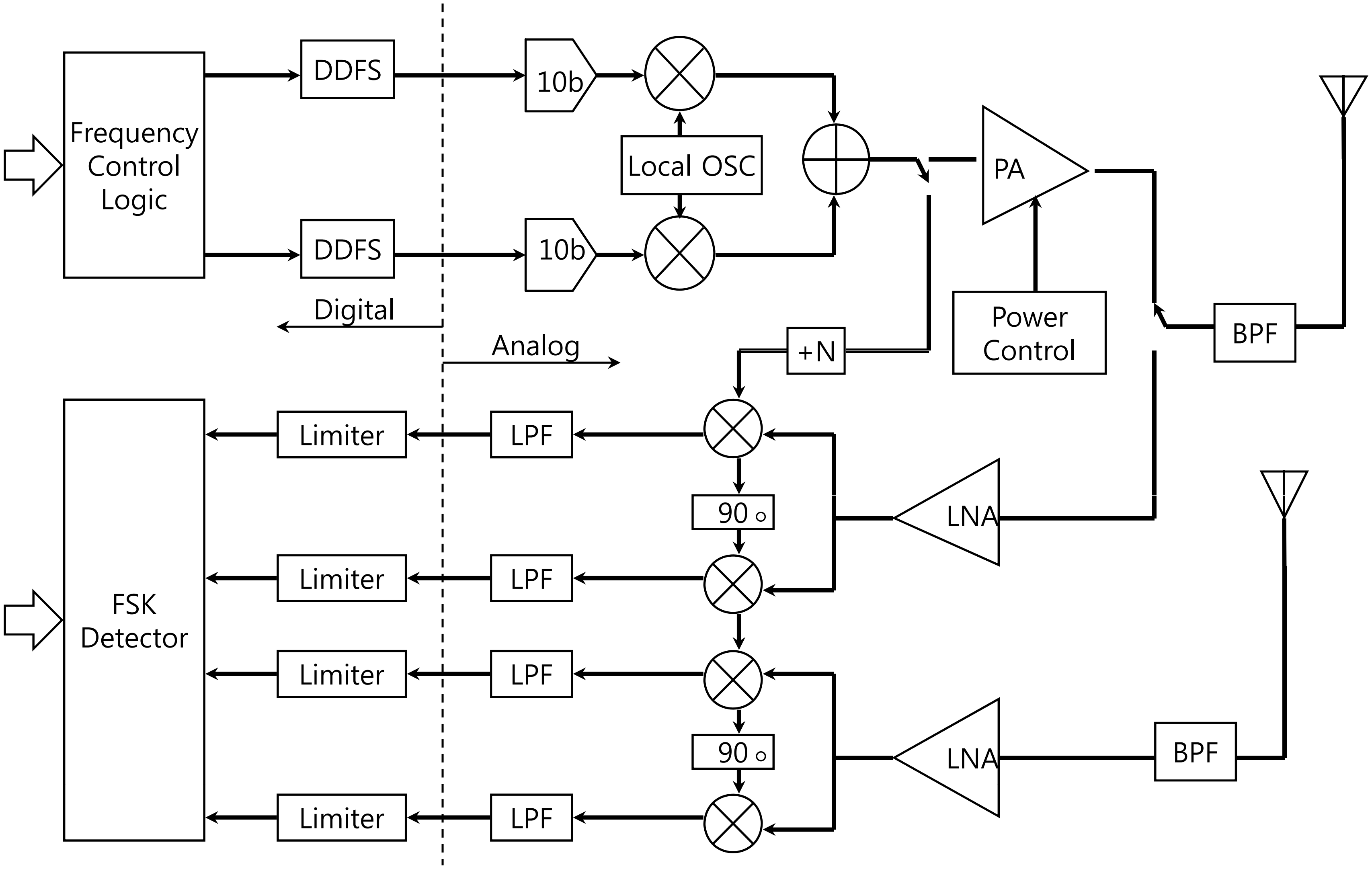
b) Recceiver Block Diagram

BPSK DS Spread Spectrum Receiver





A Low Power Frequency-Hopped Spread-Spectrum Tranceiver



Conclusions

- **The digital information superhighway will be placing increasing demands on the world's communications infrastructures thereby requiring ever more sophisticated communications devices**
- **Communications IC's will eventually become as dominant in the semiconductor industry as microprocessors and memory devices are today**

Tutorial Reference

– Communication Systems Engineering
John G. Proakis and Masoud Salehi, Prentice Hall, 1994

– "Universal Digital Portable Radio Communications"
Donald C. Cox, Proceedings of the IEEE, pp. 436-477, Apr. 1987

(Extensive bibliography provided by both references)