MOHAMED SATHAK A J COLLEGE OF ENGINEERING

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

III YEAR/ VI SEMESTER

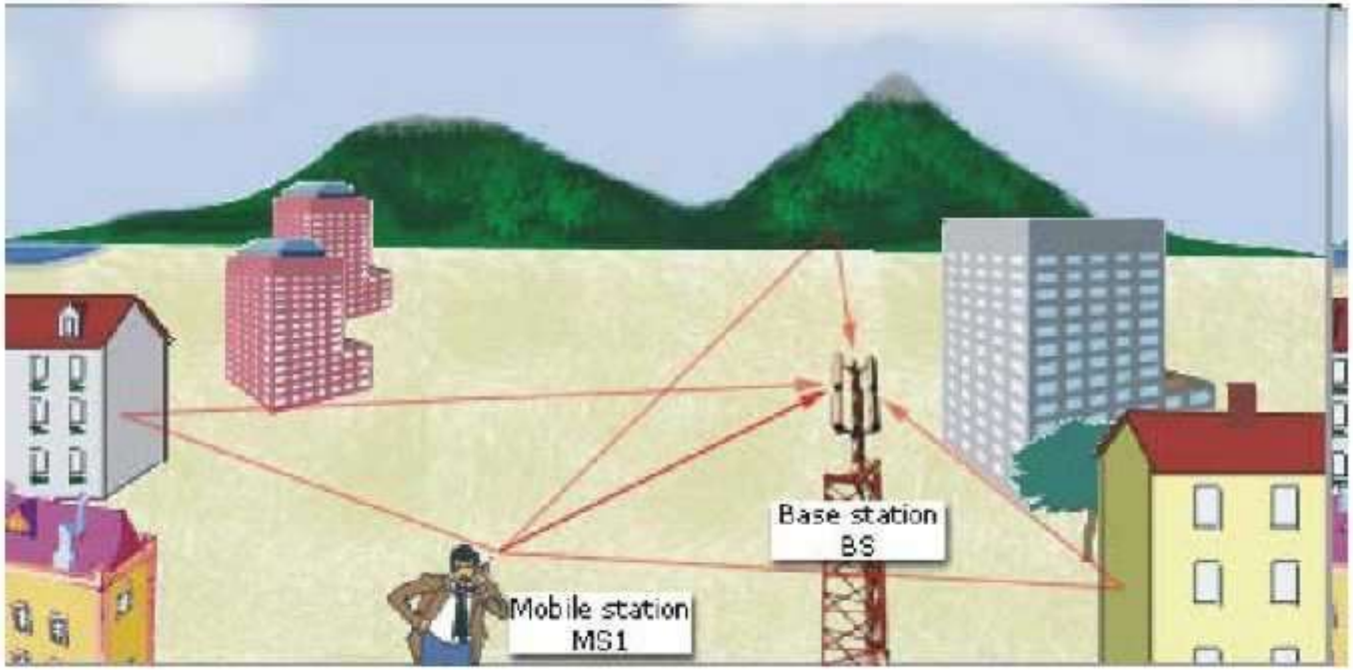
EC8652- WIRELESS COMMUNICATION

COURSE MATERIAL

UNIT - I
WIRELESS CHANNELS

Large scale path loss – Path loss models: Free Space and Two-Ray models - Link Budget design – Small scale fading-Parameters of mobile multipath channels – Time dispersion parameters-Coherence bandwidth – Doppler spread & Coherence time, Fading due to Multipath time delay spread – flat fading – frequency selective fading – Fading due to Doppler spread – fast fading – slow fading.

MULTI-PATH PROPAGATION



More version of the transmitted signal takes more than one transmission path to reach the receiver from the transmitter at a slightly different times.
MULTIPATH PROPAGATION

EFFECTS OF FADING

- Rapid changes in signal strength over a small travel distance or time interval
- Random frequency modulation due to varying Doppler shifts on different multi-path signals
- Time dispersion (echoes) caused by multi-path propagation delays

Factors Influencing small scale Fading

1. Multipath Propagation

–Presence of reflecting objects and scatters in the channel creates a constantly changing environment that dissipates the signal energy in terms of amplitude, phase and time

- time delay of signal arrival
- Multiple version of transmitted signal arrive at the receiving antenna

–Increases the time required for the actual information portion of the signal to reach the receiver due to Inter Symbol Interference

Factors Influencing small scale Fading

2. Speed of Mobile

- relative motion between base station & mobile causes random frequency modulation due to different Doppler shift (f_d)
- Different multipath components may have different frequency shifts.
- The shift in received signal frequency due to motion-Doppler shift-movement of mobile terminal towards or away from base station transmitter

Factors Influencing small scale Fading

3. Speed of Surrounding Objects

- Objects in radio channel are in motion, induce time varying Doppler shifts on multipath components.
- Doppler shift effect dominates small-scale fading if speed of objects > mobile speed

4. Transmission bandwidth of the signal

- If the transmitted radio signal bandwidth is greater than the bandwidth of the multipath channel, received signal will be distorted.
- Small scale signal will not be significant

Parameters of mobile multipath channels

Time Dispersion Parameters

“ –excess delay detupmoc seulav lla : ”**relative** to the time of first signal arrival τ_o

–

Mean excess delay (→(τAverage delay measured with respect to the first (arrival) moment of the power delay profile

$$\bar{\tau} = \frac{\sum_k P(\tau_k)\tau_k}{\sum_k P(\tau_k)}$$

Parameters of mobile multipath channels

RMS delay spread

- Square root of the second central moment of the power delay profile

$$\sigma_\tau = \sqrt{Avg(\tau^2) - (\bar{\tau})^2}$$

- RMS delay spread and mean excess delay are defined from a **single power delay profile** which is temporal or spatial average of the consecutive impulse response measurements and **averaged** over a local area

Parameters of mobile multipath channels

Maximum excess delay (X dB):

- Measured with respect to certain power level
- Defined as the time delay during which multipath energy falls to XdB below the maximum level
- Maximum excess delay = τ_X - τ_0
- τ_0 first arriving signal
- τ_X Maximum delay at which multipath components is within XdB of the strongest arriving multipath signal

TYPES OF SMALL SCALE FADING

Small-Scale Fading
(Based on multipath time delay spread)

Flat Fading
1. BW of signal < BW of channel
2. Delay spread < Symbol period

Frequency Selective Fading
1. BW of signal > BW of channel
2. Delay spread > Symbol period

Small-Scale Fading
(Based on Doppler spread)

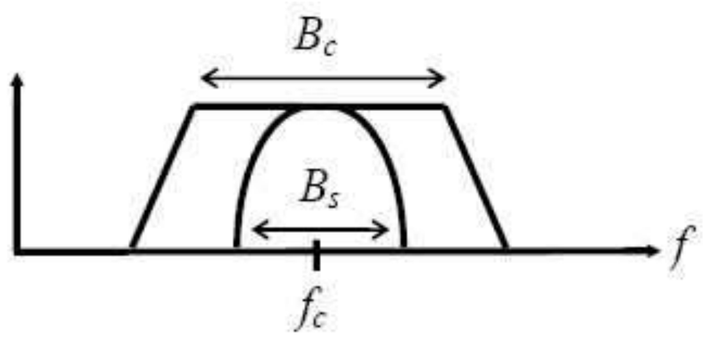
Fast Fading
1. High Doppler spread
2. Coherence time < Symbol period
3. Channel variations faster than base-band signal variations

Slow Fading
1. Low Doppler spread
2. Coherence time > Symbol period
3. Channel variations slower than baseband signal variations

Fading due to Multipath Delay

A. Flat Fading → $B_S << B_c$ or $T_S >> \sigma_\tau$

- Symbol period > Delay spread
- signal fits easily within the bandwidth of the channel
- channel BW >> signal BW
- most commonly occurring type of fading

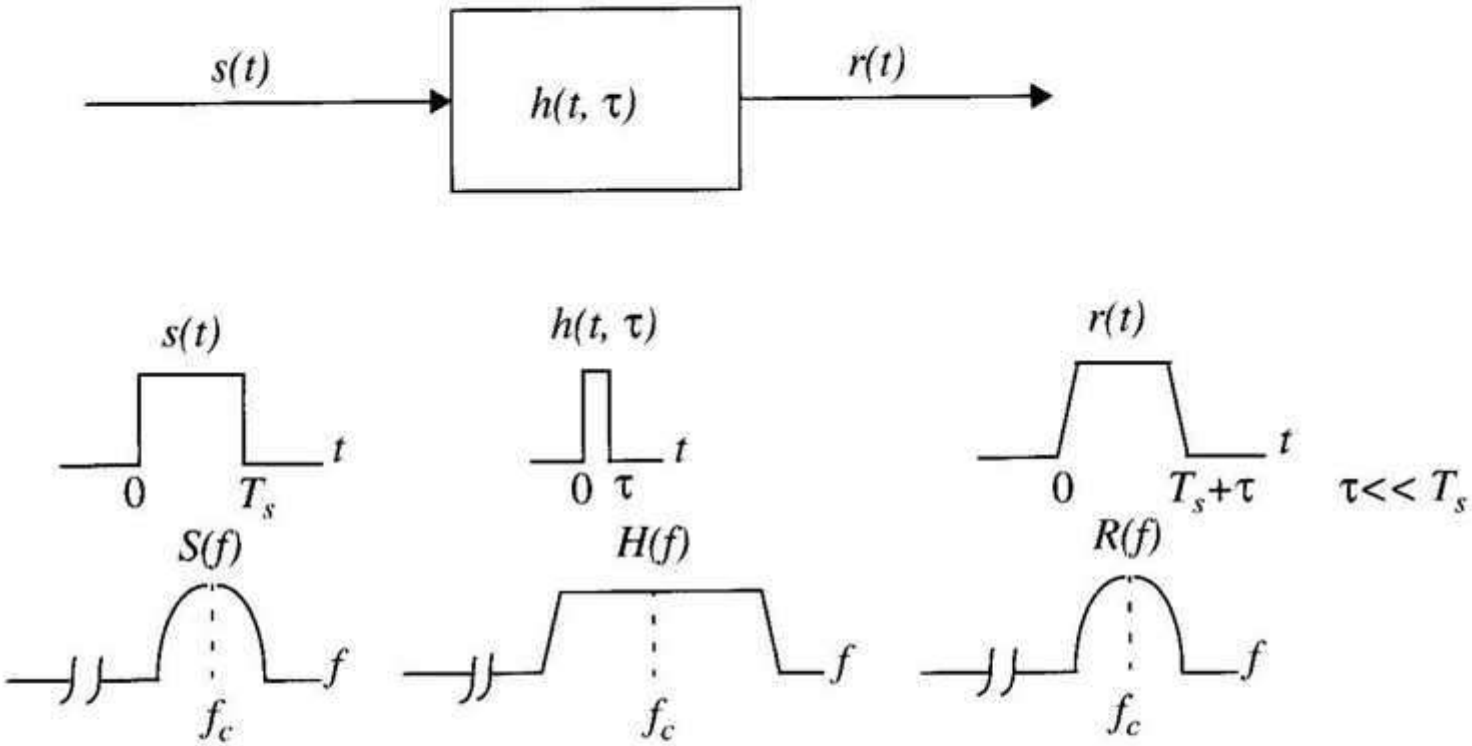


Fading due to Multipath Delay

A) Flat Fading $\rightarrow B_s << B_c$ or $T_s >> \sigma_\tau$

- spectral properties of Tx signal are preserved
  - signal is called a *narrowband* channel, since the bandwidth of the signal is narrow with respect to the channel bandwidth
  - signal is not distorted
  - Flat fading channels are known as amplitude varying channels

Flat Fading

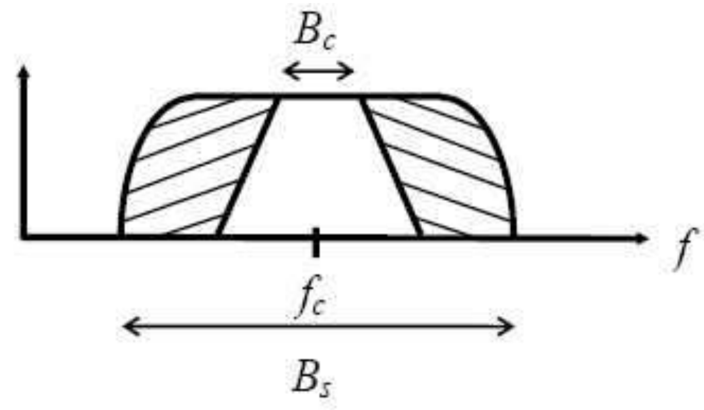


Fading due to Multipath Delay

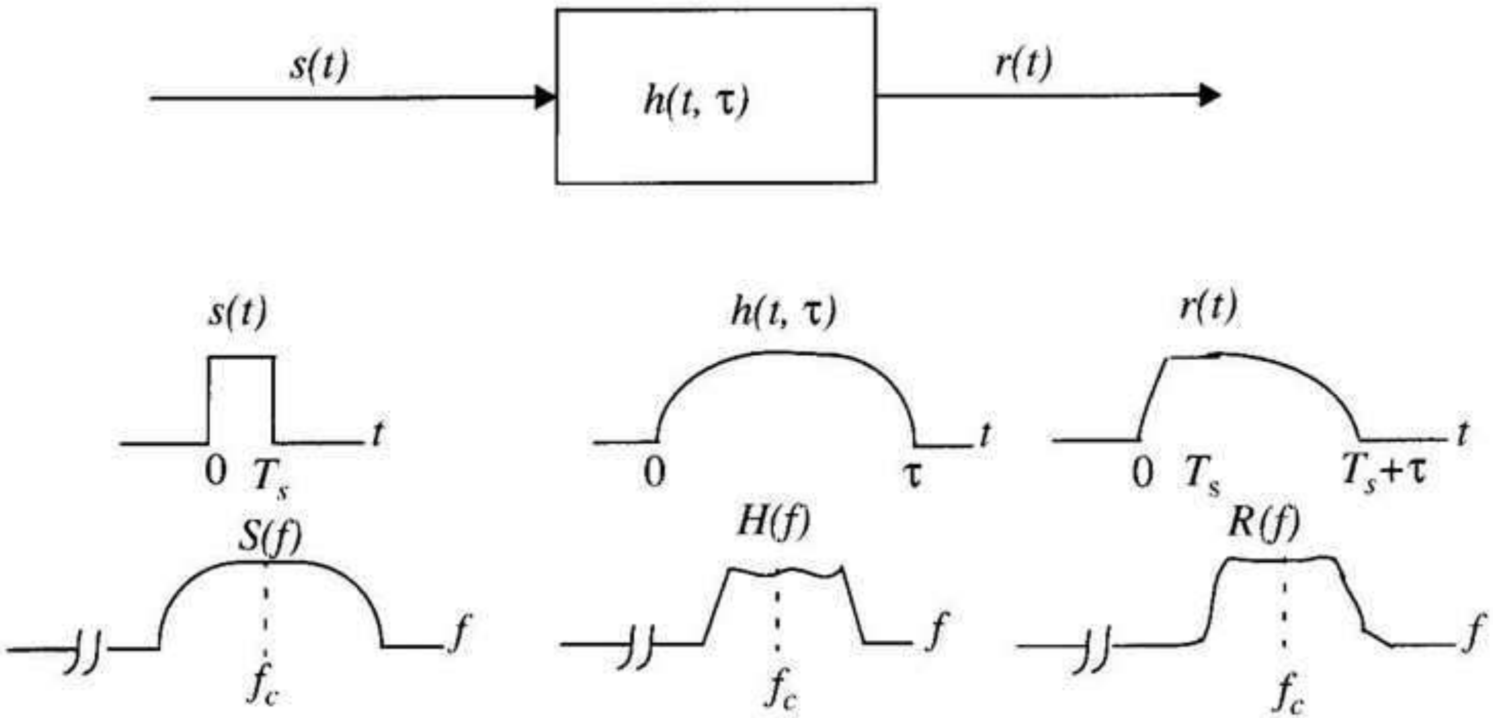
B) Frequency Selective Fading → $B_s > B_c$ or $T_s < \sigma_\tau$

– *Symbol period < Delay spread*

– $B_s > B_c$ → certain frequency components of the signal are attenuated much more than others



Frequency Selective Fading



Fading due to Doppler Spread

A) Fast Fading $\rightarrow B_s < B_D$ or $T_s > T_c$

Channel impulse response changes rapidly within the symbol duration

– $B_s < B_D$
  - Doppler shifts significantly alter spectral BW of TX signal
  - signal "spreading"

– $Ts > Tc$
  - Coherence time of the channel is smaller than the symbol period of the transmitted signal
  - Signal distortion increases relative to the bandwidth of the transmitted signal

Fading due to Doppler Spread

B) Slow Fading $\rightarrow T_s << T_c$ or $B_s >> B_D$

- Channel impulse response changes at a rate much slower than the transmitted baseband signal.
- slow amplitude fluctuations
  - B_s almost always $>> B_D$ for most applications

- Velocity of the mobile and the baseband signaling determines whether a signal undergoes fast fading or slow fading.

UNIT - 2
CELLULAR ARCHITECTURE

Multiple Access techniques - FDMA, TDMA, CDMA – Capacity calculations–Cellular concept- Frequency reuse - channel assignment- hand off- interference & system capacity- trunking & grade of service – Coverage and capacity improvement

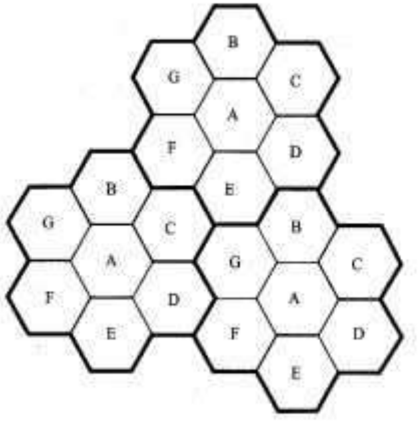
Cellular Concept

- If a given set of frequencies or radio channels can be reused without increasing the interference, then the large geographical area covered by a single high power transmitter can be divided into a number of small areas, each allocated a subset of frequencies
- Small geographical coverage, lower power transmitters with lower antennas can be used

Frequency Reuse

- Each cellular base station is allocated a group of radio channels within a small geographic area called a cell.
- Neighboring cells are assigned different channel groups.
- By limiting the coverage area to within the boundary of the cell, the channel groups may be reused to cover different cells.
- Frequency reuse or frequency planning
- Hexagonal geometry has
  - exactly six equidistance neighbors
  - each of its neighbors are

  separated by multiples of 60 degrees.



Frequency Reuse

- Consider a cellular system which has a total of S duplex channels.
- Each cell is allocated a group of k channels, $k < S$.
- The S channels are divided among N cells.
- The total number of available radio channels

$$S = kN$$

- The N cells which use the complete set of channels is called *cluster*.
- The cluster can be repeated M times within the system. The total number of channels, C, is used as a measure of capacity

$$C = MkN = MS$$

- The capacity is directly proportional to the number of replication M.
- The cluster size, N, is typically equal to 4, 7, or 12.
- Small N is desirable to maximize capacity.
- The frequency reuse factor is given by $1/N$

Rules for determining the nearest co-channel neighbours

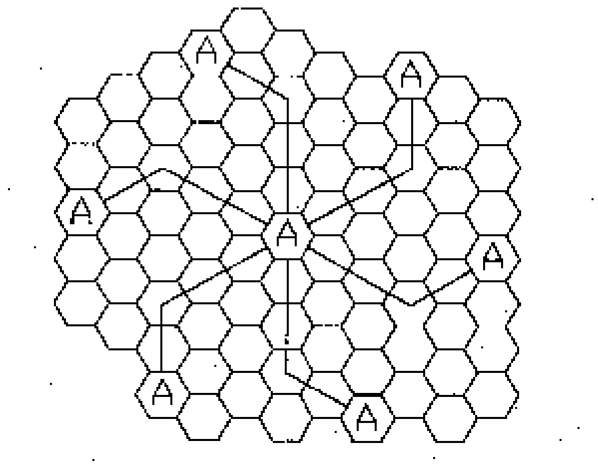
- Step 1: Move i cells along any chain of hexagons.
- Step 2: Turn 60 degrees counter clockwise and more j cells
- Only certain cluster sizes and cell layout are possible.
- The number of cells per

cluster, *N*, can only have values which satisfy

$$N = i^2 + ij + j^2$$

- Co-channel neighbors of a particular cell,

ex, *i=3* and *j=2*.



Channel Assignment Strategies

Two Channel assignment approach
- fixed channel assignment
- dynamic channel assignment

- Fixed channel assignment(FCA)
  - each cell is allocated a predetermined set of voice channel
  - any new call attempt can only be served by the unused channels in that particular cell
  - the call will be *blocked* if all channels in that cell are occupied
  - A cell is allowed to borrow channels from neighbouring cell if all of its own channels are already occupied

Channel Assignment Strategies

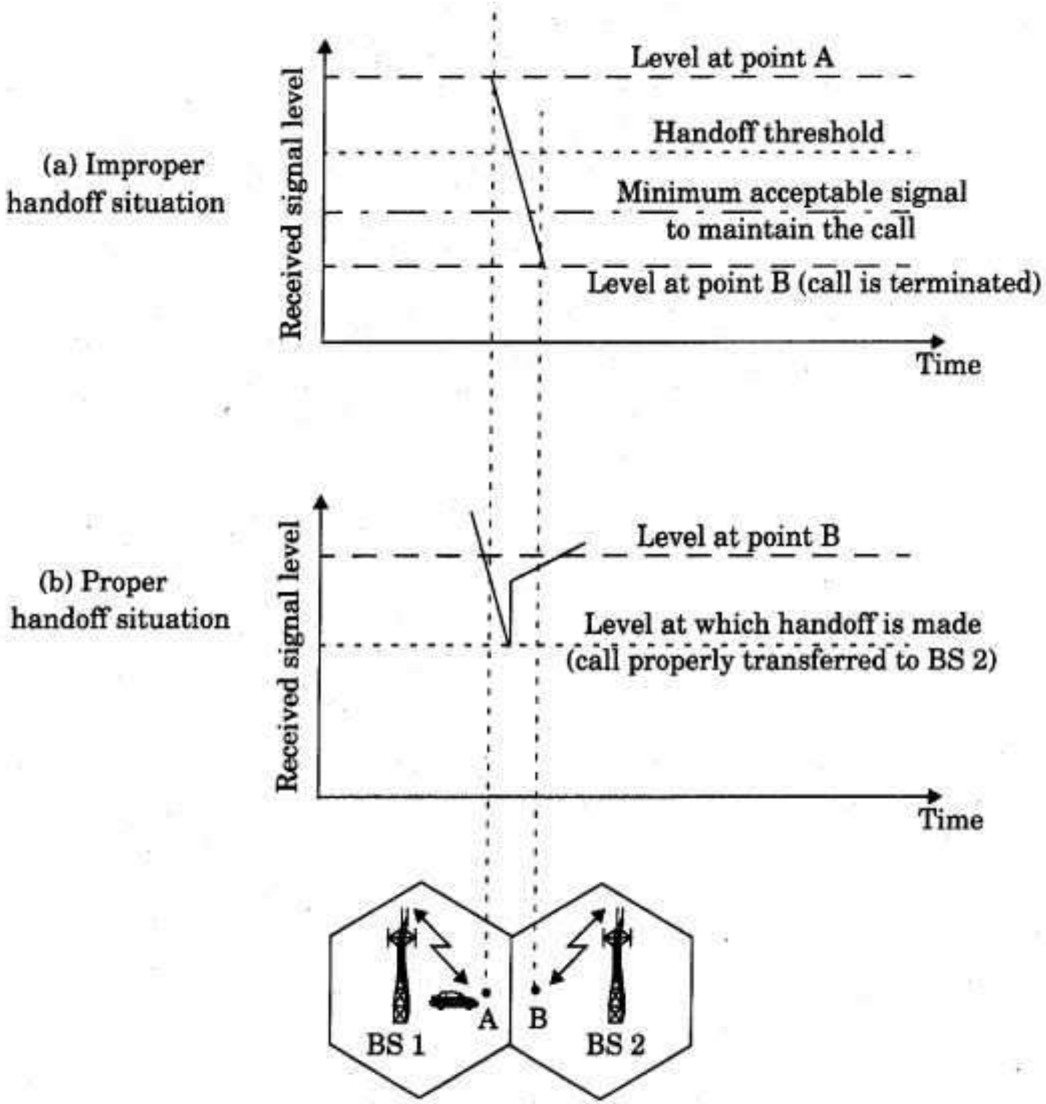
- **Dynamic channel assignment(DCA)**
  - allocate channels based on request.
  - channels are not allocated to cells permanently
  - reduce the likelihood of blocking, increase capacity.
  - Simple to implement
  - If traffic in the network is uniform , number of active users in each cell are same, results in optimum channel allocation strategy

Handoff Strategies

When a mobile moves into a different cell while a conversation is in progress, the MSC automatically transfers the call to a new channel belonging to the new base station.

- Handoff operation
  - identifying a new base station
  - re-allocating the voice and control channels with the new base station.
- Handoff margin $\Delta = P_{r,handoff} - P_{r,minimum\ usable}$
  - If is too large, unnecessary handoffs burden the MSC
  - If is too small, there may be insufficient time to complete handoff before a call is lost.

Handoff Strategies



Handoff Methods

- **Mobile controlled Handoff(MCHO)**

  Desirable method because it reduces the burden on the network
- **Network controlled Handoff(NCHO)**

  BS or Access point(AP) monitor the signal quality from the mobile and report the measurements to the MSC. MSC is responsible for choosing AP and initiating the handoff
- **Mobile Assisted Handoff(MAHO)**

  Mobile measures the signal levels from various APs. The mobile collects a set of power levels from different AP and feeds it back to MSC for decision making

Co-channel Interference and System Capacity

- Interference between the signals from co-channel cells
- To reduce co-channel interference, co-channel cell must be separated by a minimum distance.
- When the size of each cell is $\longrightarrow$ same
- Base station transmit $\longrightarrow$ same power
  - co-channel interference is independent of the transmitted power
  - co-channel interference is a function of
    - *R*: Radius of the cell
    - *D*: distance to the center of the nearest co-channel cell

Co-channel Interference and System Capacity

- Let i_0 be the number of co-channel interfering cells. The signal-to-interference ratio (S/I) for a mobile receiver can be expressed as

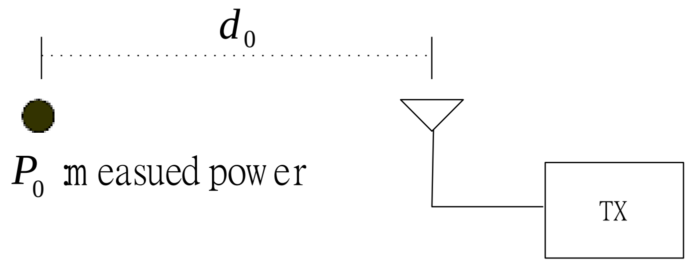
$$\frac{S}{I} = \frac{S}{\sum_{i=1}^{i_0} I_i}$$

S: the desired signal power

I_i : interference power caused by the *i*th interfering co-channel cell base station

- The average received power at a distance *d* from the transmitting antenna is

$$P_r = P_0 \left(\frac{d}{d_0} \right)^{-n}$$



n is the path loss exponent.

P_0 Power received at the reference point in far field region at a distance d_0

Co-channel Interference and System Capacity

•When the transmission power of each base station is equal path loss exponent is same, throughout the coverage area,

S/I is

$$\frac{S}{I} = \frac{R^{-n}}{\sum_{i=1}^{i_0} (D_i)^{-n}}$$

- Di – distance between the i-th interferer and the mobile
- If all the interfering base stations are equidistant from the desired station and distance is equal to distance D between the cell centers,

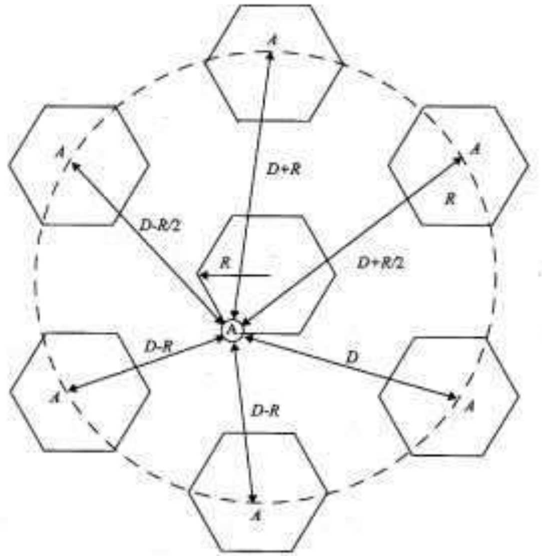
$$\frac{S}{I} = \frac{(D/R)^n}{i_0} = \frac{\left(\sqrt{3N}\right)^n}{i_0}$$

Co-channel Interference and System Capacity

- For hexagonal geometry with 7-cell cluster, with the mobile being at the cell boundary, experiences worst case Co-channel Interference , **signal-to-interference ratio for the worst case** can be approximated as

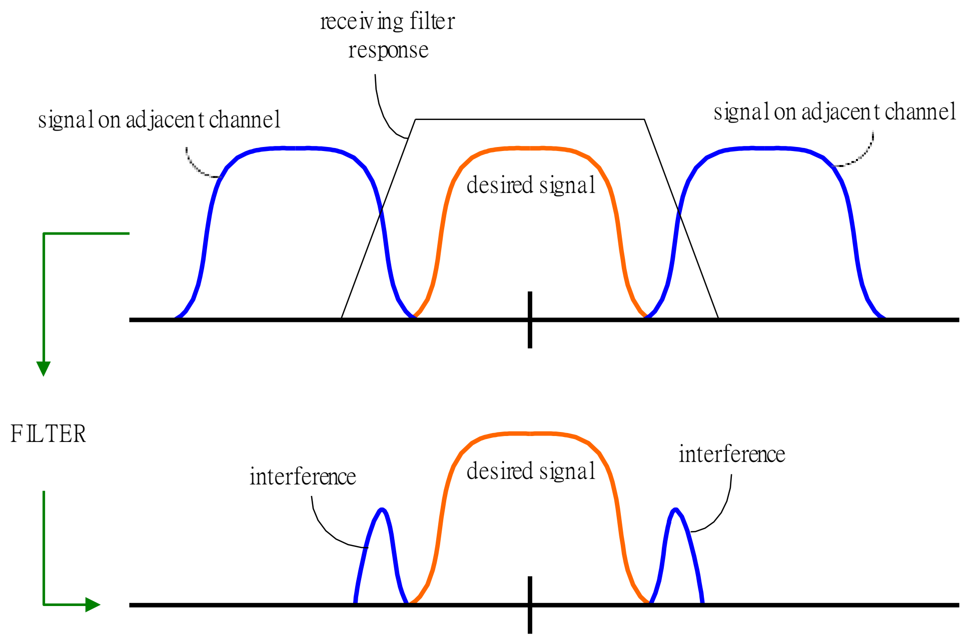
$$\frac{S}{I} = \frac{R^{4-}}{2(D-R)^{-4} + (D-R/2)^{-4} + (D+R/2)^{-4} + (D+R)^{-4} + D^{-4}}$$

The mobile is at a distance D-R from the two nearest co-channel Interfering cells and is exactly D+R/2, D, D-R/2 and D+R from the other interfering cells



Adjacent Channel Interference

- An interference resulting from the signals which are adjacent in frequency to the desired signal.
  - Imperfect receiver filters allow nearby frequencies to leak into the passband
  - Performance degrade seriously due to *near-far* effect.



Power Control for Reducing Interference

- Power levels transmitted by every subscriber unit are under constant control
- Each mobile transmits small power to maintain good quality link
- Power control
  - maintain prolong battery life
  - reduces S/I in the system
  - Solve near far problem

Trunking and Grade of Service

Erlangs: One Erlangs represents the amount of traffic density carried by a channel that is completely occupied

- **Grade of Service (GOS)**: Measure of ability of a user to access the trunking system during the busiest hour
- Trunking allows a large number of users to share relatively small number of channels in a cell by providing access to each user on demand.

Trunking and Grade of Service

- Each user generates a traffic intensity of Erlangs given by

$$A_u \quad \mathbf{H\lambda} =$$

H: average duration of a call.

λ : average number of call requests per unit time

- For a system containing U users and an unspecified number of channels, the total offered traffic intensity A, is given by

$$A = UA_u$$

- For C channel trunking system, the traffic intensity per channel, A_c is given as

$$A_c = UA_u / C$$

Trunking and Grade of Service

Trunking theory : is used to **determine the number of channels** required to service a certain offered traffic at a specific GOS.

Set-Up time: Time required to allocate a trunked radio channel to a requesting user

Blocked Call: Call which cannot be completed at the time of request due to congestion, also referred as lost call

Holding Time: Average duration of a typical call

Trunking efficiency(A) : Measure of number of users which can be offered a particular GOS with a particular configuration of fixed channels

Trunking and Grade of Service

***Types of trunked system*:**

i) Blocked calls cleared

ii) Blocked calls delayed

Blocked calls cleared

-Offers no queing for call requests. User is given an immediate access to a channel if one is available

-If no channels are available the requesting user is blocked

-In terms of Traffic intensity A, probability of blocking given Erlang loss formula

-Pr[blocking] = GOS $= \frac{A^C/C!}{\sum A^K/k!}$

C – Number of trunked channels

A – Total offered traffic

Trunking and Grade of Service

Blocked calls delayed

- If a channel is not immediately available, the call request may be delayed until a channel becomes available

- Defined as "the probability that a call is blocked after waiting a specific length of time in the queue"

- Likelihood of a call not having an immediate access to a channel is determined by Erlang C formula

- $\Pr[\text{delay}>0] = \dfrac{A^C}{A^C + C!(1 + A/C)\sum A^K/k!}$

Trunking and Grade of Service

Blocked calls delayed

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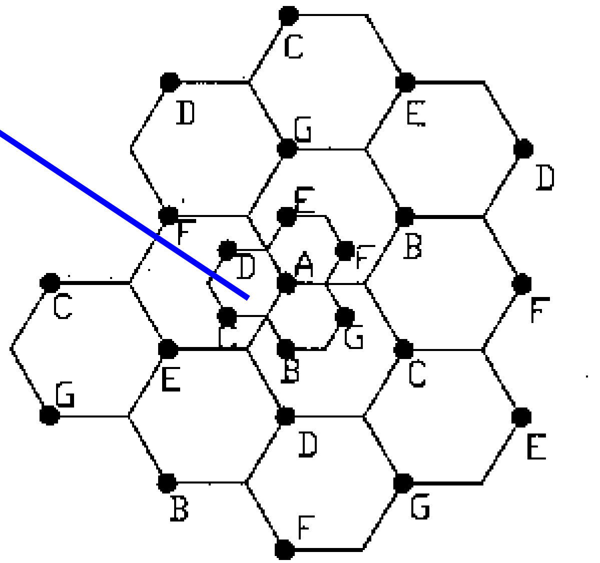
Improving the coverage and Capacity in Cellular Systems

- Methods for improving capacity in cellular systems
  - Cell Splitting: subdividing a congested cell into smaller cells.
  - Sectoring: directional antennas to control the interference
  - Coverage zone approach: Distributing the coverage of a cell and extends the cell boundary

Cell Splitting

- Split congested cell into smaller cells(reduction in antenna height and transmitted power)
- Increases the capacity of cellular system(frequency reuse)
- Radius is reduced half that of original

microcell



Reduce *R* to *R/2*

Cell Splitting

- Cell splitting reduces the cell blocking probability and increases the frequency
- Transmission power reduction from P_{t1} to P_{t2}
- Examining the **receiving power** at the new and old cell boundary

$$P_r[\text{at old cell boundary}] \propto P_{t1} R^{-n}$$

$$P_r[\text{at new cell boundary}] \propto P_{t2} (R/2)^{-n}$$

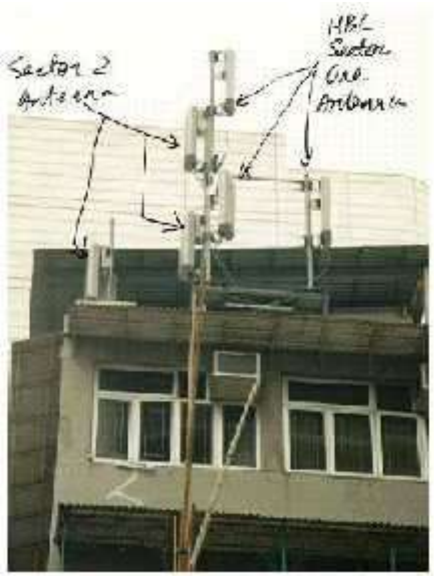
- If we take $n = 4$ and set the received power equal to each other

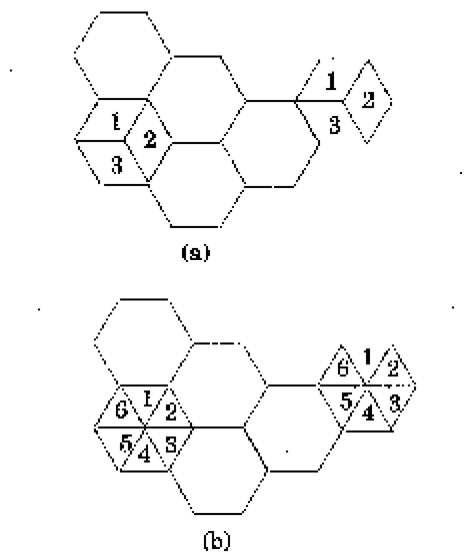
$$P_{t2} = \frac{P_{t1}}{16}$$

- Problem: if only part of the cells are splitted
  – Different cell sizes will exist simultaneously

Sectoring(Directional Antennas)

- Decrease the *co-channel interference* and keep the cell radius *R* unchanged
  – Replacing single omni-directional antenna by several directional antennas
  – Radiating within a specified sector
  – Cell is partitioned into three 120 sectors or six 60 sectors



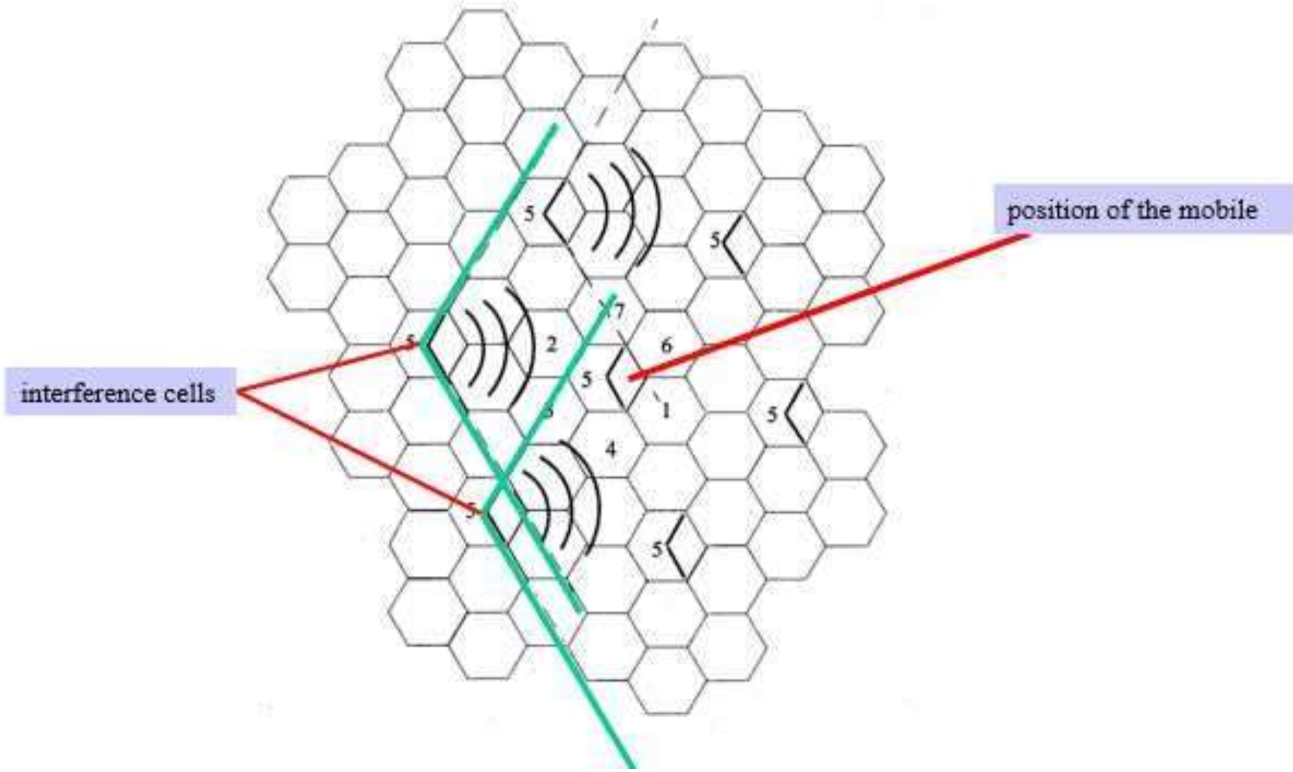


Sectoring(Directional Antennas)

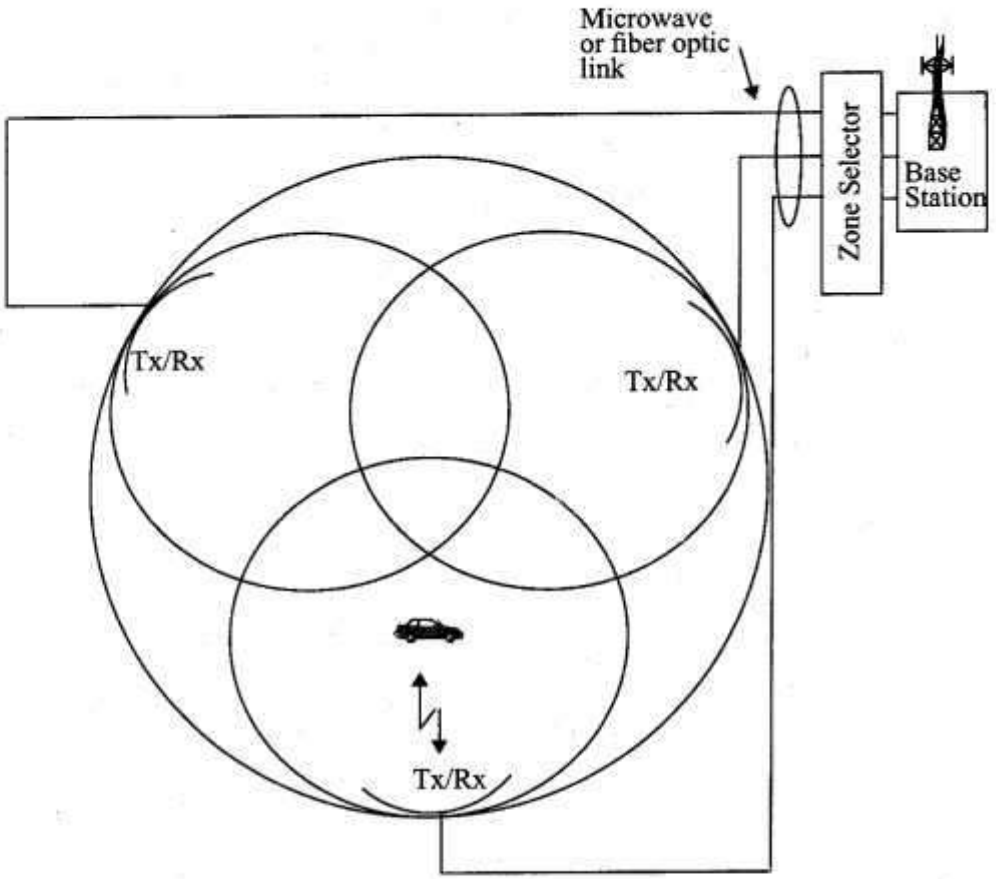
- Directional antennas increase the system capacity.
- **Reduce signal-to-interference** ratio(S/I) thus reduces cluster size, thereby **increasing the capacity**
- Reduces Co-channel interference thus increases the system performance
- Area covered by each directional antenna – section or sector
- Co-channel interference is reduced by the amount of sectoring used.

Sectoring(Directional Antennas)

- Interference Reduction



Microcell Zone Concept



Microcell Zone Concept

- Antennas are placed at the outer edges of the cell(reduce the number of BS)
- **Lee's** Microcell zone technique exploits corner excited BS to reduce the number of handoffs and thus eliminates partitioning of channels between the sectors of a cell
- Any channel may be assigned to any zone by the base station
- Mobile is served by the zone with the strongest signal.

Microcell Zone Concept

- Handoff within a cell
  - No channel re-assignment
- Mobile travels from one zone to another within the cell, it retains same channel
- Base station switches the channel to a different zone site
- Given channel is active only in the particular zone in which the mobile is traveling
- Reduce interference
  - Low power transmitters are employed

UNIT - 3
DIGITAL SIGNALING FOR FADING CHANNEL

Structure of a wireless communication link, Principles of Offset-QPSK, p/4-DQPSK, Minimum Shift Keying, Gaussian Minimum Shift Keying, Error performance in fading channels, OFDM principle – Cyclic prefix, Windowing, PAPR.

Wireless communication link

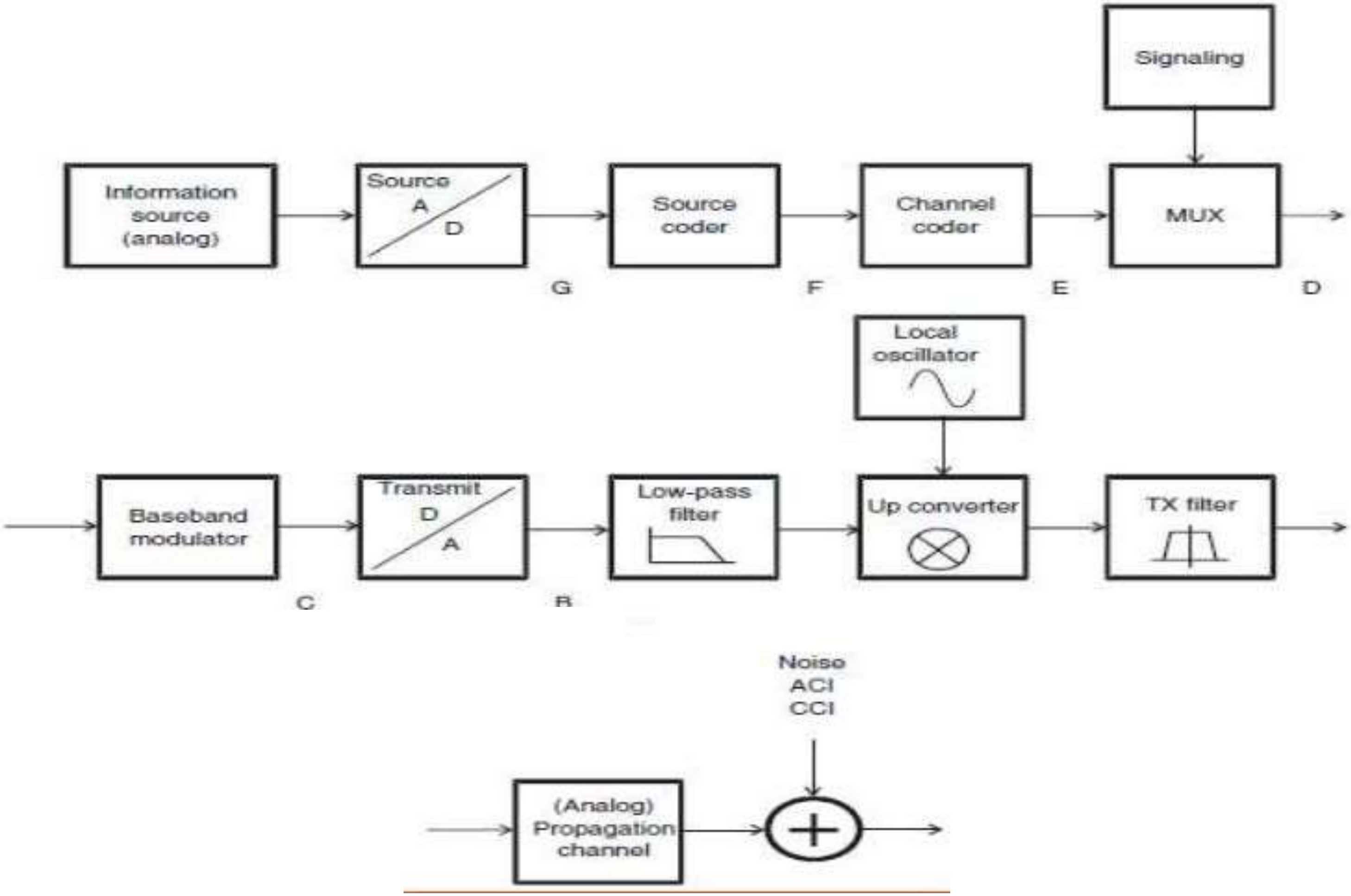


Fig: Block diagram of a digital transmitter and channel

Wireless communication link

The information source provides an analog source signal and feeds it into the source ADC and then converts the signal into a stream of digital data

- The source coder uses a priori information on the properties of the source data in order to reduce redundancy in the source signal.

Eg:(GSM) speech coder reduces the source data rate from 64 kbit/s mentioned above to 13 kbit/s.

- The channel coder adds redundancy in order to protect data against transmission errors.
- Signaling adds control information for the establishing and ending of connections, for associating information with the correct users, synchronization, etc.

Wireless communication link

- The TX Digital to Analog Converter (DAC) generates a pair of analog, discrete amplitude voltages corresponding to the real and imaginary part of the transmit symbols.
- The analog low-pass filter in the TX eliminates the (inevitable) spectral components outside the desired transmission bandwidth.
- The TX Local Oscillator (LO) provides an unmodulated sinusoidal signal, corresponding to one of the admissible center frequencies.
- The upconverter converts the analog, filtered baseband signal to a passband signal by mixing it with the LO signal.
- The RF TX filter eliminates out-of-band emissions in the RF domain.
- The (analog) propagation channel attenuates the signal, and leads to delay and frequency dispersion

Wireless communication link

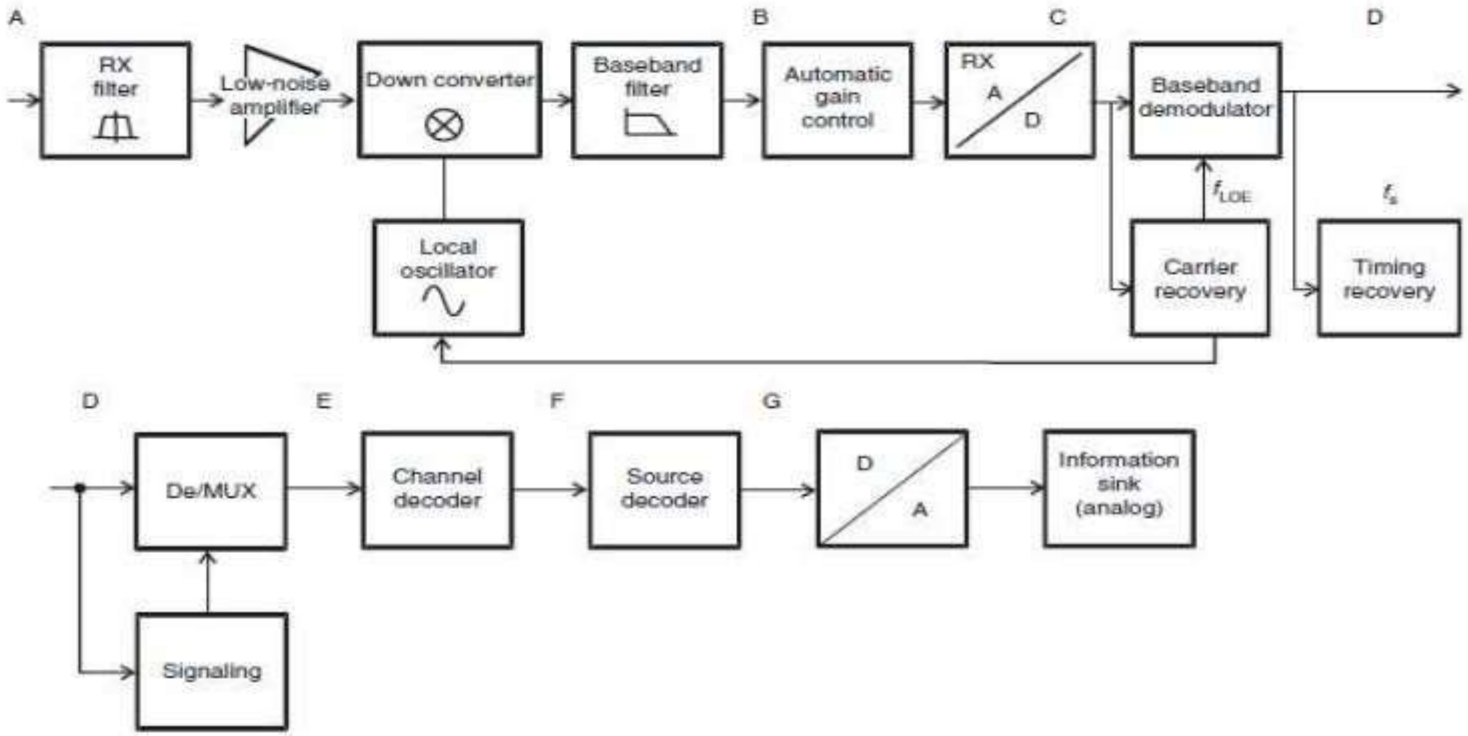


Fig: Block diagram of a digital receiver

Wireless communication link

The RX filter performs a rough selection of the received band.

- The low-noise amplifier amplifies the signal, so that the noise added by later components of the RX chain has less effect on the Signal-to-Noise Ratio (SNR).
- The RX LO provides sinusoidal signals corresponding to possible signals at the TX LO.
- The RX down converter converts the received signal (in one or several steps) into baseband.
- The RX low-pass filter provides a selection of desired frequency bands for one specific user. It eliminates adjacent channel interference as well as noise.
- The Automatic Gain Control (AGC) amplifies the signal such that its level is well adjusted to the quantization at the subsequent ADC.

Wireless communication link

- The RX ADC converts the analog signal into values that are discrete in time and amplitude.
- Carrier recovery determines the frequency and phase of the carrier of the received signal, and uses it to adjust the RX LO.
- The baseband demodulator obtains soft-decision data from digitized baseband data, and hands them over to the decoder.
- The decoder uses soft estimates from the demodulator to find the original (digital) source data
- The demultiplexer separates the user data and signaling information and reverses possible time compression of the TX multiplexer.

Quadrature Phase Shift Keying (QPSK)

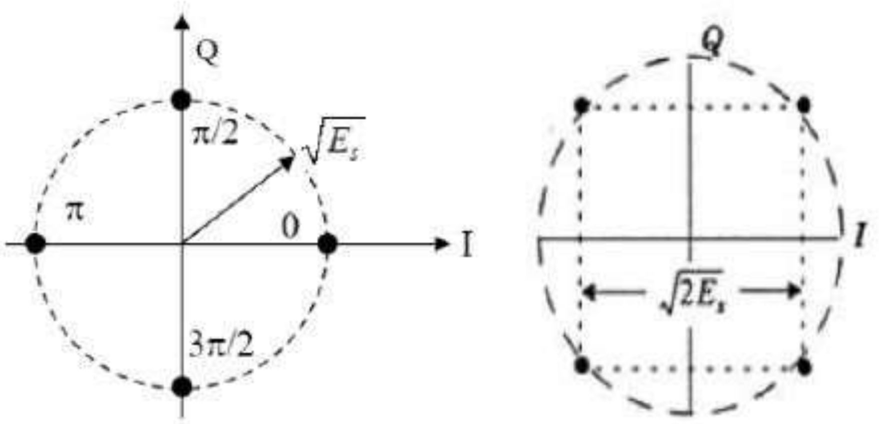
- Quadrature Phase Shift Keying (QPSK) has twice the bandwidth efficiency of BPSK, since 2 bits are transmitted in a single modulation symbol.
- The phase of the carrier takes on 1 of 4 equally spaced values, such as 0, π/2, π, 3π/2 [or] π/4, 3π/4, 5π/4,7π/4. where each value of phase corresponds to a unique pair of message bits.
- The QPSK signal for this set of symbol states may be defined as,

$$S_{QPSK}(t)=\sqrt{\frac{2E_S}{T_S}}\cos\left[2\pi f_c t+(i-1)\frac{\pi}{2}\right] \qquad \begin{array}{l} 0\le t\le T_S \\ i=1,2,3,4 \end{array}$$

- where Ts - symbol duration and is equal to twice the bit period

Quadrature Phase Shift Keying (QPSK)

- Constellation Diagram



(a) QPSK constellation where the carrier phases are 0, π/2, π, 3π/2;

(b) QPSK constellation where the carrier phases are π/4, 3π/4, 5π/4, 7π/4.

The average probability of bit error in the AWGN channel is obtained as

$$P_{e,\,\text{QPSK}} = Q\left(\sqrt{\frac{2E_b}{N_0}}\right)$$

Spectrum and Bandwidth of QPSK

$$P_{QPSK} = \frac{E_S}{2}\left[\left(\frac{\sin \pi(f - f_c)T_S}{\pi(f - f_c)T_S}\right)^2 + \left(\frac{\sin \pi(-f - f_c)T_S}{\pi(-f - f_c)T_S}\right)^2\right]$$

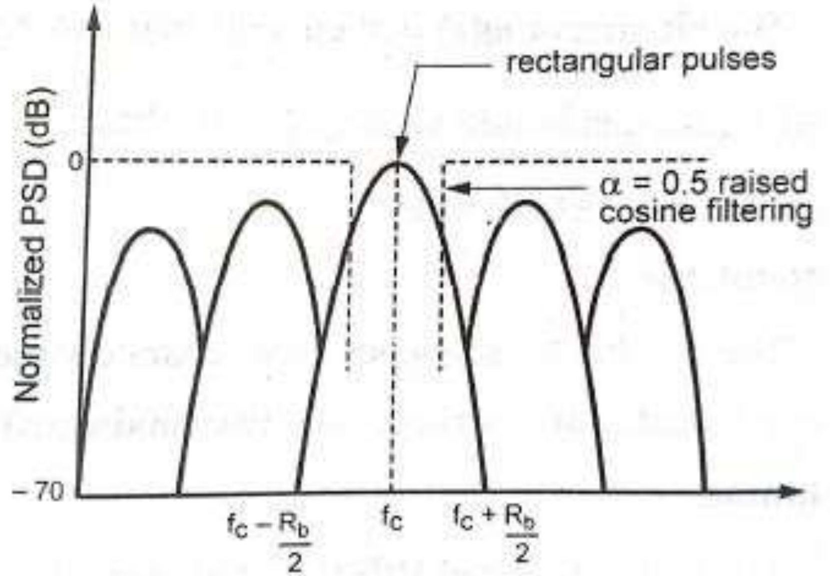


Fig: Power Spectral Density of QPSK

QPSK Transmitter

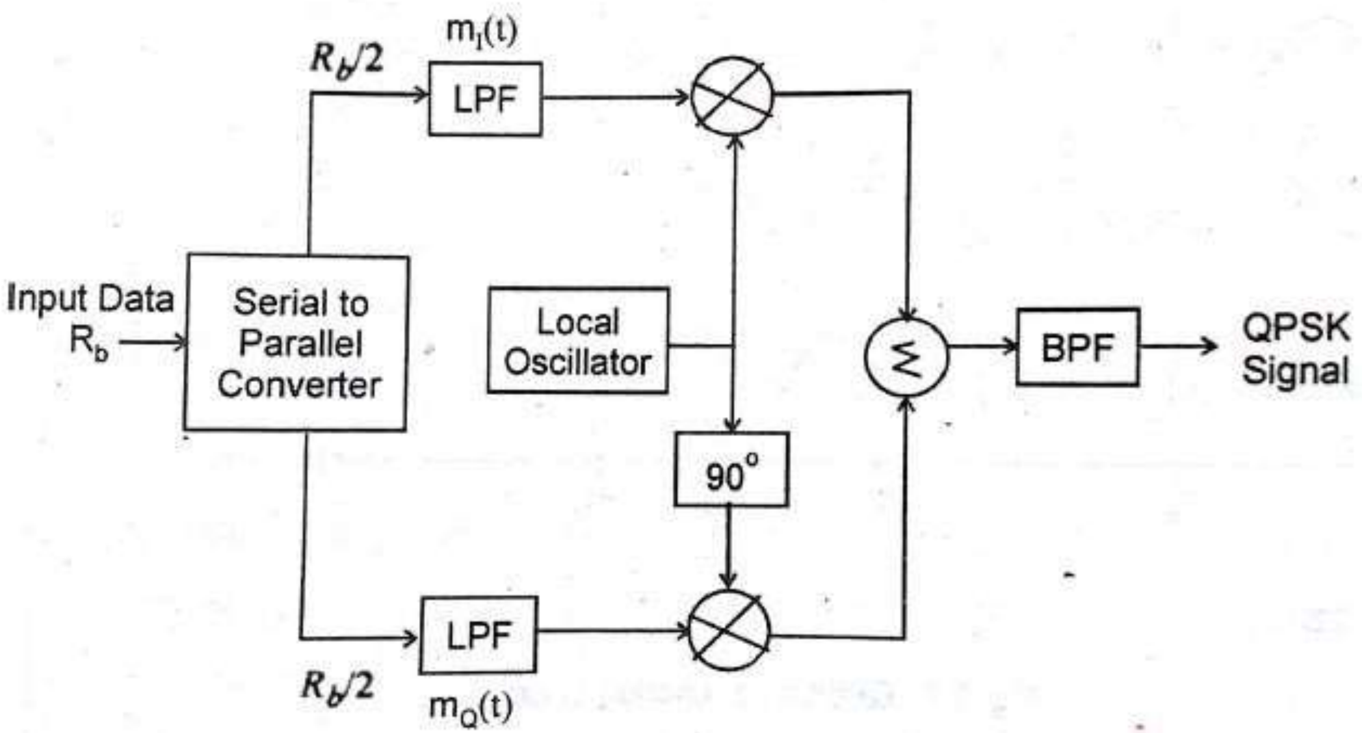


Fig. QPSK Transmitter

QPSK Transmitter

- The unipolar binary message stream has bit rate Rb and it is first converted into a bipolar NRZ sequence.
- The bitstream m(t) is then split into two bit streams in mI(t) and mQ (t) (in-phase and quadrature streams),each having a bit rate of Rs = Rb/2.
- The bit stream mI(t) is called the "even" stream and mQ (t) is called the "odd" stream.
- The two binary sequences are separately modulated by two carriers $\phi1(t)$ and $\phi2(t)$, which are in quadrature.
- The filter at the output of the modulator confines the power spectrum of the QPSK signal within the allocated band.

QPSK Receiver

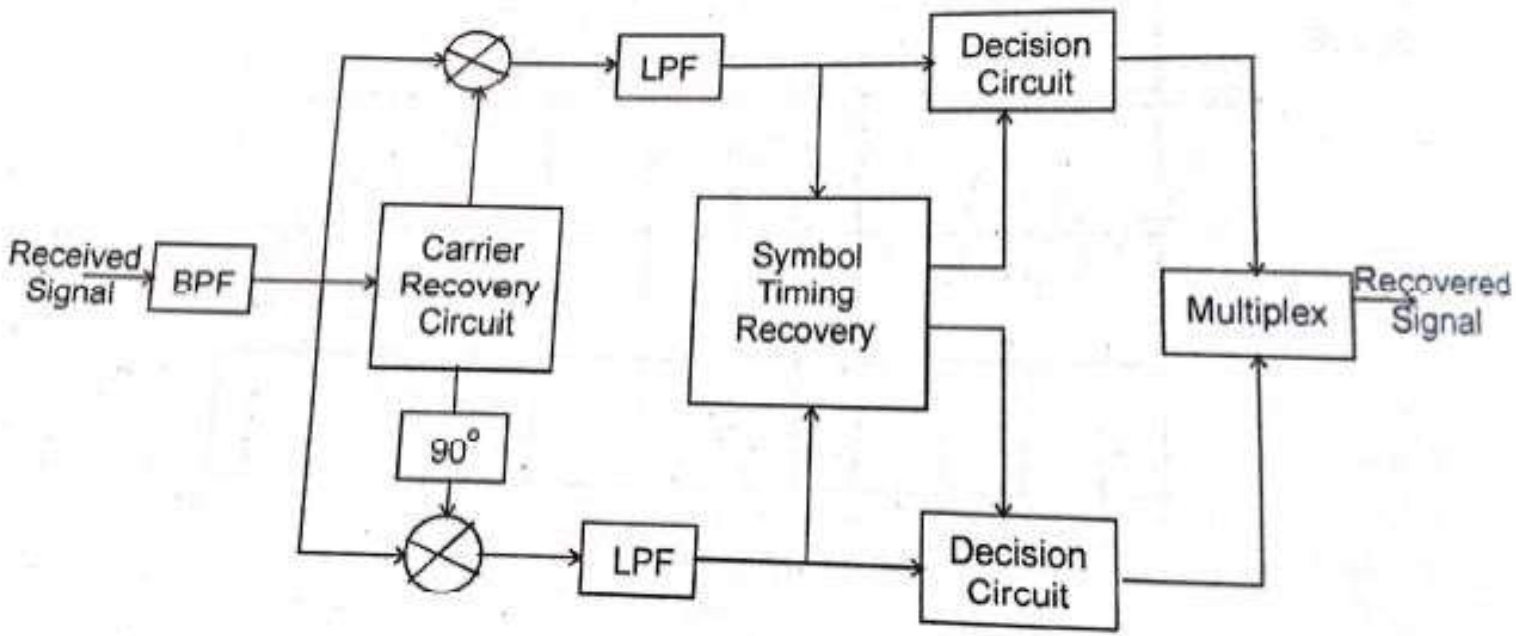


Fig. QPSK Receiver

QPSK Receiver

- The frontend bandpass filter removes the out-of-band noise and adjacent channel interference.
- The filtered output is split into two parts, and each part is coherently demodulated using the in-phase and
- quadrature carriers.
- The outputs of the demodulators are passed through decision circuits which generate the in-phase and
- quadrature binary streams.
- The two components are then multiplexed to reproduce the original binary sequence.

Offset QPSK(OQPSK)

- OQPSK signaling is similar to QPSK signaling, as represented by equation

$$S(t) = \sqrt{\frac{2E_s}{T_S}}\cos(2\pi f_c t)\cos[(i-1)\frac{\pi}{2}] - \sqrt{\frac{2E_s}{T_S}}\sin(2\pi f_c t)\sin[(i-1)\frac{\pi}{2}]$$

- In OQPSK, after splitting the even and odd bit streams, mI(t) and mQ(t), one bit stream is made offset by one bit period with each other.
- Phase transitions occur every Tb sec. maximum phase shift = ±90°. Does not cause the signal envelope to go to zero.
- Allows nonlinear amplification, so regeneration of side lobes are eliminated.
- OQPSK signal is identical to that of a QPSK signal. Hence, both signals same bandwidth.

Minimum Shift Keying (MSK)

- Minimum shift keying (MSK) is a special type of Continuous Phase Frequency Shift Keying (CPFSK)
- wherein the peak frequency deviation is equal to 1/4 the bit rate.
- In other words, MSK is continuous phase FSK with a modulation index of 0.5.
- The modulation index is defined as,

$$k_{FSK} = \frac{(2\Delta F)}{R_b}$$

Where,

ΔF - peak RF frequency deviation

Rb - bit rate.

Minimum Shift Keying (MSK)

MSK signal can be defined as,

$$S_{MSK}(t) = \sum_{i=0}^{N-1} m_I(t)P(t-2iT_b)\cos 2\pi f_c t + \sum_{i=0}^{N-1} m_Q(t)P(t-2iT_b-T_b)\sin 2\pi f_c t$$

$$\text{where} \quad P(t) = \begin{cases} \sin\left(\dfrac{\pi t}{2T_b}\right) & 0 \le t \le 2T_b \\ 0 & \text{elsewhere} \end{cases}$$

where mI(t) and mQ(t) are the "odd" and "even" bits of the bipolar data streams.

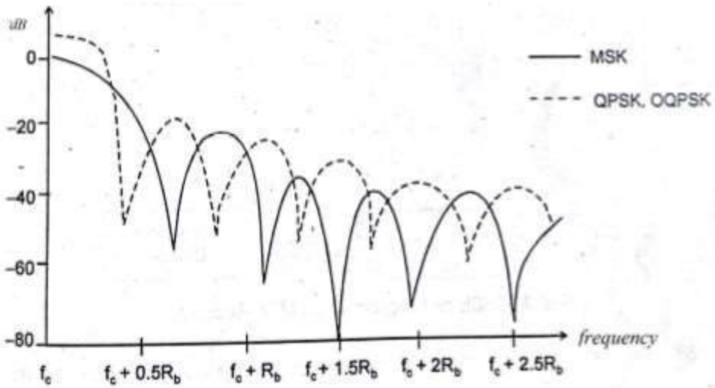
- MSK waveform can be seen as a special type of a continuous phase FSK and the above equation is rewritten as,

$$S_{MSK}(t) = \sqrt{\frac{2E_b}{T_b}}\cos\left[2\pi f_c t - m_I(t)m_Q(t)\frac{\pi t}{2T_b} + \phi_k\right]$$

MSK Power Spectrum

The normalized power spectral density for MSK is given by

$$P_{MSK} = \frac{16}{\pi^2}\left(\frac{\cos 2\pi(f+f_c)T}{1.16 f^2 T^2}\right)^2 + \frac{16}{\pi^2}\left(\frac{\cos 2\pi(f-f_c)T}{1.16 f^2 T^2}\right)^2$$



- From above Figure, it is seen that the MSK spectrum has lower side lobes than QPSK and OQPSK.
- Main lobe of MSK is wider than that of QPSK and OQPSK.
- MSK is less spectrally efficient than the phase-shift keying techniques

MSK-Transmitter

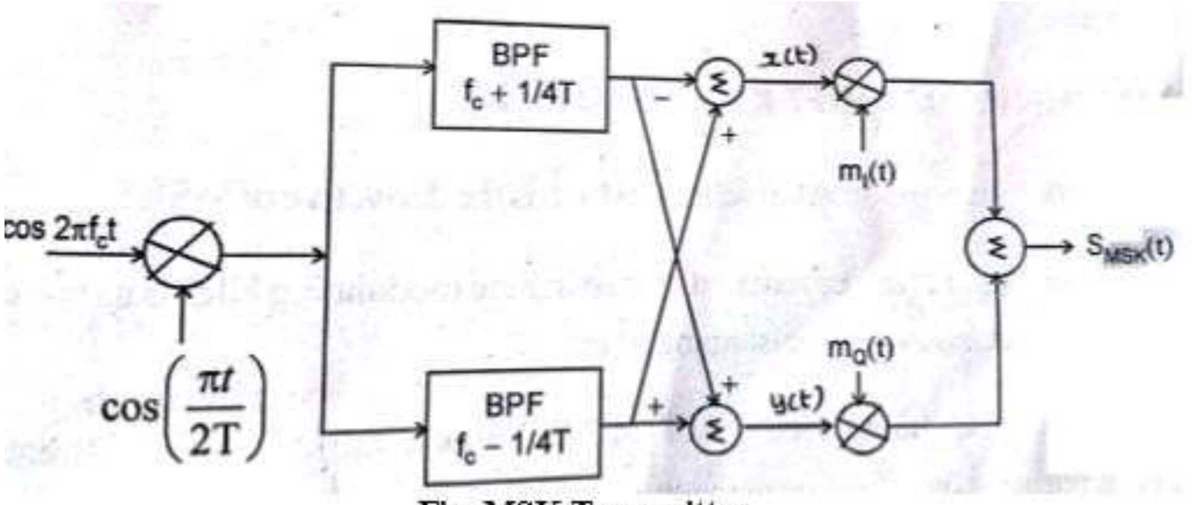


Fig: MSK Transmitter

- Multiplying a carrier signal with cos (πt/2T) produces two phase-coherent signals at fc+ 1/4T and fc- I/4T.
- These two FSK signals are separated using two narrow bandpass filters and appropriately combined to form the in-phase and quadrature carrier components x(t) and y(t), respectively.
- These carriers are multiplied with the odd and even bit streams, mI(t) and mQ(t), to produce the MSK modulated signal SMSK(t).

MSK-Receiver

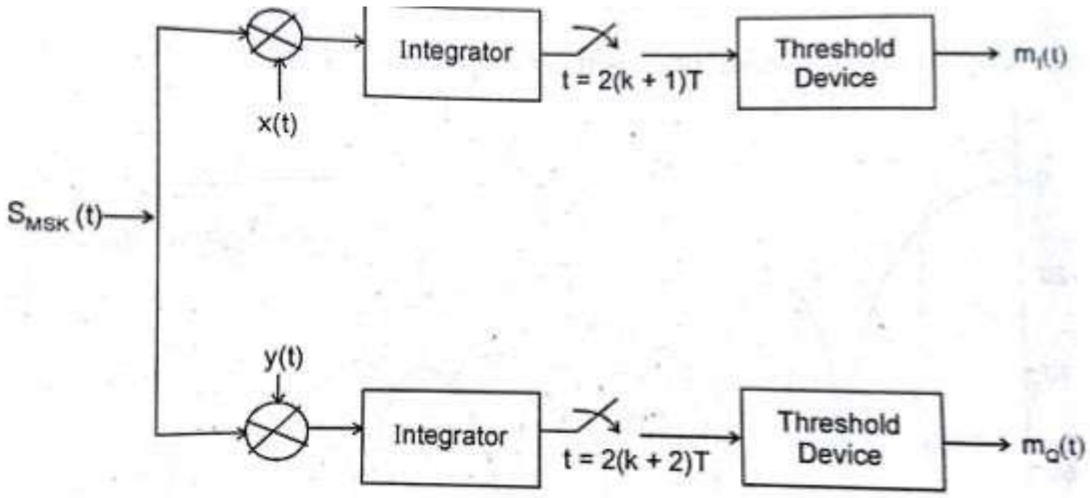


Fig: MSK Receiver

- The received signal SMSK(t) is multiplied by the respective in-phase and quadrature carriers x(t) and y(t).
- The output of the multipliers are integrated over two bit periods. Then, it is given to a decision circuit which has a threshold detector.
- Based on the level of the signal above or below the threshold, the threshold detector decides whether the signal is 0 or 1.

GMSK(Gaussian Minimum Shift Keying)

- It is a simple binary modulation scheme
- It is a derivative of MSK
- The **sidelobes level** of the spectrum are further **reduced** by passing the modulating waveform through a pre-modulation Gaussian pulse shaping filter
- Has excellent power and spectral efficiency due to constant envelope
- GMSK premodulation filter has an impulse response given by, $h_G(t) = \frac{\sqrt{\pi}}{\alpha}\exp\left(-\frac{\pi^2}{\alpha^2}t^2\right)$

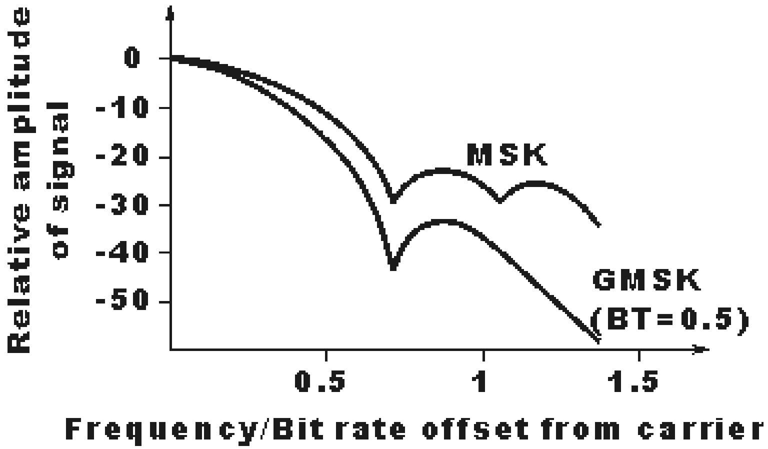
GMSK(Gaussian Minimum Shift Keying)

– Transfer function is given by, $H_G(f) = \exp(-\alpha^2 f^2)$

– The parameter α is related to B, the 3dB baseband bandwidth of $H_G(f)$ is

$$\alpha = \frac{\sqrt{\ln 2}}{\sqrt{2}B} = \frac{0.5887}{B}$$

– Power spectrum



Spectral density of MSK and GMSK signals

GMSK - Bit Error rate

- BER for GMSK was found for AWGN channels
- Bit error probability is a function of BT, since the pulse shaping impacts ISI
- Bit Error probability of GMSK is

$$P_e = Q\left\{\sqrt{\frac{2\gamma E_b}{N_0}}\right\}$$

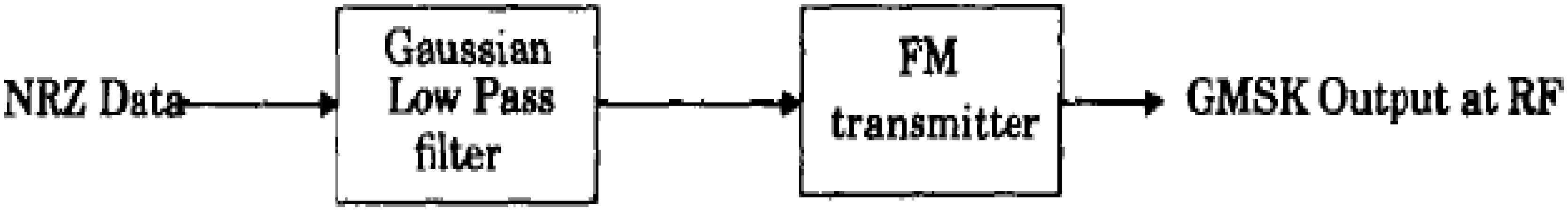
- Where γ is related to BT by

$$\gamma \cong \begin{cases} 0.68 & \text{for GMSK with } BT = 0.25 \\ 0.85 & \text{for simple MSK } (BT = \infty) \end{cases}$$

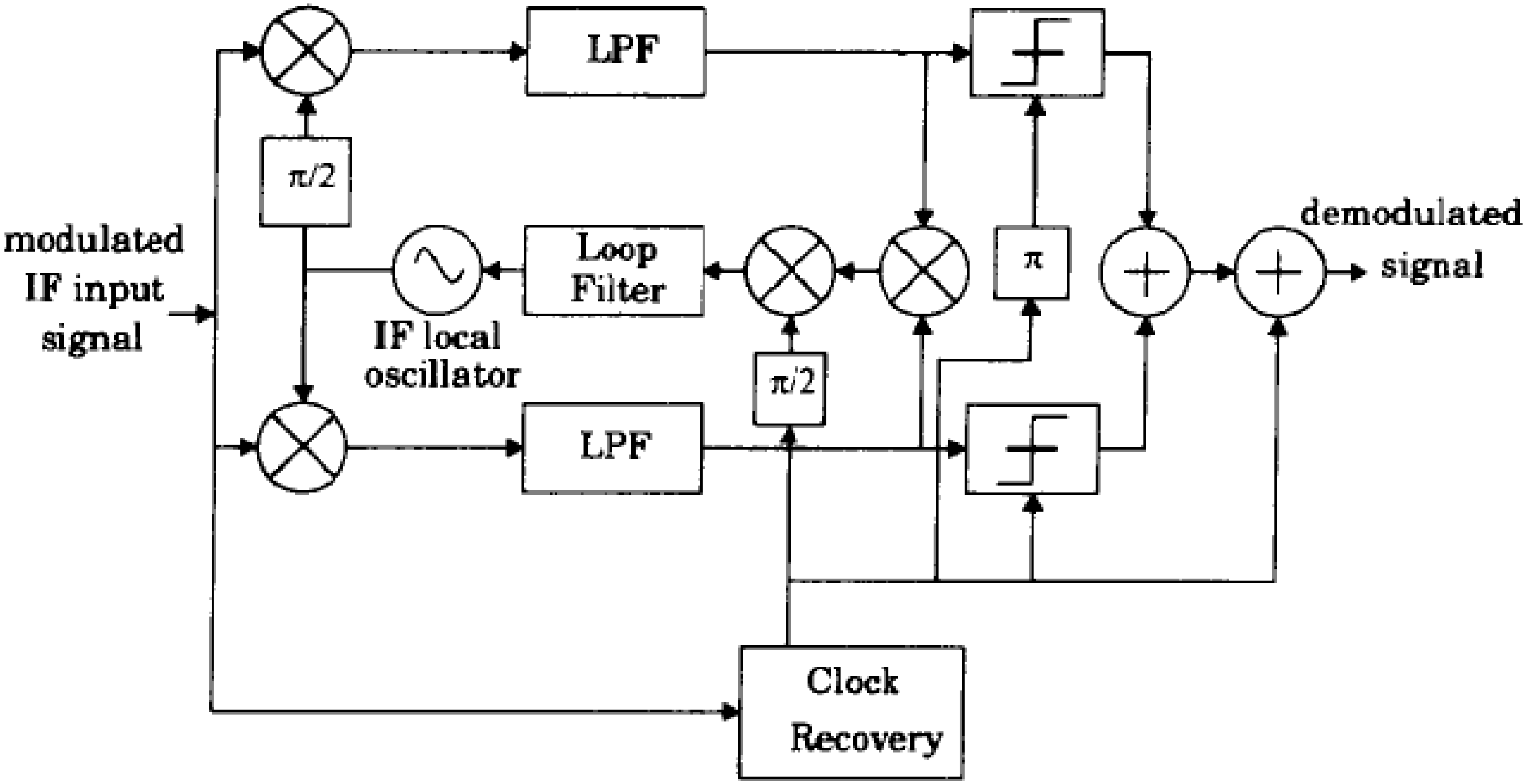
GMSK Transmitter

-The simplest way to generate a GMSK signal is to pass a message bit stream through a Gaussian baseband filter followed by an FM detector.

- Spectrum of LPF has Gaussian shape



GMSK Receiver



– GMSK signals can be detected using orthogonal coherent detectors or with simple noncoherent detectors

GMSK Receiver

- Carrier recovery is performed by using a method suggested by de Buda
- Sum of two discrete frequency components contained at the output of a frequency doubler is divided by four
- This type of receiver can be implemented using digital logic circuit
- Two D-Flip flops act as quadrature product demodulator and XOR gates act as baseband multipliers
- Mutually orthogonal reference carriers are generated using two D-Flip Flops

Error Performance of Modulation Techniques in Fading Channels

Error Probability in Flat-Fading Channels

– In fading channels, the received signal power (and thus the SNR) is not constant but changes as the fading of the channel changes

– To compute BER in such a channel, we have three steps:

1. Determine the BER for any arbitrary SNR

2. Determine the probability that a certain SNR occurs in the channel – in other words, determine the pdf of the power gain of the channel.

3. Average the BER over the distribution of SNRs

Error Performance of Modulation Techniques in Fading Channels

- In AWGN channel, the BER decreases exponentially as the SNR increases
- In contrast, a fading channel the BER decreases only linearly with the (average) SNR
- Sometimes fading leads to high SNRs, sometimes it leads to low SNRs, and it could be assumed that high and low values would compensate for each other.
- The relationship between (instantaneous) BER and (instantaneous) SNR is highly nonlinear

Error Performance of Modulation Techniques in Fading Channels

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OFDM(Orthogonal Frequency Division Multiplexing)

- Orthogonal Frequency Division Multiplexing (OFDM) is a modulation scheme that is especially suited for **high-data-rate** transmission in delay dispersive environments.
- It converts a high-rate data stream into a number of low-rate streams that are transmitted over parallel, narrowband channels that can be easily equalized.

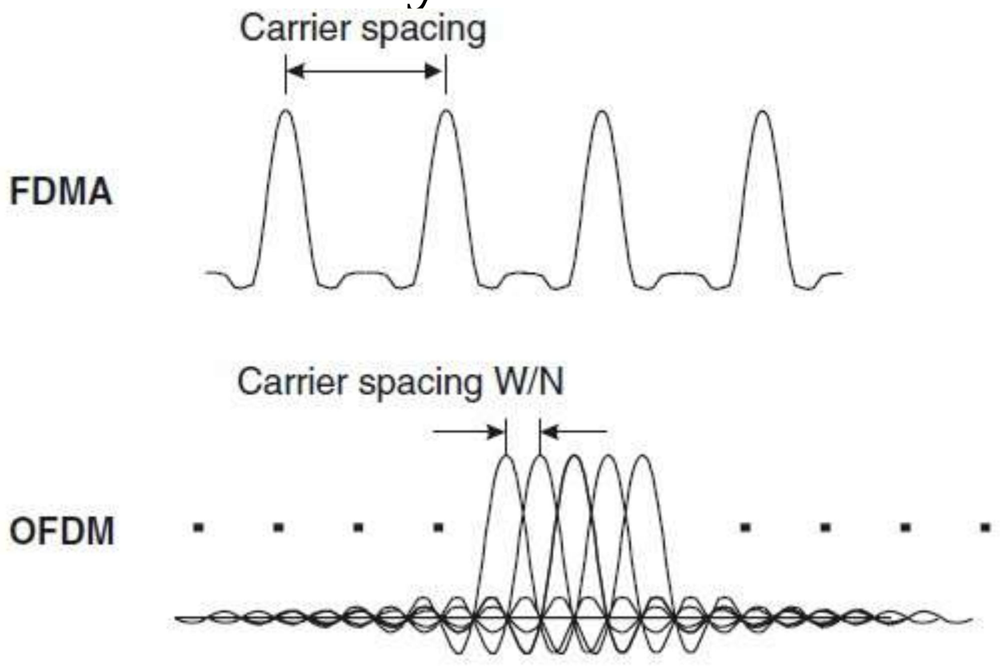
Principle of OFDM

- OFDM splits a high-rate data stream into N parallel streams, which are then transmitted by modulating N distinct carriers (called subcarriers or tones)
- let subcarriers be at the frequencies fn = nW/N, where n is an integer, W is the total available bandwidth
- subcarriers are mutually orthogonal,

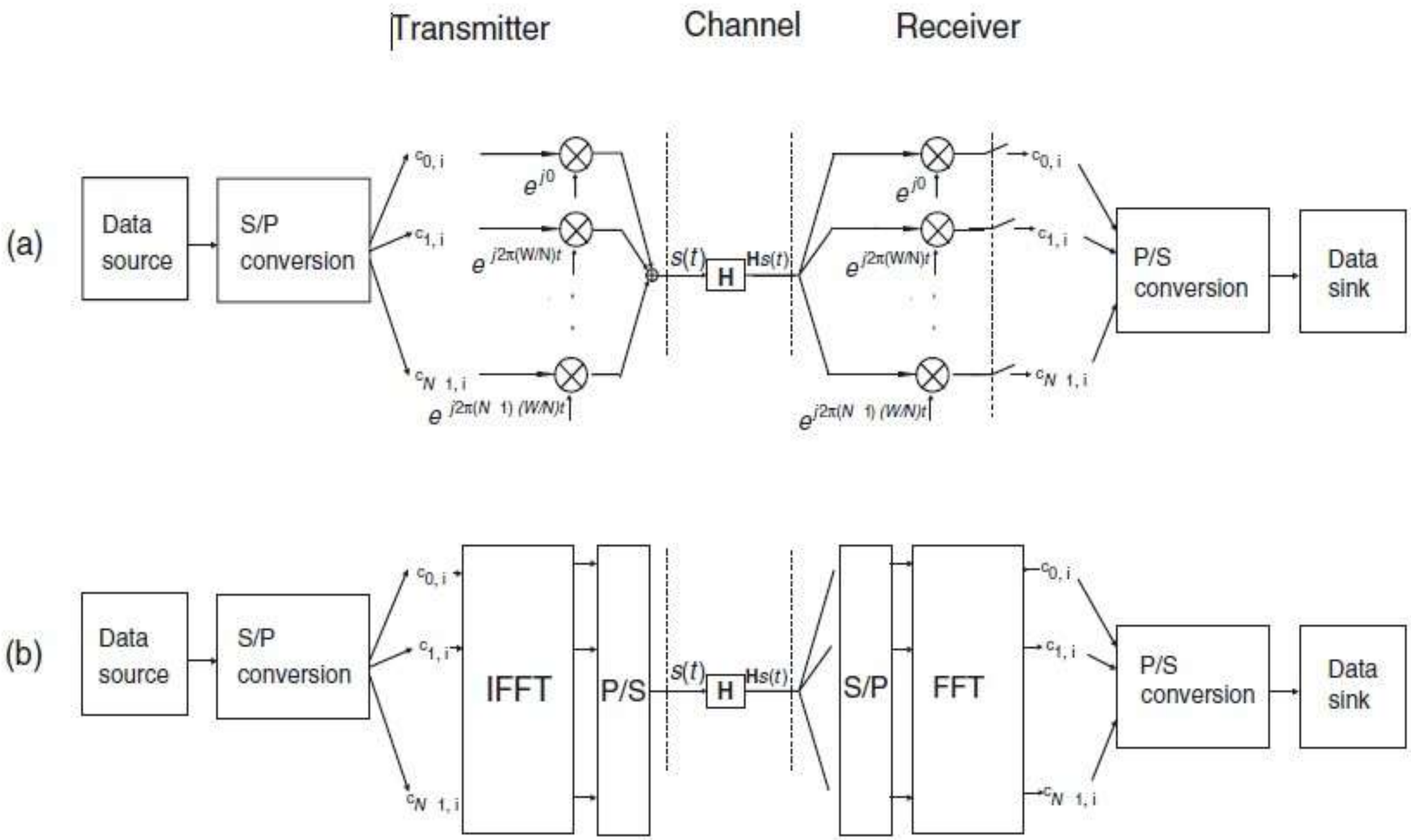
$$\int_{iTs}^{(i+1)TS} \exp(j2\pi fkt)\,\exp(-j2\pi fnt)\,dt = \delta nk$$

Principle of OFDM

- The spectra of different modulated carriers overlap, but each carrier is in the spectral nulls of all other carriers.
- The data streams of any two subcarriers will not interfere.



Implementation of transceivers



Implementation

- Let us first consider the analog interpretation. Let the complex transmit symbol at time instant i on the nth carrier be Cn,i . The transmit signal is then

$$s(t) = \sum_{i=-\infty}^{\infty} s_i(t) = \sum_{i=-\infty}^{\infty} \sum_{n=0}^{N-1} c_{n,i} g_n(t - iT_S)$$

where the basis pulse $g_n(t)$ is a normalized, frequency-shifted rectangular pulse:

$$g_n(t) = \begin{cases} \frac{1}{\sqrt{T_S}} \exp\left(j2\pi n \frac{t}{T_S}\right) & \text{for } 0 < t < T_S \\ 0 & \text{otherwise} \end{cases}$$

Implementation

- consider the signal for i = 0, sample it at instances $t_k = kT_s/N$

$$s_k = s(t_k) = \frac{1}{\sqrt{T_S}} \sum_{n=0}^{N-1} c_{n,0} \exp\left(j2\pi n \frac{k}{N}\right)$$

- The input to IFFT is made up of N samples (the symbols for the different subcarriers), and therefore the output from the IFFT also consists of N values.
- These N values have to be transmitted, one after the other, as temporal samples – this is the reason why we have a P/S (Parallel to Serial) conversion directly after the IFFT.

Implementation

- At the **receiver**, we can reverse the process: sample the received signal, write a block of N samples into a vector – i.e., an S/P (Serial to Parallel) conversion and perform an FFT on this vector.
- The result is an estimate $\tilde{c}_n$ of the original data Cn.
- OFDM can also be interpreted in the time–frequency plane.

Cyclic Prefix(CP)

- The OFDM transmitter and receiver work in an Additive White Gaussian Noise (AWGN) channel
- Which leads to **delay dispersion** also leads to a **loss of orthogonality** between the subcarriers, and thus to *Inter Carrier Interference* (ICI).
- Both these negative effects can be eliminated by a special type of guard interval, called the *cyclic prefix (CP)*.

Cyclic Prefix(CP)

- Base function for transmission

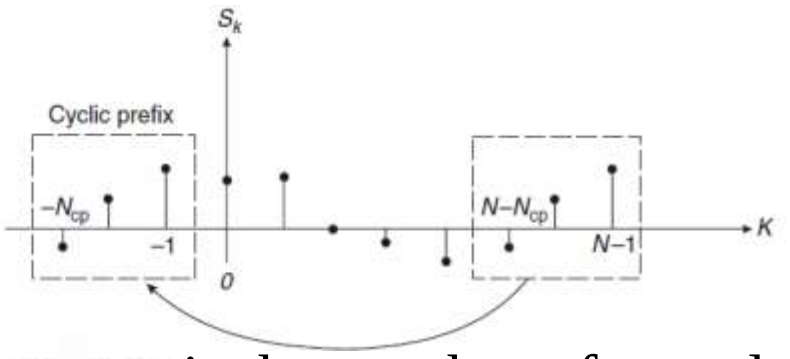
$$g_n(t) = \exp\left[j2\pi n\frac{W}{N}t\right] \quad \text{for} - T_{\text{cp}} < t < \hat{T}_{\text{S}}$$

- W/N is the carrier spacing,
- Ts = $\hat{T}_{\text{S}} + T_{\text{cp}}$.
- For duration $\hat{T}_{\text{S}} = N/W$., normal OFDM symbol is transmitted
- During time $0 < t < \hat{T}_{\text{S}}$ a copy of the last part of the symbol is $-T_{\text{cp}} < t < 0,$

•Total signal s(t) transmitted during time $-T_{\text{cp}} < t < 0,$ is a copy of s(t) during the last part

Cyclic Prefix(CP)

- Cyclic Prefix converts linear convolution into a cyclical convolution



- $N_{cp} = NT_{cp}/(N/W)$ is the number of samples in the cyclic prefix
- During the time period, the received signal suffers from Inter Symbol Interference (ISI), as echoes of the last part of the preceding symbol interfere with the desired symbol.
- ISI is eliminated by discarding the received signal during this time interval.

Peak to Average Power Ratio(PAPR)

- Drawback of OFDM
- When the signals of all the sub-carriers are added, the peak power can be the number of sub-carriers times the average power
- It is very high compared to the whole system
- The ratio of peak to average power value is called PAPR
- Addition of N signals of same phase produces a peak which is N times the average signal

Peak to Average Power Ratio(PAPR)

- OFDM signals have high PAPR.
- It is passed through High Power Amplifier at the transmitter
- Large PAPR – ADC (large dynamic range)
- Increase in complexity
- Efficiency is reduced

PAPR in OFDM systems

- OFDM signal consists of N data symbol
- Transmitted signal s(t),

$$x(t) = \frac{1}{\sqrt{N}} \sum_{n=0}^{N-1} X_n \cdot e^{j2\pi n \Delta f t}, 0 \le t \le T$$

- The PAPR of the transmit signal is defined as ratio between instantaneous power to its average power during the OFDM symbol period

$$PAPR = \frac{\max\limits_{0 \le t \le T} |x(t)|^2}{1/T \cdot \int_0^T |x(t)|^2 \, dt}$$

PAPR in OFDM systems

- PAPR reduction is achieved by minimizing the maximum instantaneous signal power
- Maximum PAPR can be expressed as

$$\text{MAX PAPR}[x(t)] = N$$

- PAPR increases with the number of sub-carriers
- Best way is to reduce the number of sub-carriers

PAPR Reduction Techniques

- **Coding for PAPR reduction** – PAPR which is lower than the given threshold is acceptable
- **Phase Adjustments-** Phase adjustments vector are known to both transmitter and receiver

  OFDM symbol x Phase vector = Transmitter signal
- **Correction by multiplicative function-** OFDM signal is multiplied by a Gaussian function which reduces PAPR
- **Correction by Additive function** – Out of band inference is reduced, but BER is increased

UNIT - 4

MULTIPATH MITIGATION TECHNIQUES

Equalisation – Adaptive equalization, Linear and Non-Linear equalization, Zero forcing and LMS Algorithms. Diversity – Micro and Macro diversity, Diversity combining techniques, Error probability in fading channels with diversity reception, Rake receiver

Introduction

- Different techniques can improve link performance without altering air interface
  increasing transmit power or bandwidth

1. Equalization used to counter ISI (time dispersion)

2.Diversity used to reduce depth & duration of fades due to motion

3.Channel Coding: coded bits improve small-scale link performance

Equalization

Inter symbol Interference (ISI) is caused by multipath time delay spread

- results in signal distortion
- occurs in time dispersive, frequency selective fading (bandlimited) channels

Equalization is a method of overcoming ISI

- **adaptive equalizers** can cancel interference
  - mobile fading channel is random & time varying
  - adaptive equalizers **track time varying** channel characteristics
  - Adapts to channel variations

Adaptive Equalizers

Two Operating Modes

(1) training

(2) tracking

(1) training adaptive equalizer

i. send training sequence of known fixed-length & bit pattern

- typically a pseudo random binary signal
- designed to permit acquisition of filter coefficients in worst case

ii. receiver's equalizer recovers training sequence

- adapts settings to minimize BER

iii. convergence: training obtains near optimal filter coefficients

Adaptive Equalizers

(1) tracking adaptive equalizer

Continually track and adjust filter coefficients as data is received

- adjustments compensate for time-varying channel
- data can be encoded (channel coded) for better performance

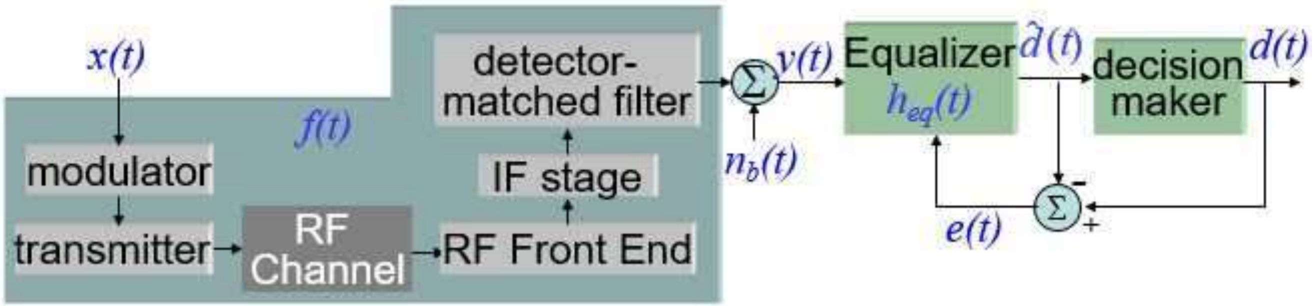
At the receiver

- recursive algorithm evaluates channel & estimates filter coefficients
- filter compensates for multipath in the channel

Adaptive Equalizer

Communication System with Adaptive Equalizer

$f(t)$	combined complex baseband impulse response of *transmitter, channel,* & *receiver's RF/IF sections*
$h_{eq}(t)$	impulse response of equalizer
$x(t)$	initial base band signal
$y(t)$	input to equalizer
$n_b(t)$	baseband noise at equalizer input
$e(t)$	equalizer prediction error
$d(t)$	reconstructed data
$\hat{d}(t)$	equalizer output



Adaptive Equalizer

Equalizer **input:** $y(t) = x(t) * f^*(t) + n_b(t)$

Equalizer **output:**

$$\hat{d}(t) = x(t) * f^*(t) * h_{eq}(t) + n_b(t) * h_{eq}(t)$$

$$= x(t) * g(t) + n_b(t) * h_{eq}(t)$$

where $g(t)$ = combined response of $f(t)$ & $h_{eq}(t)$

assume $n_b(t) = 0$ →

goal: force $\hat{d}(t)$ to equal $x(t)$, the desired output

this **requires** $g(t)$ to be: $g(t) = f^*(t) * h_{eq}(t) = \delta(t)$

complex baseband impulse response of transversal filter equalizer given by:

$$h_{eq}(t) = \sum_n c_n \delta(t - nT)$$

c_n = complex coefficients of equalizer

Adaptive Equalizer

Frequency Domain expression of

$$g(t) = f^*(t) * h_{eq}(t) = \delta(t) \qquad 1$$

$$H_{eq}(f)F^*(-f) = 1 \qquad 2$$

Equalizers **Goal** is to satisfy eq. 1 and 2

- makes combined *transmitter, channel, receiver* ≈ all-pass channel

Thus ideal equalizer is inverse filter of channel transfer function

- provides flat composite received response with linear phase response
- for time varying channel → eq.2 should be satisfied
- for frequency selective channel it is required to
  - amplify frequency components with small amplitudes
  - attenuate frequency components with large amplitudes

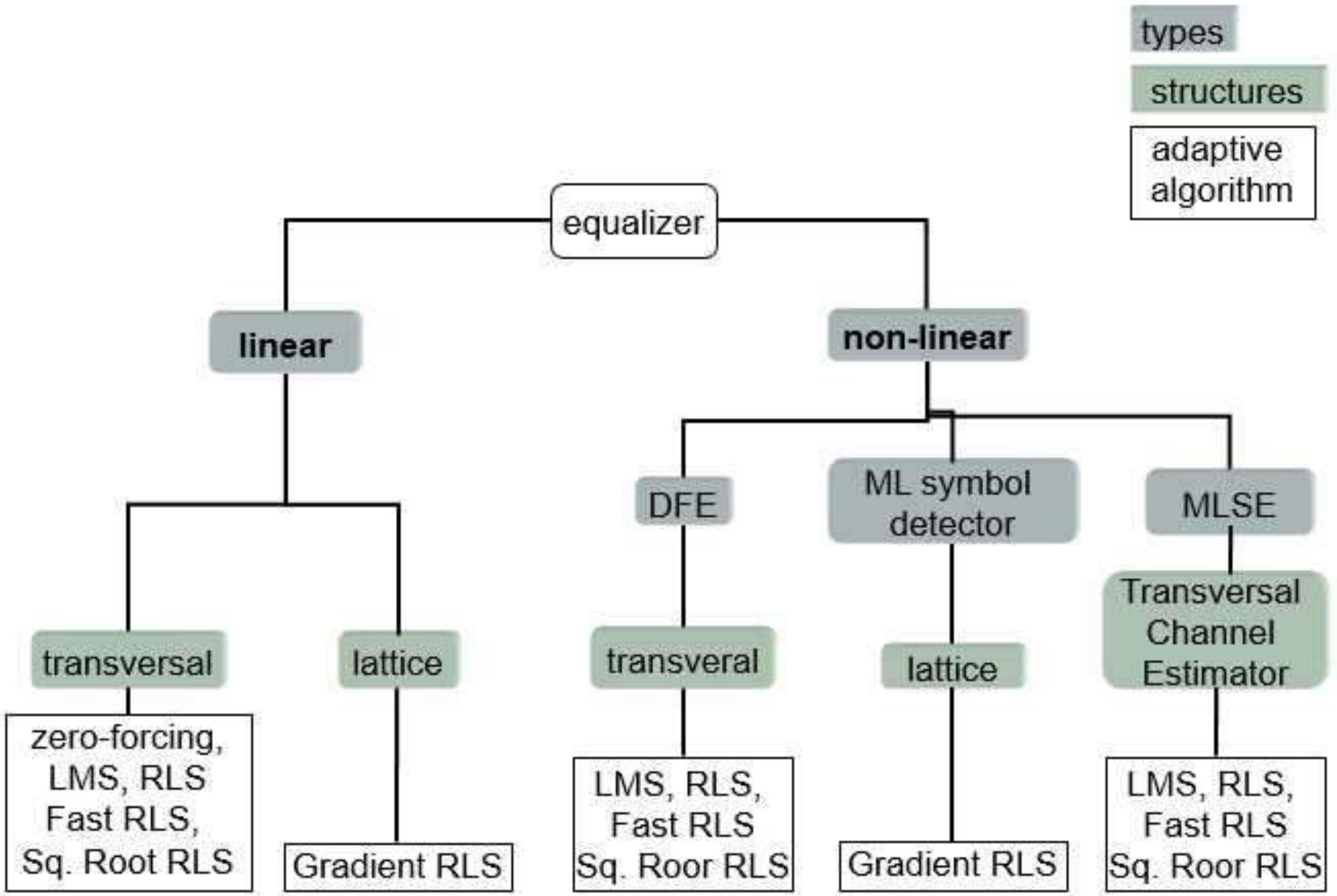
Classification of Equalizers

linear equalizer: If the output *d(t)* is not used in feedback path to adapt the equalizers

non-linear equalizer:

If the output d(t) is used in feedback path

Classification of Equalizers



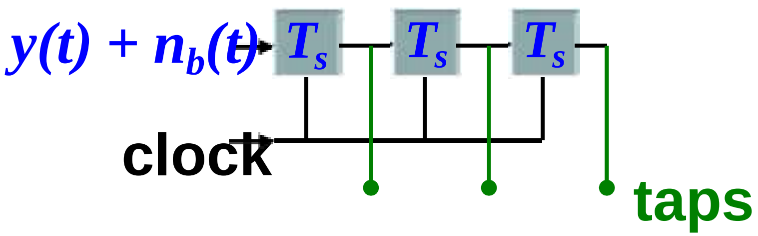
Linear Transversal Equalizer

most common equalizer structure

- tapped delay lines spaced at symbol period, T_s
- delay element's **transfer function** given by z^{-1} or $exp(-jwT_s)$
- delay elements have **unity gain**

(i) Finite Impulse Response (FIR) filter is the simplest LTE

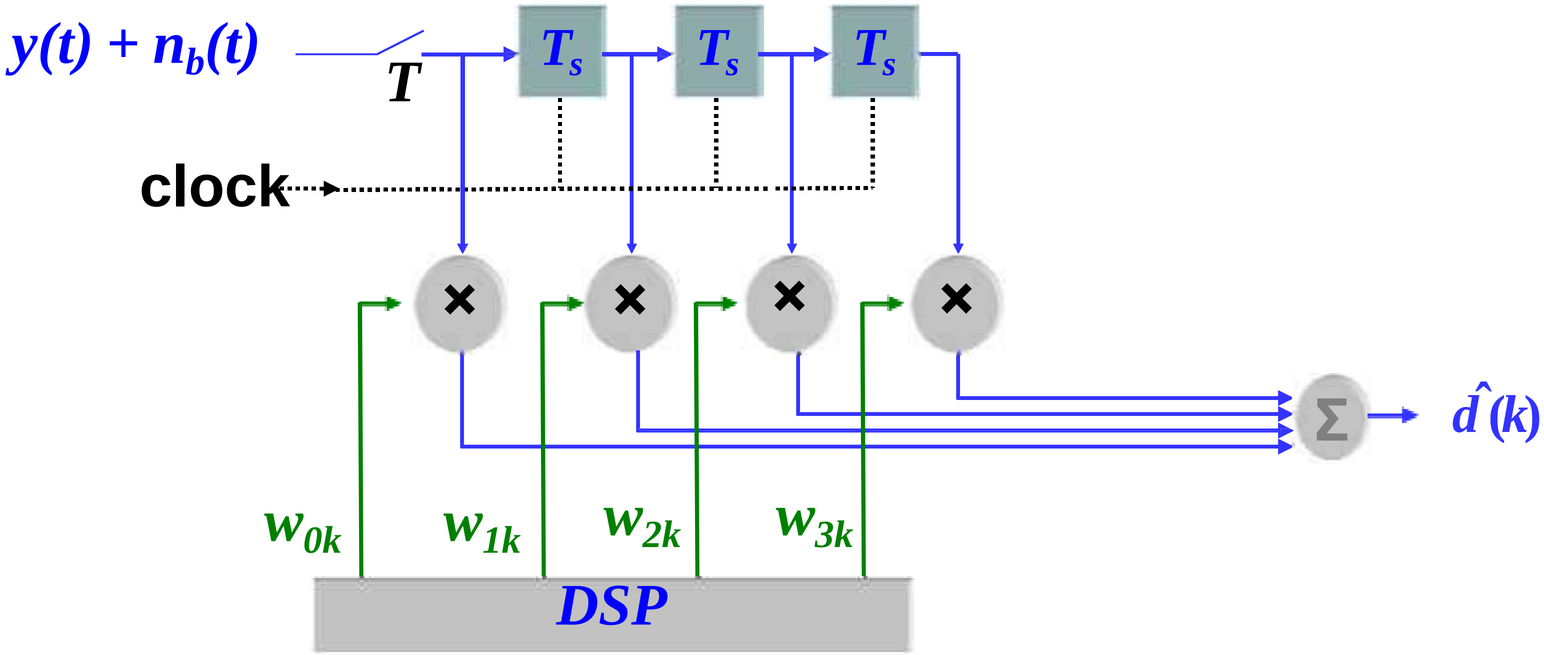
- only **feed-forward** taps
- transfer function = polynomial in z^{-1}
- many zeros
- poles only at $z = 0$



Basic FIR LTE Structure

Linear Transversal Equalizer

- **FIR LTE** Structure showing weights



$$\hat{d}(k) = \sum_{k=0}^{N-1} w_{ik}\big(y(k-i) + n_b(k-i)\big)$$

Lattice Filter

Input signal $y(k)$ is transformed into N intermediate signals

classified as

i. $f_n(k)$ slangis rorre drawrof =

ii. $b_n(k)$ slangis rorre drawkcab =

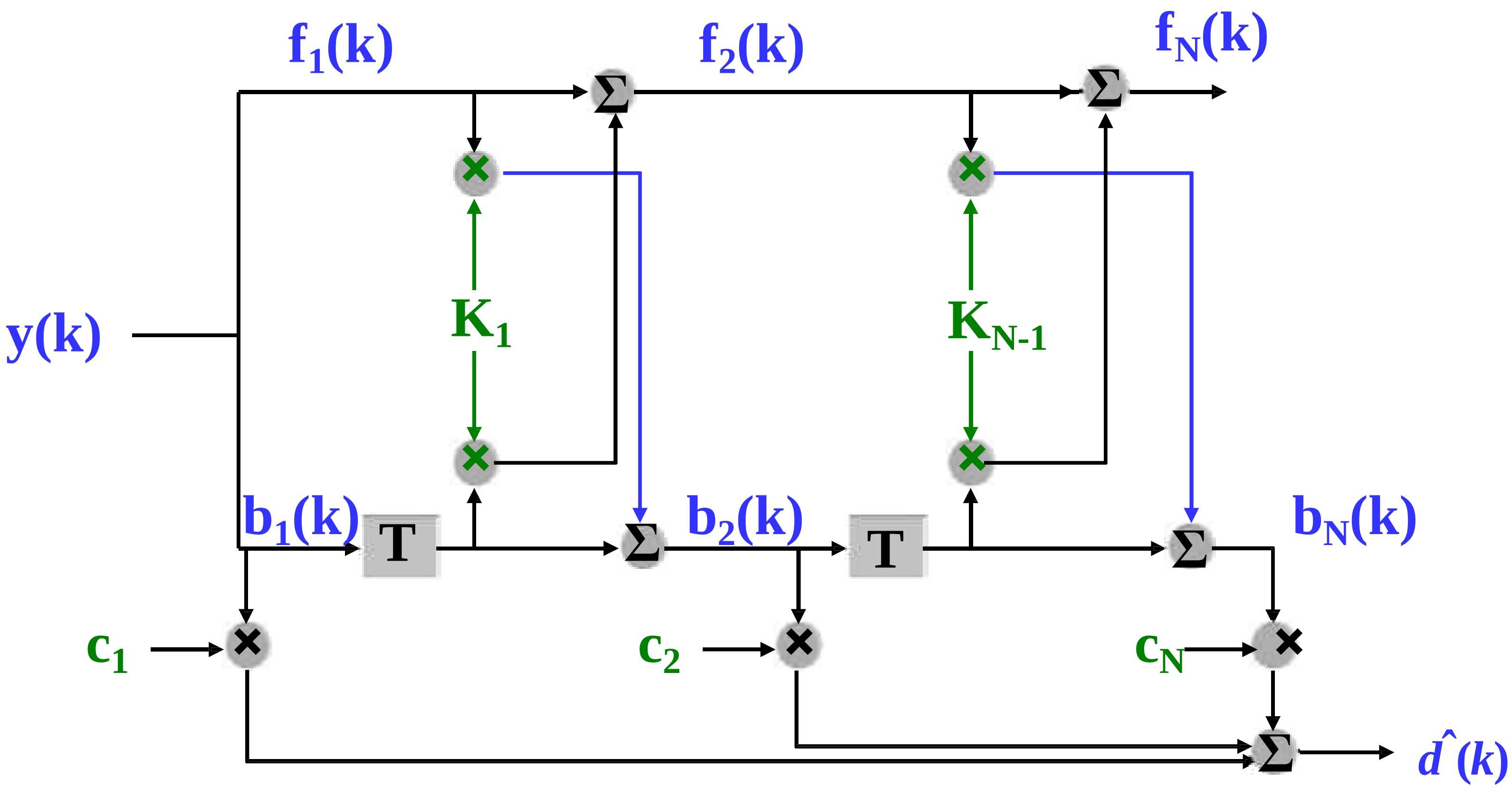
intermediate signals are used:

- as input into tap multipliers
- to calculate & update coefficients

each stage characterized by recursive equations for $f_n(k)$, $b_n(k)$

Lattice Filter

– Structure



Lattice Filter

– Evaluation of lattice equalizer

$$y(k) = f_1(k) = b_1(k)$$

$$f_n(k) = y(k) - \sum_{i=1}^{n} K_i y(k-i)$$

$$= f_{n-1}(k) + K_{n-1}(k) b_{n-1}(k-1)$$

$$b_n(k) = y(k-n) - \sum_{i=1}^{n} K_i y(k-n+i)$$

$$= b_{n-1}(k) + K_{n-1}(k) f_{n-1}(k-1)$$

- $K_n(k)$ = reflection coefficient for n^{th} stage of lattice
- b_n = backward error signals used as inputs to tap weights

equalizer output given by $\hat{d}(k) = \sum_{n=1}^{N} c_n(k) b_n(k)$

Non-Linear Equalization

common in **practical** wireless systems

- used with severe channel distortion
- Used when ISI is severe
- **linear equalizers** don't perform well in channels with deep spectral nulls
- tries to compensate for distortion
- high gain enhances noise at spectral null 3

effective non-linear methods used

1. **Decision Feedback Equalization** (DFE)
2. **Maximum Likelihood Symbol Detection** (MLSD)
3. **Maximum Likelihood Sequence Estimation** (MLSE)

Decision Feedback Equalization (DFE)

basic idea:

i. **detect** information symbol and pass it through the decision device

ii. after detection - **estimate ISI** induced

iii. subtract estimated **ISI** from detection of future symbols

Direct Transversal Form of DFE consists of FFF and FBF filter

- FFF = feed forward filter with N_1+N_2+1 taps
- FBF = feedback filter with N_3 taps

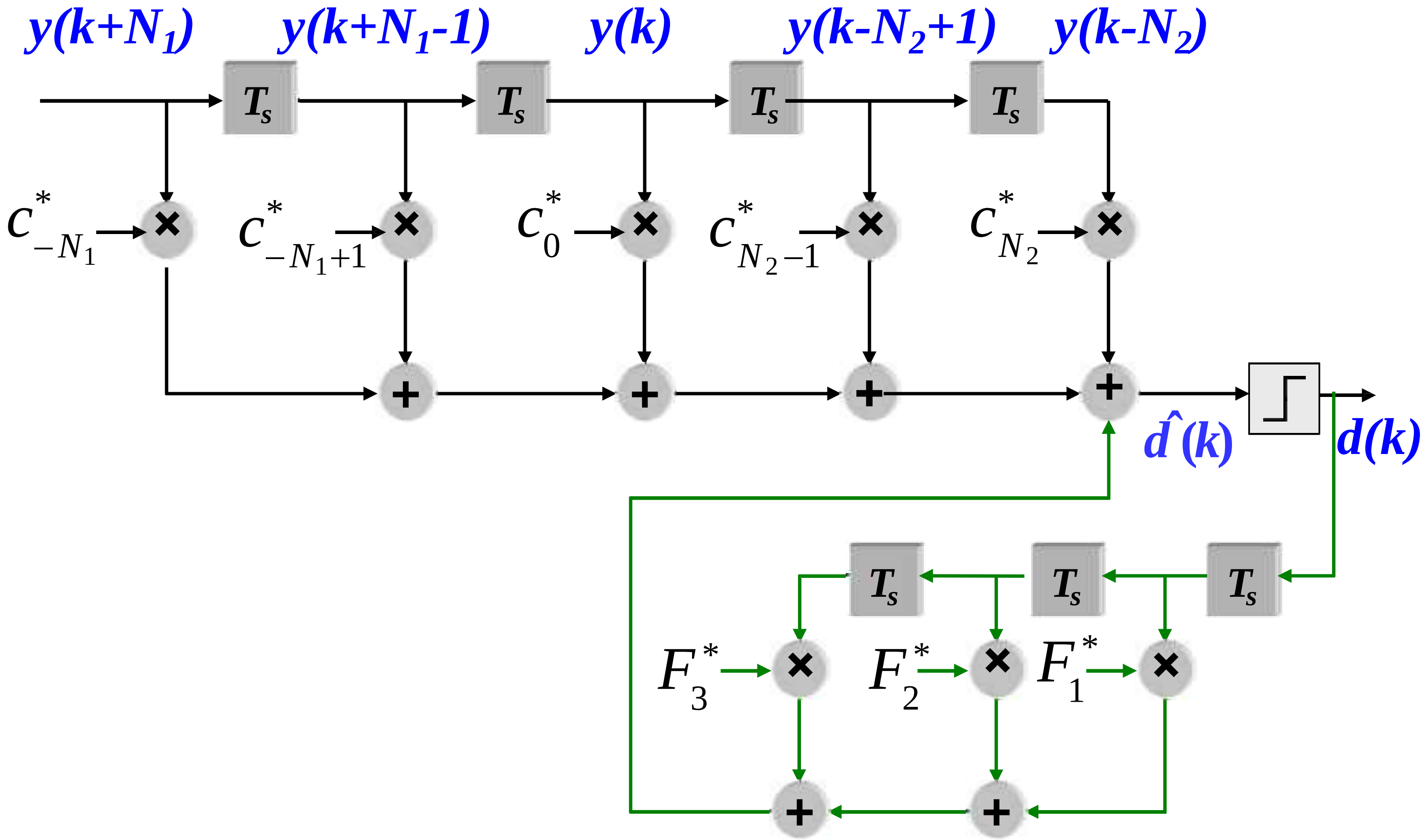
- driven by detector's output (decision threshold)

- filter coefficients adjusted based on past detected symbol

- goal is to cancel ISI on current symbol

Decision Feedback Equalization (DFE)

- $y(k - N +_2 i($ are past equalizer inputs used for **FFF** segment
- $y(k + N -_1 i)$ are delayed equalizer inputs for **FFF** segment



Decision Feedback Equalization (DFE)

DFE output given by:

$$\hat{d}(k) = \sum_{n=-N_1}^{N_2} c_n^*(k)\,y(k-n) + \sum_{i=1}^{N_3} F_i\,d(k-i)$$

d(k-i) = previous decision outputs

y (k-n) = equalizer inputs, including past, current, and delayed

Minimum MSE for Direct Transversal DFE given by ξDFE

$$\boldsymbol{\xi_{DFE}} = \exp\left\{\frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} \ln\left(\frac{N_0}{\left|\mathbf{F}\left[e^{jwT}\right]\right|^2 + N_0}\right) dw\right\}$$

Predictive DFE

consists of FFF and FBF

FBF is driven by the input sequence given by

$d(k)$ – FFF output

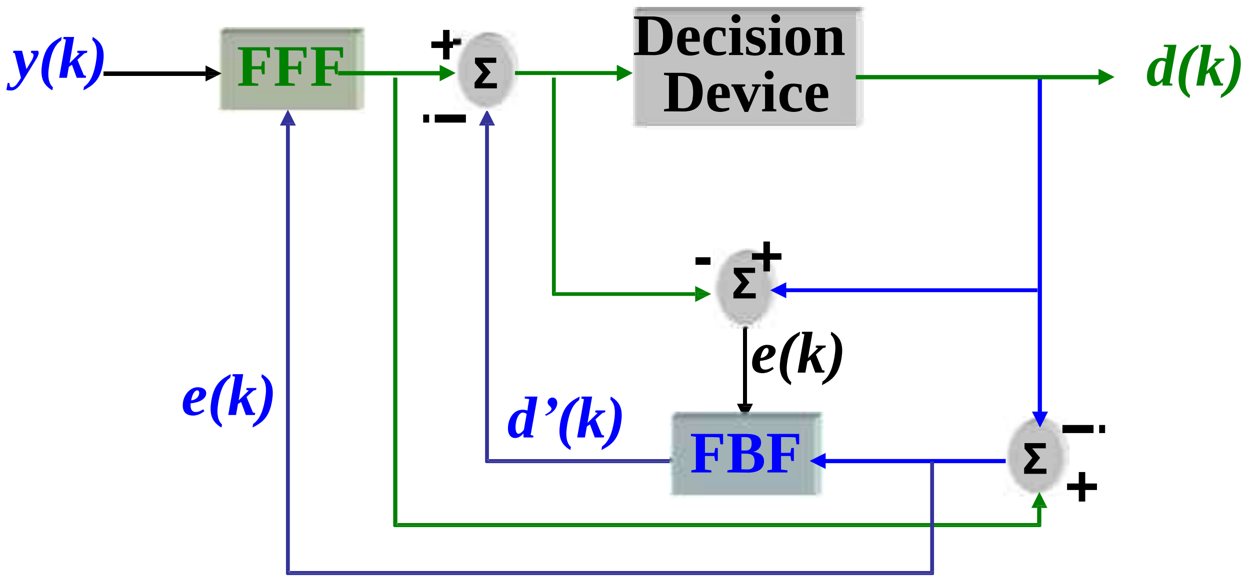
FBF is called noise & ISI predictor

- predicts noise & residual ISI contained in signal at FFF output
- subtracts this from $d(k)$ after some feedback delay

Predictive DFE performs well as conventional DFE as number of taps in FFF and FBF approach infinity

Predictive Equalizer can also be realized as lattice equalizer

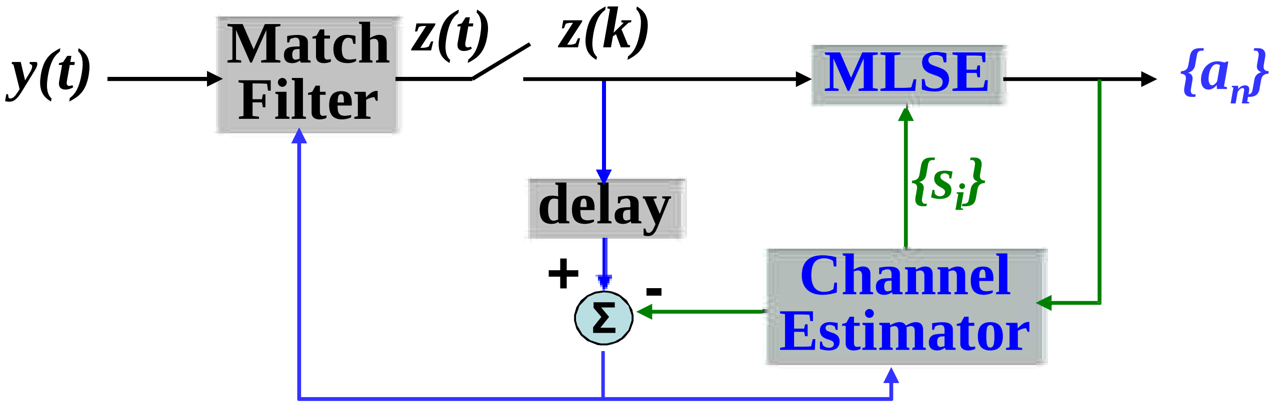
Predictive DFE



y(k) = kth received signal sample
d(k) = output decision
e(k) = error signal
d'(k) = output prediction

Maximum Likelihood Sequence Estimation Equalizer (MLS)

- optimal in the sense that it minimizes probability of sequence error
- requires knowledge of
  i. channel characteristics to compute metrics for making decision
  ii. statistical distribution of **noise** corrupting the signal



Algorithms for Adaptive Equalization

- A wide range of algorithms exist to compensate for unknown & time-varying channel
- algorithms are used to
  i. update equalizer coefficients
  ii. track channel variations
- 3 basic algorithms for adaptive equalization
  (i) zero-forcing (ZF)
  (ii) least mean square (LMS)
  (iii) recursive least squares (RLS)

Performance Measure of Algorithms

(1) Rate of Convergence (RoC)

(2) Misadjustment

(3) Computational Complexity

(4) Numerical Properties & Inaccuracies

(5) practical cost & power issues

(6) radio channel characteristics

(1) Rate of Convergence (RoC): iterations needed for converge to optimal solution in response to stationary inputs

- **fast RoC** allows rapid adaptation to stationary environment of unknown statistics

2) Misadjustment: quantitative measure of the amount by which the final value of mean square error, averaged over adaptive filters

- Deviations from minimum mean square error

$$\xi_{obtained} - \xi_{minimum}$$

Performance Measure of Algorithms

(3) Computational Complexity: number of operations required to complete iteration of the algorithm

(4) Numerical Properties: inaccuracies produced by round-off noise

- representation errors in digital format (floating point) in computers
- errors can influence stability of the algorithm

5) Practical Issues in the choice of equalizer structure & algorithms

- equalizers must justify cost, including relative cost of computing platform
- power budget-battery drain: radio propagation characteristics

Zero Forcing (ZF) Algorithm

- assume **tapped delay line filter** with N taps delayed by T & weights c_n's
- c_n's selected to force samples of $h_{ch}(t)*h_{eq}(t)$ to 0
  - N sample points each delayed by T_s (symbol duration)

  let n increase without bound → obtain infinite length equalizer with

  zero ISI at ouput

The equalizer coefficients are chosen to force the samples of the combined channel and equalizer impulse response to zero

Nyquist Criterion must be satisfied by combined channel response

$$H_{ch}(f)H_{eq}(f) = 1, \quad |f| < 1/2T_s$$

$H_{eq}(f)$ = frequency response of equalizer that is periodic with $1/T_s$

$H_{ch}(f)$ = folded channel frequency response

Least Mean Square (LMS)

LMS seeks to compute minimum mean square error, ξ_{min}

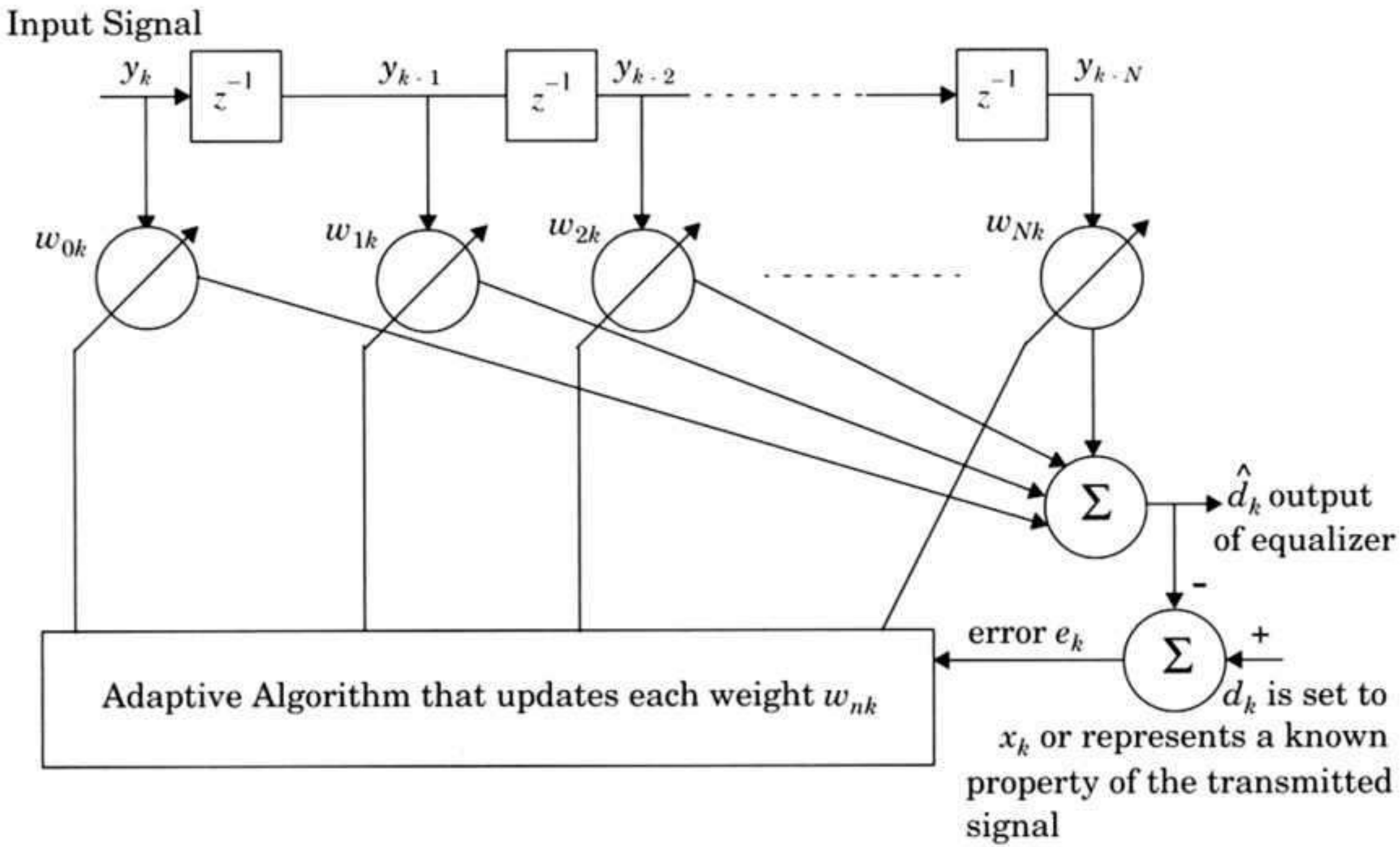
- more robust than **ZF** algorithm for adaptive equalizers
- **criterion** = minimization of MSE between desired and actual equalizer outputs
- *e(k)* = prediction error given by

$$\boldsymbol{e(k) = d(k) -}\ \hat{d}(k)$$
$$\boldsymbol{= x(k) -}\ \hat{d}(k)$$

x(k) = original transmitted baseband signal

d(k) = *x(k)* → known training sequence transmitted

Least Mean Square (LMS)



Least Mean Square (LMS)

equalizer output given by

$$\hat{d}(k) = y(k)w(k) + y(k\text{-}1)w(k\text{-}1) + \ldots + y(k\text{-}n)w(k\text{-}n)$$

- $\mathbf{w}_N$= tap gain vector

then **error** is given by

$$e(k) = x(k) - \mathbf{y}_k^T \mathbf{w}_k$$
$$= x(k) - \mathbf{y}_k \mathbf{w}_k^T$$

mean square error at time k given by

$$\xi = \mathbf{E}[e^*(k)e(k)]$$

- for specific channel condition→ $e(k)$ depends on tap gain vector $\mathbf{w}_N$
- thus ξ is function of $\mathbf{w}_N$

Least Mean Square (LMS)

cost function gives ξ as function of $\mathbf{w}_N$

$$\xi = J(\mathbf{w}_N)$$

from section → computing ξ_{min} requires

$$\frac{\partial J(\mathbf{w}_N)}{\partial \mathbf{w}_N} = -2\mathbf{p}_N + 2\mathbf{R}_{NN}\mathbf{w}_N = 0$$

then → $2\mathbf{p}_N = 2\mathbf{R}_{NN}\mathbf{w}_N$

- This is known as the **normal equation**
- error is minimized when condition holds
- error is orthogonal (normal) to projection related to $x(k)$

Least Mean Square (LMS)

when eq. is satisifed → **MMSE** is given by: $J_{opt} = J(\hat{w}_N)$

$$J(\hat{\mathbf{w}}_N) = \mathbf{E}[x(k)x^*(k)] - \mathbf{p}_N^T\hat{\mathbf{w}}_N$$

$\hat{w}_N$ = **normalized tap vector** obtained by iteratively solving eq.

- continue until J_{opt} converges to acceptable small value

(1) **matrix inversion** = *$O(N^3)$* operation

$$\hat{\mathbf{w}}_N = \mathbf{R}^{-1}{}_{NN}\mathbf{p}_N$$

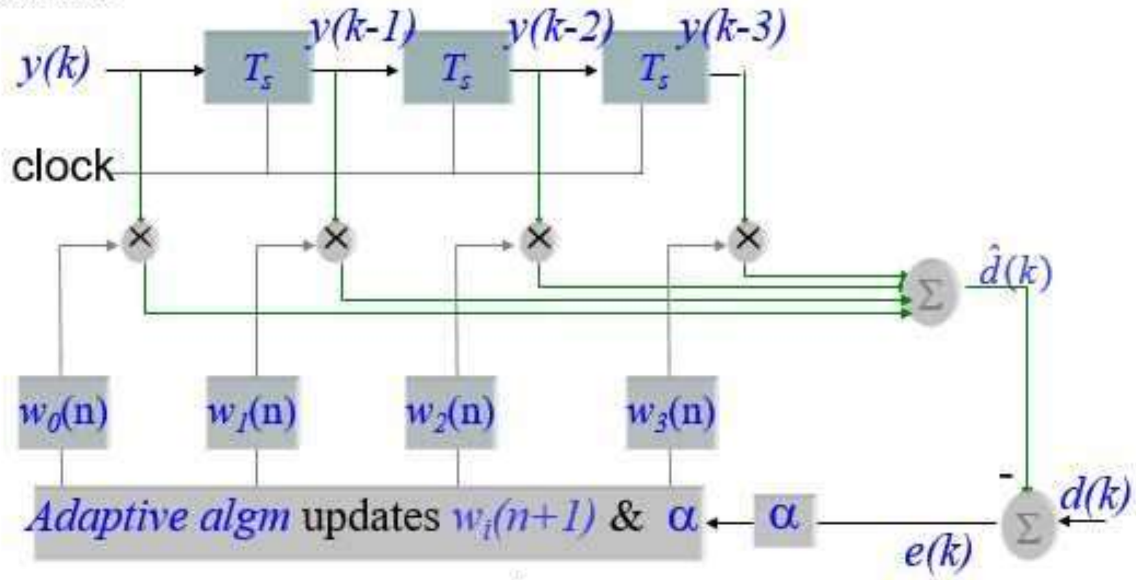
(2) **Gaussian elimination** or **Cholesky Factorization**

- *$O(N^2)$ per iteration*
- only requires N symbol inputs to solve normal equation
- avoids need for long training sequence

Least Mean Square (LMS)

LMS equalizer properties

- maximizes **signal-to-distortion** ratio within constraints of equalizers filter length, *N* as a performance constraint
- if **time-dispersion** characteristics of *y(k)* > *propagation-delay* through equalizer → equalizer is not able to reduce distortion
- **convergence rate** is slow – only one parameter, step size, α to control adaptation rate



$$w_N(n+1) = \mathbf{w}_N(n) - \alpha e_k^*(n)\mathbf{y}_N(n)$$

Least Mean Square (LMS)

to prevent **unstable adaptation** → α bounded by:

$$0 < \alpha < 2 / \sum_{i=1}^{N} \lambda_i$$

$\lambda_i = i^{th}$ eigenvalue of R_{NN} (covariance matrix) where

$$\sum_{i=1}^{N} \lambda_i = y_N^T(n) y_N(n)$$

α can be controlled by **total input power** to avoid instability

- LMS has **slow convergence rate**
- especially when $\lambda_{max} >> \lambda_{min}$ → λ_i's has large spread

DIVERSITY

– To Compensate fading channel impairments, implemented by using two or more transmitting and receiving antennas

Principle:

– Receiver has multiple copies of the transmit signal

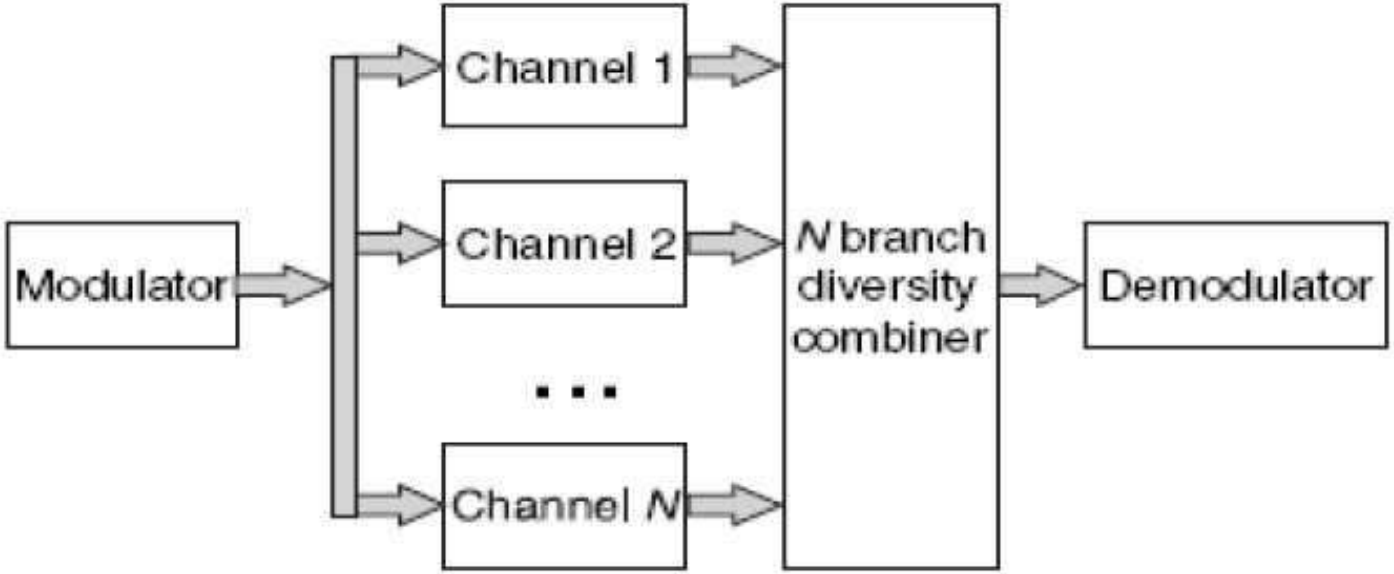
– Same signal is transmitted by more than one antenna

– Selects the best signal

– Average SNR at the receiver may be improved

DIVERSITY

- If one radio path undergoes a deep fade another independent path may have a strong signal.



CLASSIFICATION OF DIVERSITY

MACRO- DIVERSITY

- Provides a method to mitigate the effect of shadowing as in case of Large scale fading

MICRO-DIVERSITY

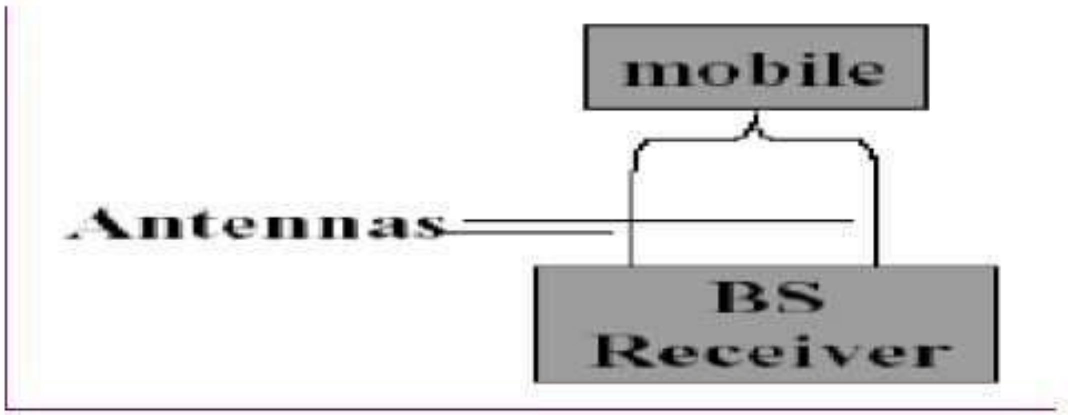
- Provides a method to mitigate the effect of multi-path fading as in case of small scale fading

MICRO DIVERSITY

- Small scale fades are characterized by deep and rapid amplitude fluctuations which occur as the mobiles moves over distances of just a few wavelengths.

- These fades are caused by multiple reflections from surroundings in the vicinity of the mobile.

- Short term fading can be mitigated by the diversity using multiple antennas on the base station or mobile unit.



CLASSIFICATION OF MICRO DIVERSITY

Five common methods are

(1)Spatial diversity : Several antenna elements separated in

space

(2)Temporal diversity : Repetition of the transmit signal at different times

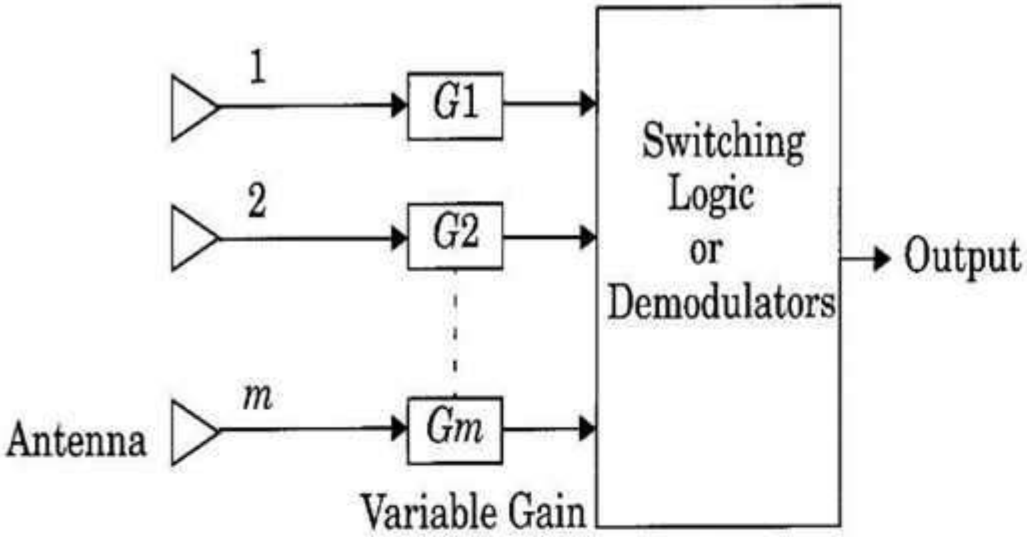
(3)Frequency diversity : Transmission of the signal on different frequencies

(4)Angular diversity: Multiple antennas with different antennas with different antenna patterns

(5)Polarization diversity: Multiple antennas receiving different polarizations

Spatial Diversity

- Space Diversity or Antenna diversity
- Use more than one antenna to receive the signal.



Generalized block diagram for space diversity.

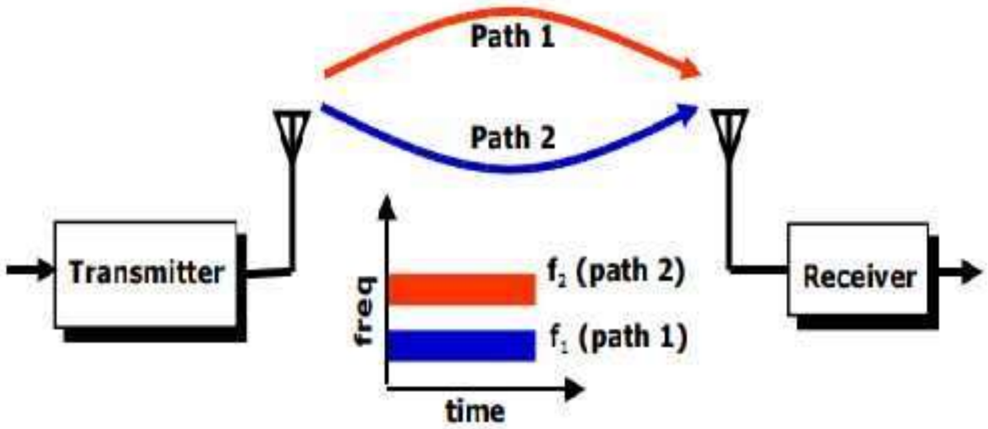
Temporal Diversity

- As the wireless propagation channels are time variant, signals that are received at different times are uncorrelated
- Temporal Distance $D = \frac{1}{2V_{max}}$

  Vmax→ Maximum Doppler frequency
- Temporal diversity can be realized in 3 different ways.
- **Repetition coding-** signal is repeated several times
- **Automatic repeat request-** Rx sends message to TX
- **Combination of interleaving and coding** – Forward Error Correction codes. Transmitted code is reconstructed

Frequency Diversity

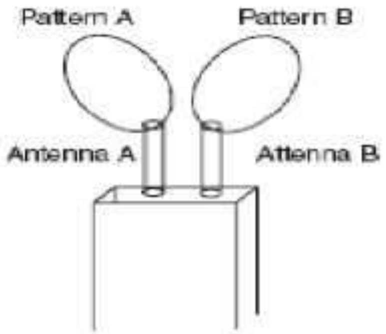
- Frequency diversity is implemented by transmitting same signal at two or more different carrier frequencies.
- Our aim is to make these carrier frequencies uncorrelated to each other, so that they will not experience the same fades.



- Frequency diversity is often employed in microwave line –of sight links.
- These links uses Frequency division multiplexing mode(FDM)

Angular or Pattern Diversity

- It enhances the decorrelation of signals at closely spaced antennas by using Different antenna patterns
- Even identical antennas can have different patterns when they are mounted close to each other.
- This effect is due to Mutual Coupling.
- Place 2 identical antennas close to each other



Angle diversity for closely spaced antennas

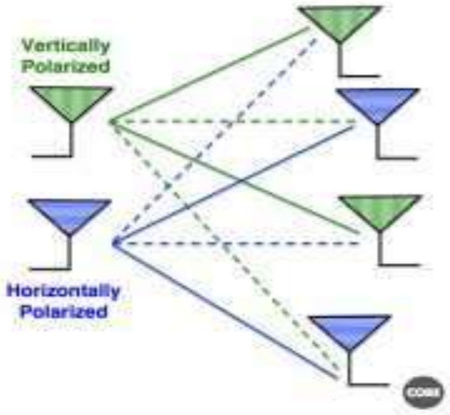
- Here antenna B acts as a reflector for antenna A.
- Different patterns are used when located on different cases

Polarization Diversity

- Multiple versions of a signal are transmitted and received via antennas with different polarization.

  i.e. horizontal and   vertical.

- A diversity combining technique is applied on the receiver side.



- Signals are transmitted through two orthogonally polarized propagation paths
- Improves link margin and capacity

Macro Diversity

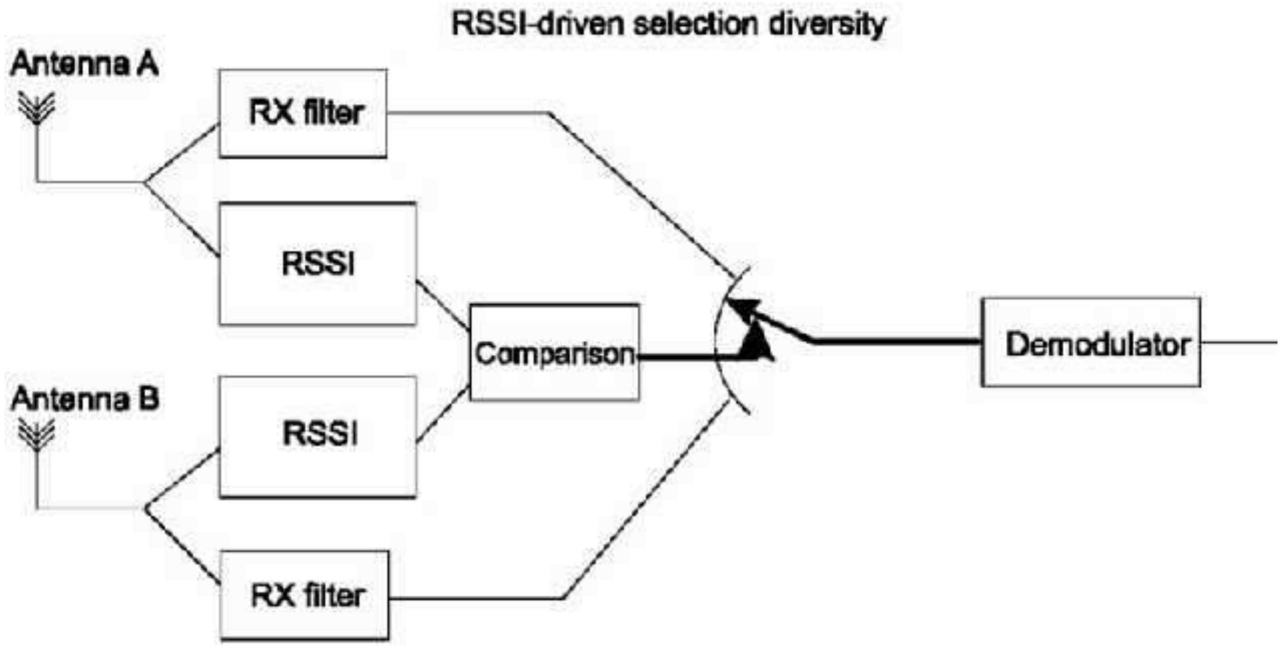
- Combat large-scale fading (fading created by shadowing effects)
- Large distance between BS1 and BS2 gives rise to macro diversity.
- Use on-frequency repeaters (receive the signal and retransmit the amplified version).
- Simulcast (same signal transmitted simultaneously from different BSs.)
- Simulcast is widely used for broadcast applications like digital TV.

Macro Diversity-Methods

- **Selection diversity - 'Best'** signal copy is selected and processed (demodulated and decoded) and all other copies are discarded
- **Combining Diversity -** All signal copies are combined and combined signal is decoded

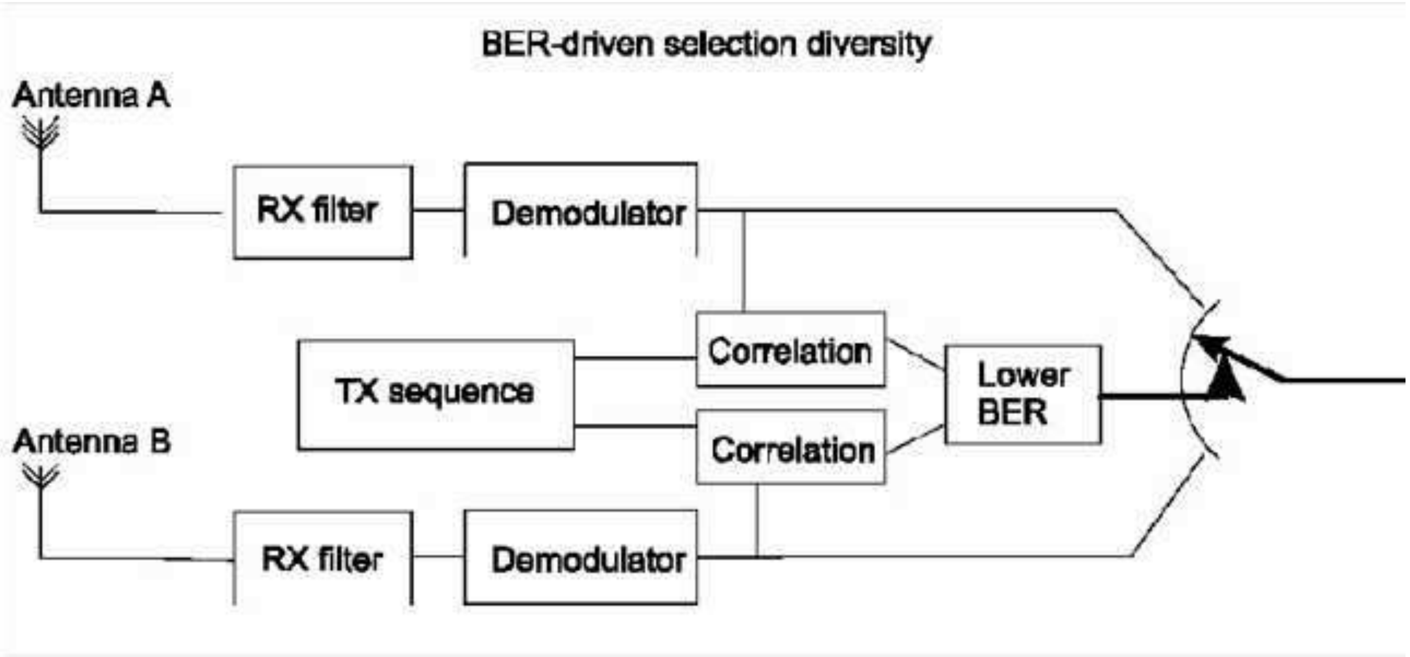
- Note: Combining diversity leads to better performance but Rx complexity higher than Selection Diversity.

Selection Diversity



- The receiver selects the signal with largest power
- RSSI-Received Signal Strength Indication

Selection Diversity



- First transmit the training sequence
- Receiver demodulates the signal and compare it with the transmit signal
- Based on smaller BER, suitable channel is selected

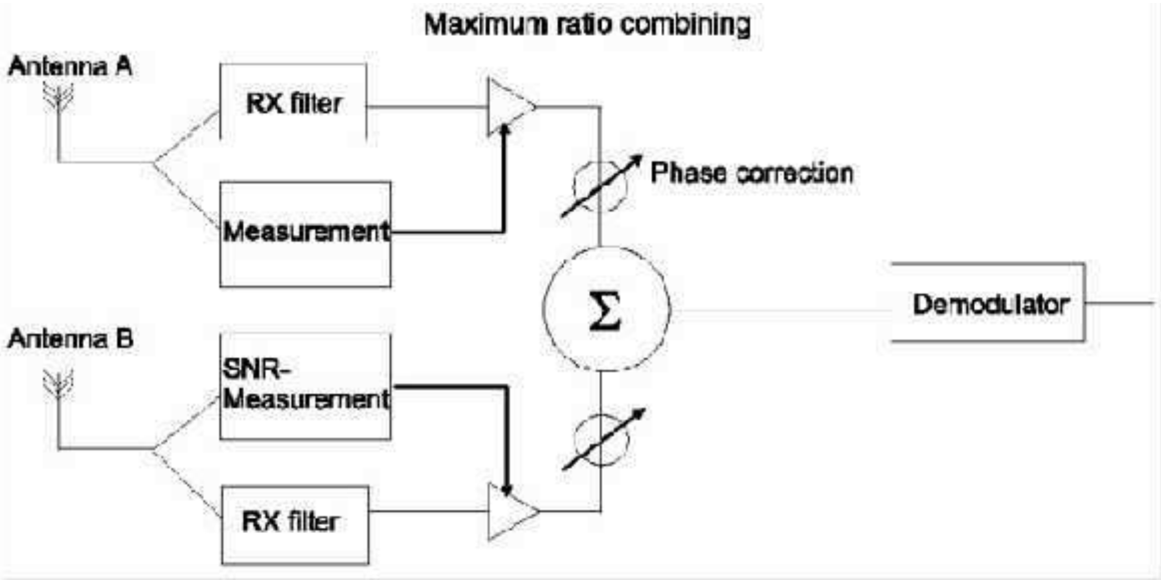
Combining Diversity Types

- **Maximal Ratio Combining** – Weights all signal copies by their amplitude

- **Equal Gain Combining** – Signals are not weighted but undergo phase correction

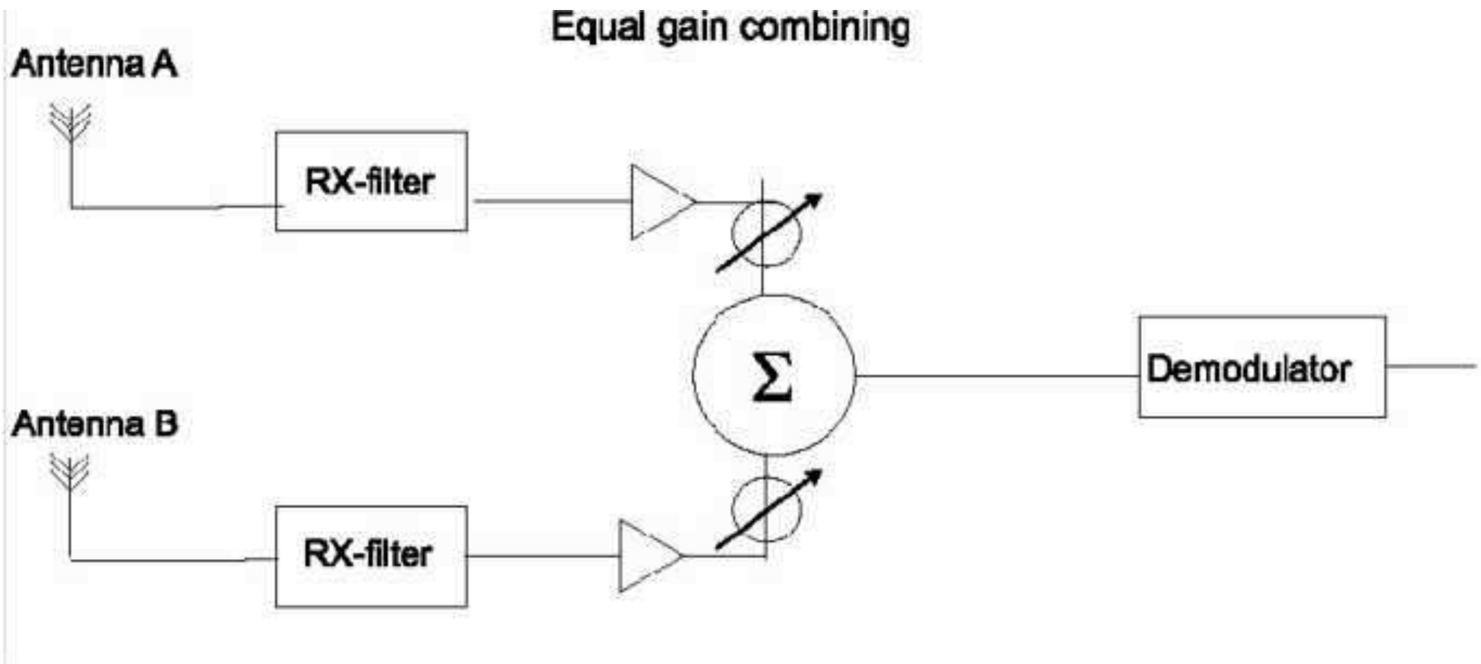
Maximal Ratio Conbining

- Signals from all the branches are weighted according to their individual signal voltage to noise power ratios and then summed
- Signals must be co-phased before being summed
- Requires an individual receiver and phasing circuit for each antenna element



Equal Gain Combining

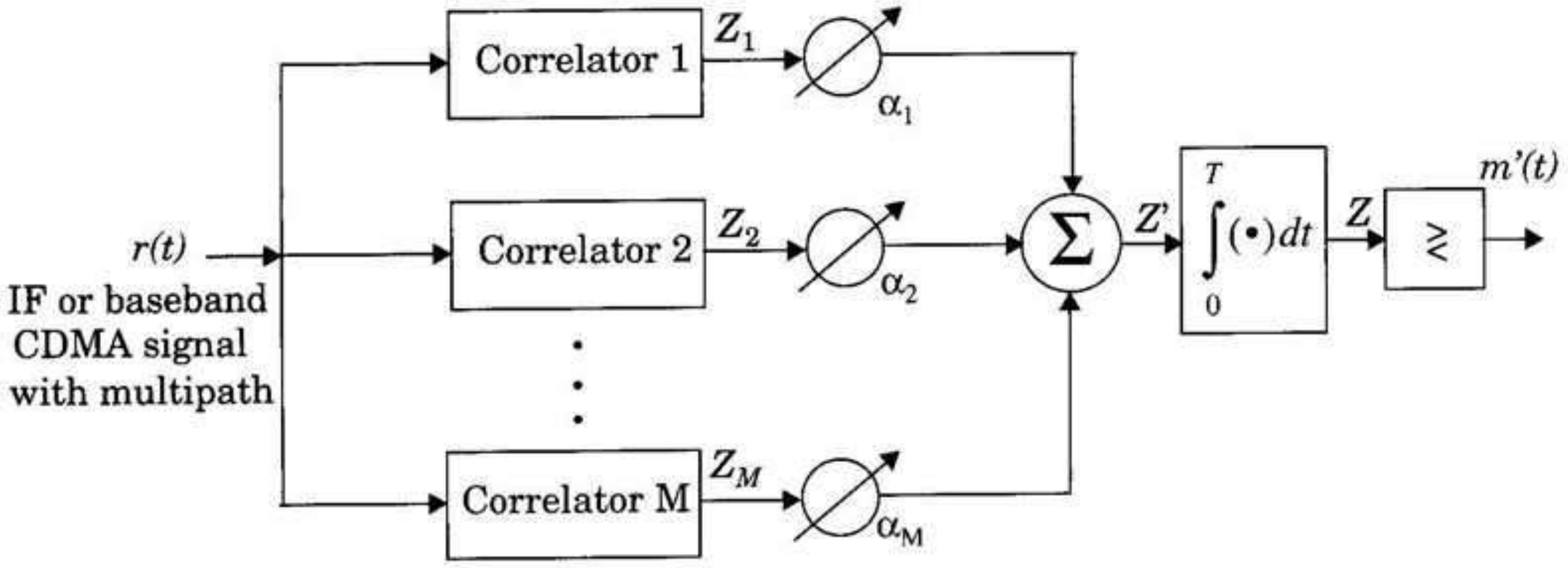
- Weighting circuits are omitted
- Branch weights are set to unity
- Signals from each branch are co-phased to provide equal gain combining diversity



RAKE Receiver

- Used in CDMA cellular systems
- Combine multipath components which are time delayed versions of original signal transmission
- Combinations improve SNR
- RAKE receiver collects the time shifted versions of the original signal by providing separate correlation receiver for each of multipath signals
- RAKE receiver uses multipath diversity principle
- It is like a rake that rakes the energy from the multipath propagated signal components

RAKE Receiver



An *M*-branch (*M*-finger) RAKE receiver implementation. Each correlator detects a time shifted version of the original CDMA transmission, and each finger of the RAKE correlates to a portion of the signal which is delayed by at least one chip in time from the other fingers.

RAKE Receiver

- Consider M correlators are used in CDMA receiver to capture M strongest multipath components
- weighting network provides a linear combination of the correlator output for bit decision.
- Correlator 1 is synchronized to the strongest multipath m1
- Multipath component m2, arrived t1 later than m1, but has low correlation than m1
- Outputs of the M correlators are denoted as Z_1, Z_2… Z_M and weighted by α_1, α_2… α_M

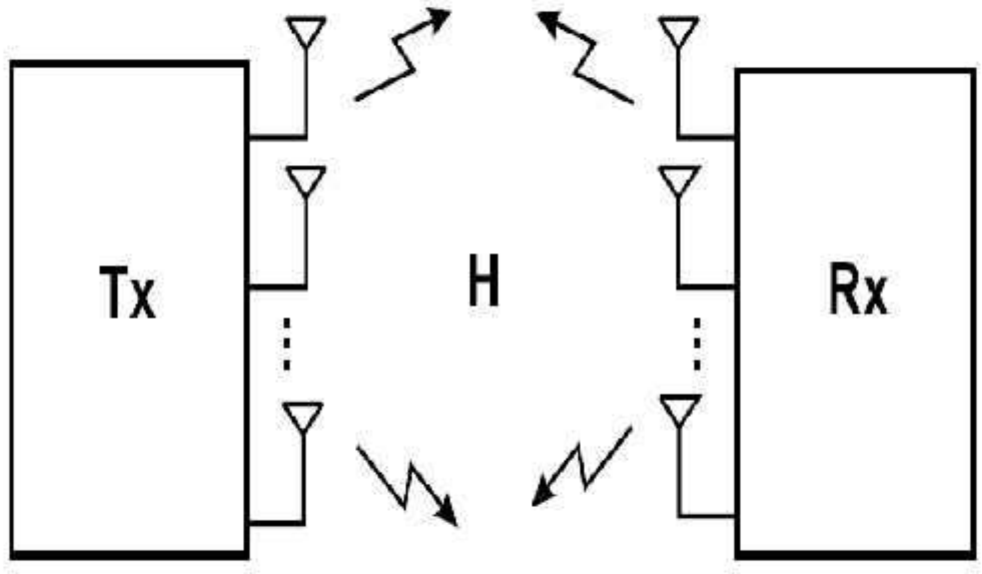
UNIT - 5
MULTIPLE ANTENNA TECHNIQUES

MIMO systems – spatial multiplexing -System model - Pre-coding - Beam forming - transmitter diversity, receiver diversity- Channel state information-capacity in fading and non-fading channels.

MIMO Systems

- By using Multiple Output Multiple Input (MIMO) systems
- Diversity gain mitigates the fading and increases coverage and improves QoS
- Multiplexing gain increases capacity and spectral efficiency with no additional power or bandwidth expenditure
- Array gain results in an increase in average receive SNR.
- Spatial Diversity and Spatial Multiplexing can be conflicting goals

MIMO Systems



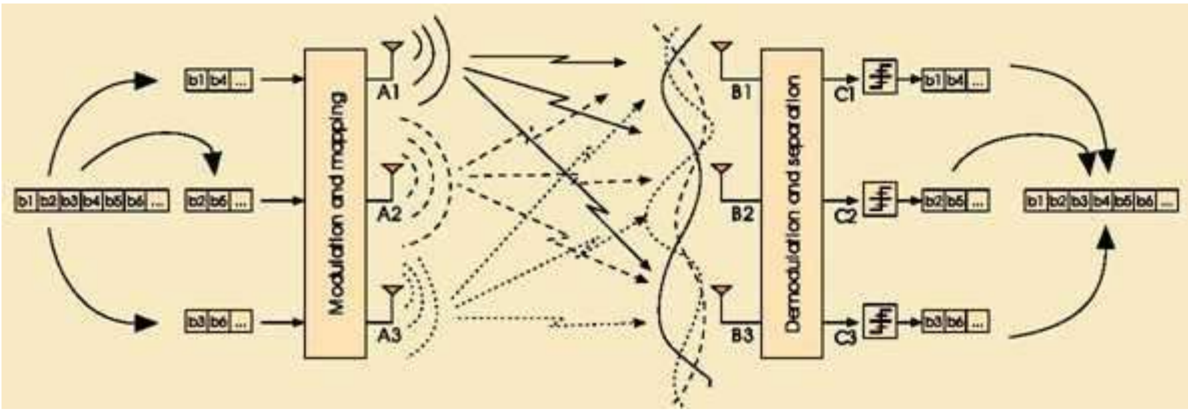
Spatial Diversity and Spatial Multiplexing

- Spatial Diversity
  - Signal copies are transferred from multiple antennas or received at more than one antenna
  - redundancy is provided by employing an array of antennas, with a minimum separation of $\lambda/2$ between neighbouring antennas

- Spatial Multiplexing
  - the system is able to carry more than one data stream over one frequency, simultaneously

Spatial Multiplexing

- MIMO channels can be decomposed into a number of R parallel independent channels → Multiplexing Gain
  - Principle: Transmit independent data signals from different antennas to increase the throughput, capacity



Pre-coding

- To decrease **MUI**(Multi-user interface) and increase data rate in MIMO system
- Performs Transmit diversity
- Similar to equalization
- Main difference is we need to optimize the precoder with a decoder
- Channel equalization minimize channel errors

Pre-coding

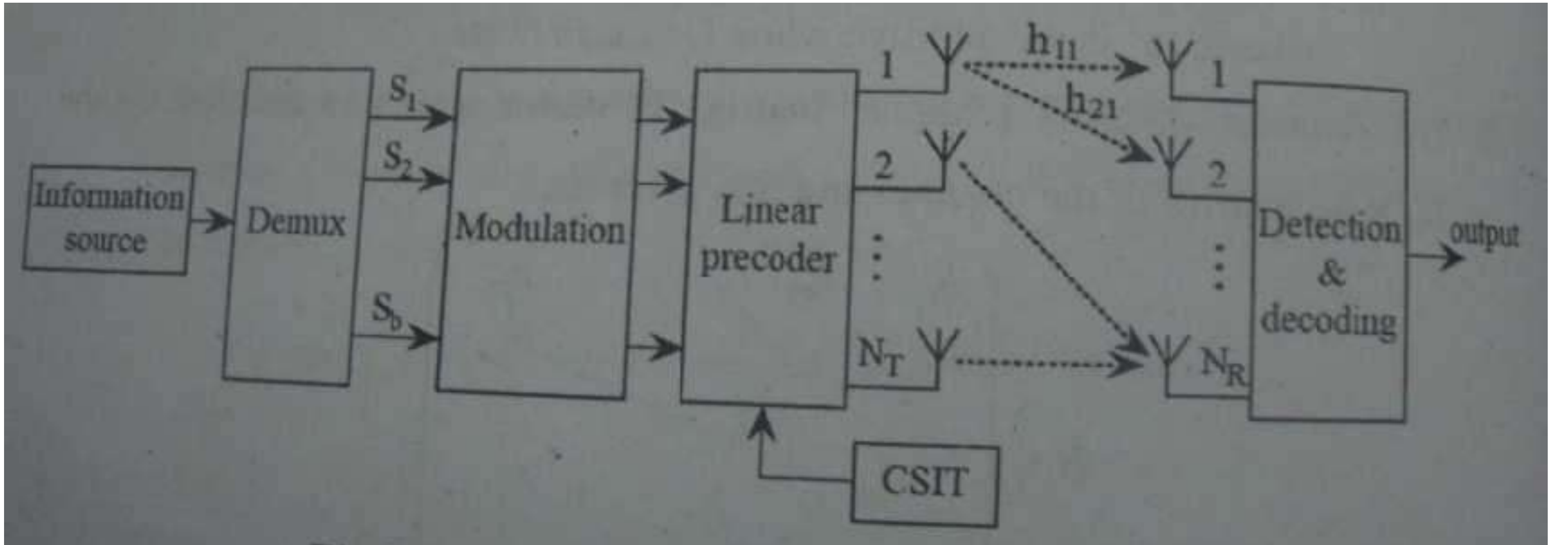


Fig.9.10. Block diagram of MIMO pre-coding

Implementation of MIMO-Vertical Bell Labs layered space-time Architecture

-V-BLAST is a transmitter –receiver which is mainly used to implement multiplexing MIMO
-Achieve diversity of the order of number of received samples
-Each coded symbol is transmitted from one antenna and received by number of received antennas.
-BLAST can average over the randomness of the individual sub-channels and get better outage performance.

Implementation of MIMO-Vertical Bell Labs layered space-time Architecture

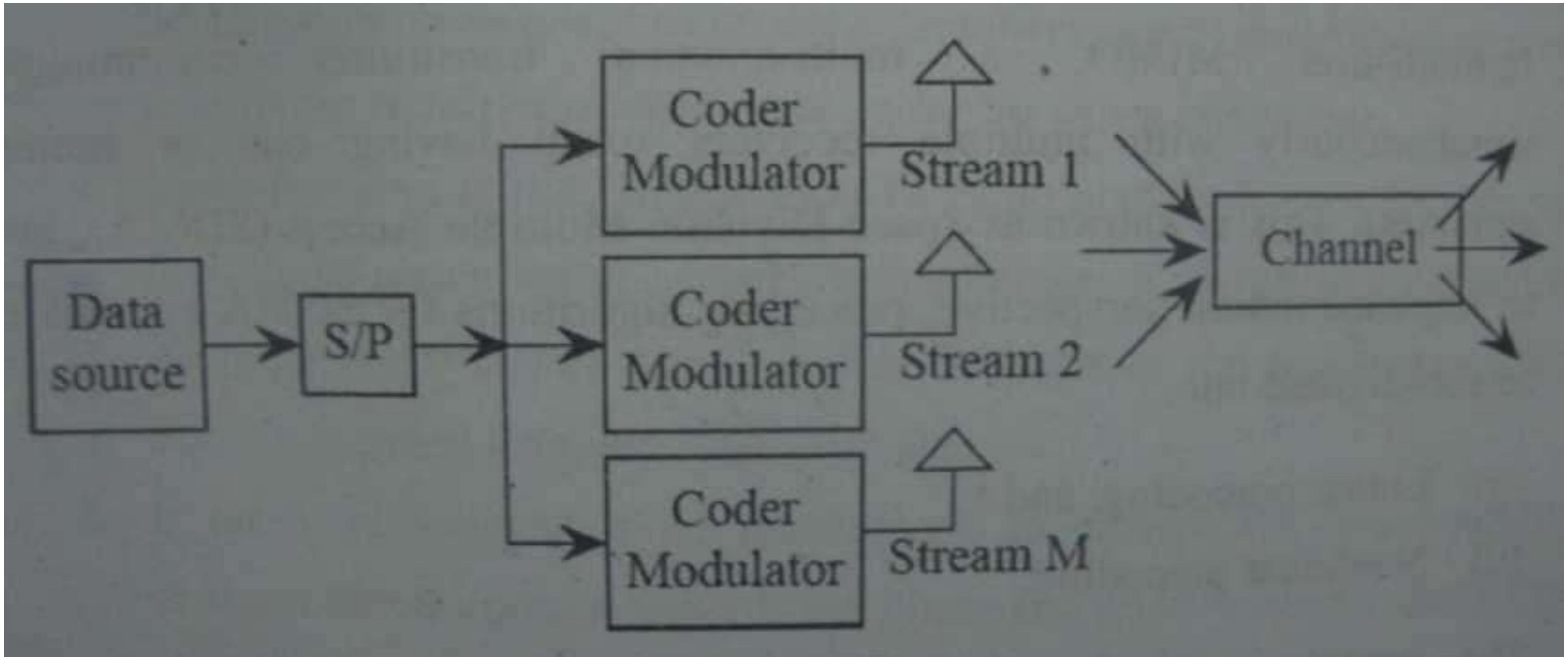


Fig.9.11. BLAST structure

Linear Precoding

Linear precoding :

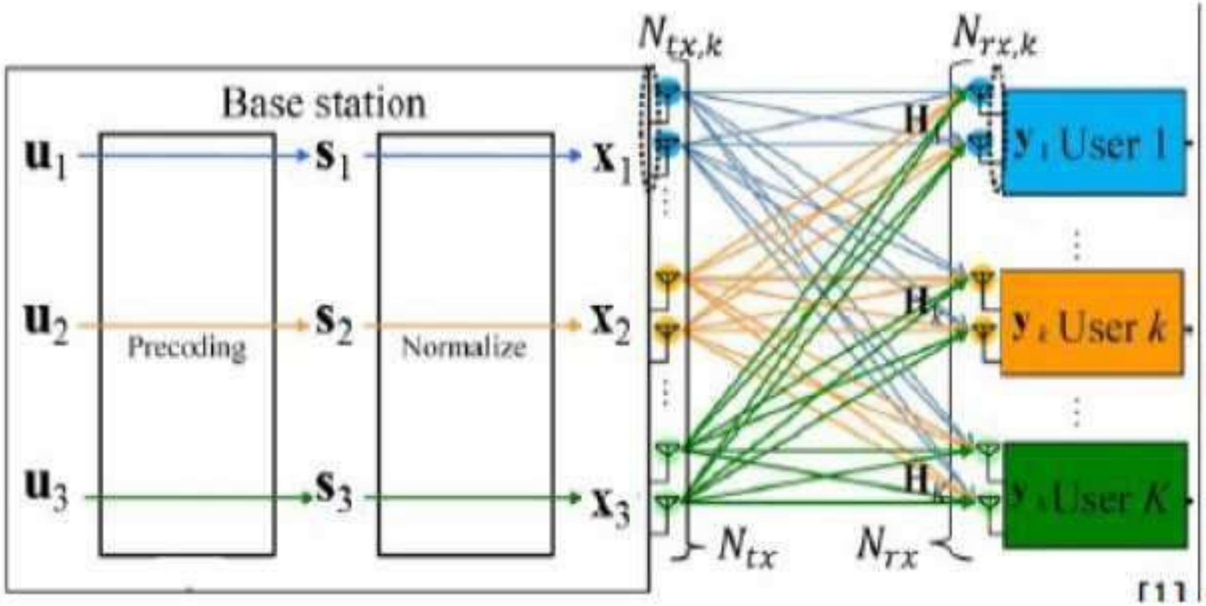
General expression:

$$S = Wu$$

where W is the weighting vector.

$$x = \frac{s}{\sqrt{\gamma}}$$

where $\gamma = E[\sqrt{s^2}]$



[1]

Non-Linear Precoding

Non linear precoding :

Vector Precoding (VP): A data-dependent perturbation vector is used after the linear precoding stage.

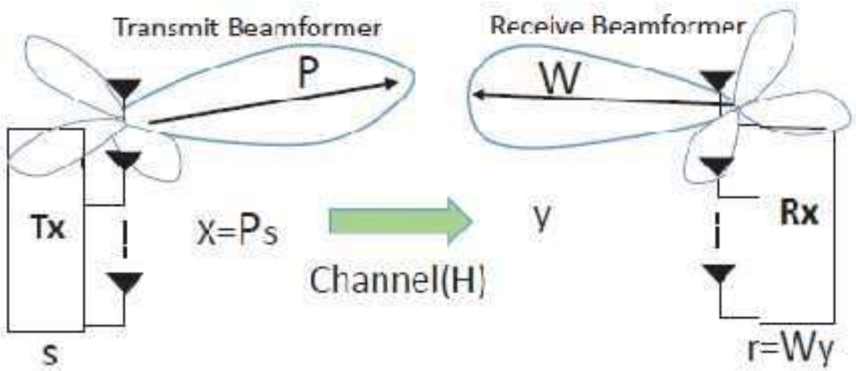
$$\mathbf{x} = \frac{\mathbf{G}(S)(\mathbf{u}-\boldsymbol{\lambda})}{\sqrt{\rho}} \qquad \boldsymbol{\lambda} = \arg\min \| \mathbf{G}(S)(\mathbf{u}-\boldsymbol{\lambda}') \|_2$$

Tomlinson-Harashima and QR decomposition: Part of the interference is eliminated by using a QR decomposition of the channel; the other part is eliminate by using the Tomlinson-Harashima precoder for each sub-channel. For the k-th streams:

$$x_k = u_k + d_k - \alpha s_k - \lambda, \quad \lambda = \arg\min_{\lambda' \in Z[j]} \| u_k + d_k - \alpha s_k - \lambda' \|_2 \qquad 1$$

Beam Forming

- Used in any antenna system, particularly in MIMO systems in order to create certain required antenna directive pattern to give required performance.
- Combination of radio signals from a set of small non-directional antennas to simulate a large directional antenna
- Smart antenna are normally used because it can be controlled automatically according to the required performance



SMART ANTENNA - TYPES

Smart antennas are divided into two types

(i)Phased array systems(PAS)

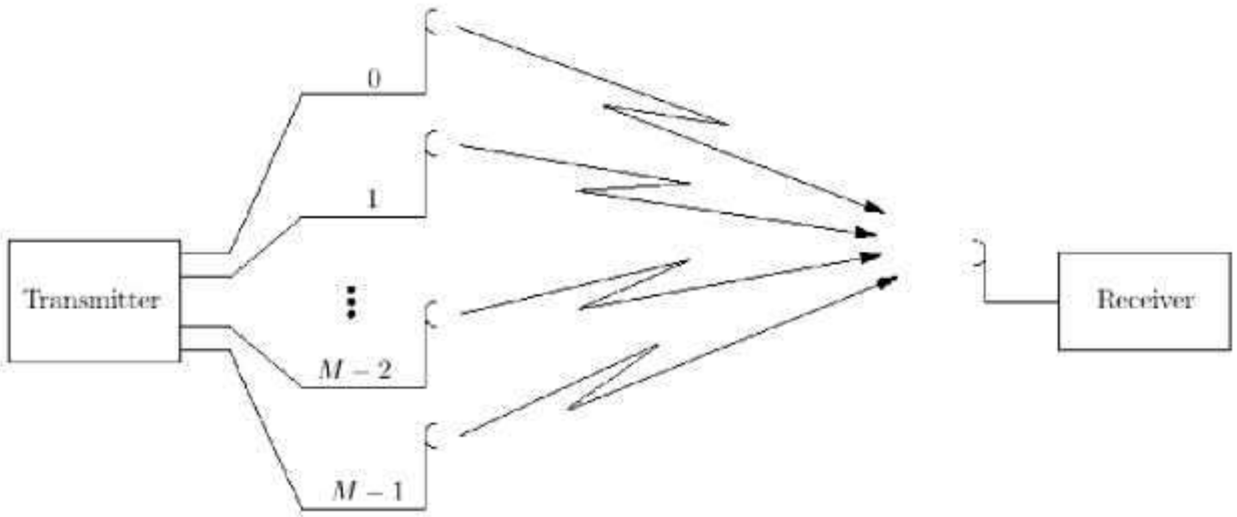
- Have number of predefined patterns
- The required one being switched according to the direction required

(ii)Adaptive array systems(AAS)

- Uses infinite number of patterns
- Adjusted based on the requirements in real time

Transmit Diversity

- Multiple antenna elements are required at the transmitter
- Only one antenna at the receiver end provides better performance



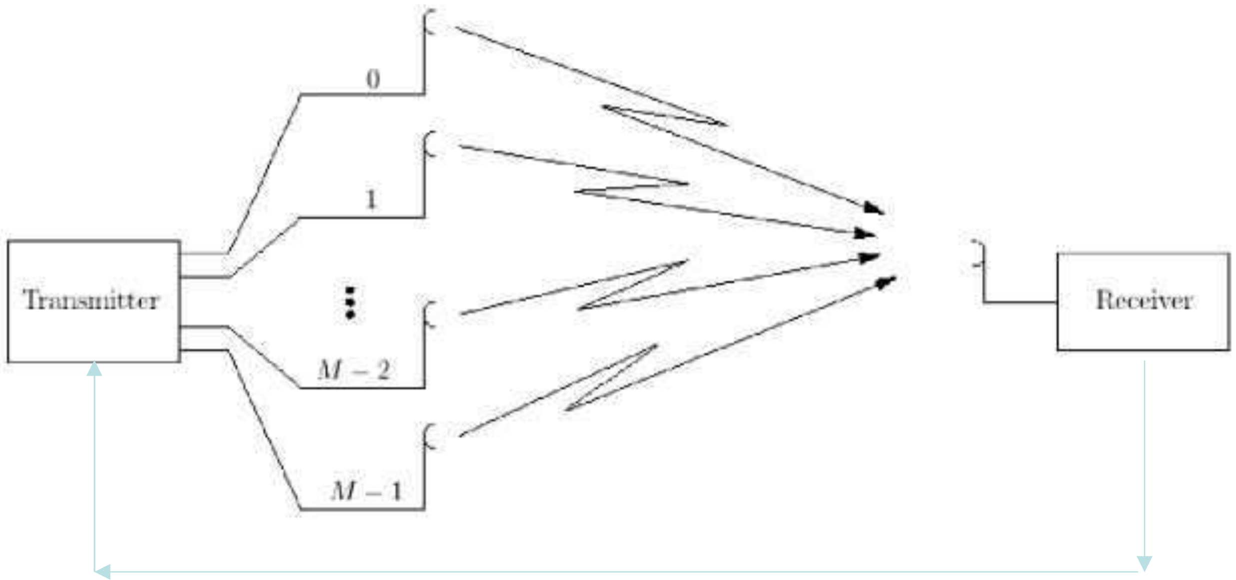
Transmit Diversity-Types

Two Types

- (i)Transmitter diversity with the channel state information(closed loop transmit diversity)
- (ii)Transmitter diversity without the channel state information(open loop transmit diversity)

Closed loop Transmit Diversity

- Transmitter knows perfectly about the channel
- This information is obtained by means of feedback from the receiver



Closed loop Transmit Diversity

- If the selected antenna maximizes the signal to Interference Noise ratio(SINR), then the transmit encoder receives feedback channel state information from the receiver
- This fine encoding matrix maximizes SNR at the receiver
- Space Time Block Coder(STBC) is an efficient means of achieving optimal transmitter diversity gain.

Open loop Transmit Diversity

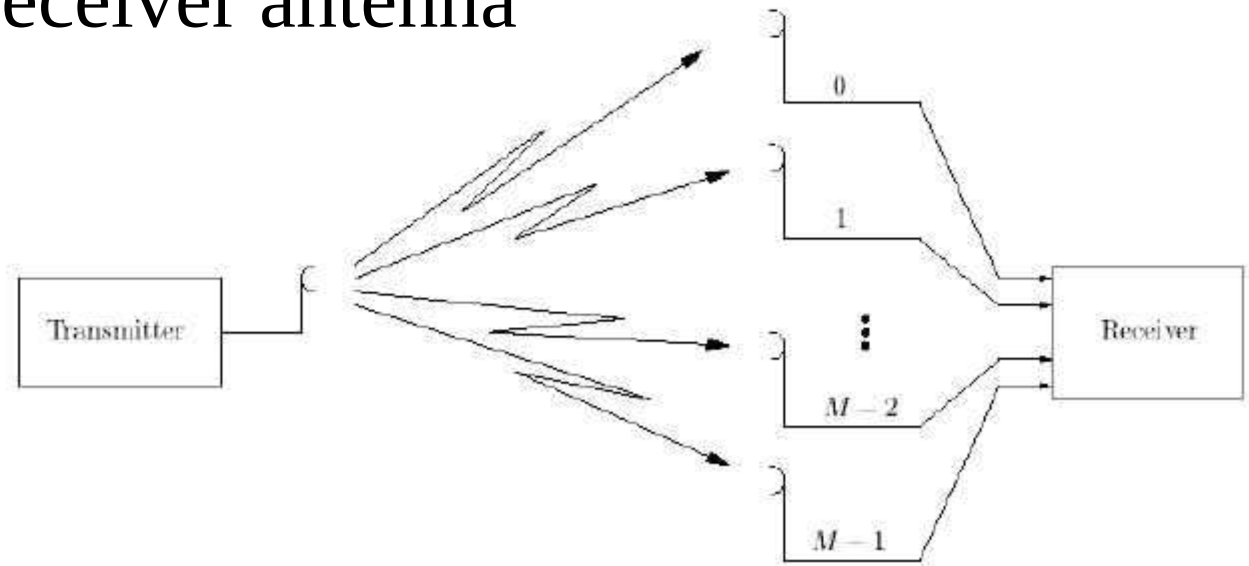
- Transmission of signals from different antenna elements allows the receiver to distinguish different transmitted signal components.
- One method is delay diversity
- We transmit data streams with the delay of one symbol duration from each of the transmit antennas

Receiver Diversity

- One transmitting antenna and many receiving antennas are used.
- Desired message is transmitted by using single transmitting antenna and received by multiple antennas
- Different antennas appropriately separated are deployed at the receiver to combine the uncorrelated fading signals called SPACE DIVERSITY

Receiver Diversity

- Single transmitting antenna and multiple receiver antenna



Classification of Space Diversity

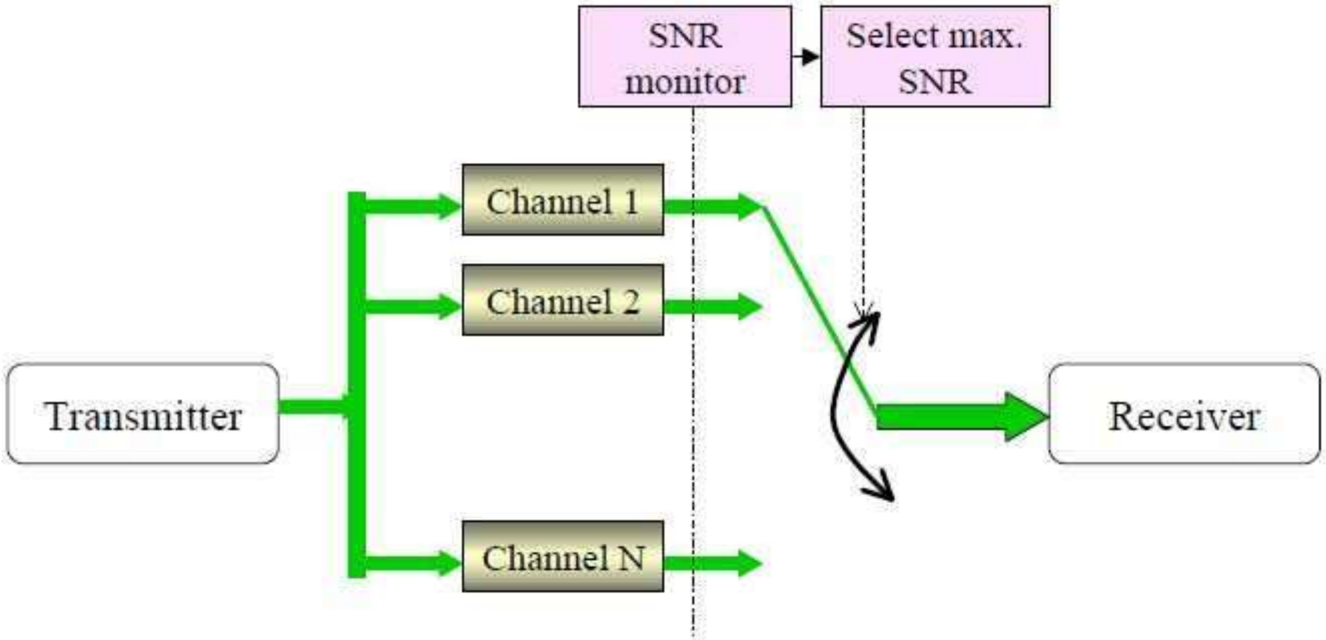
1) Selection diversity
2) Feedback diversity
3) Maximal radio combining
4) Equal gain combining diversity

Selection Diversity

- The receiver monitors the SNR value of each diversity channel and chooses the one with the maximum SNR value.
- The receiver with M demodulators are used to provide M diversity branches, whose gains are adjusted to provide the same average SNR for each branch
- Receiver branch having the highest instantaneous SNR is connected to the demodulator

Selection Diversity

- Select the strongest signal



Feedback/scanning Diversity

- Similar to selection diversity except that instead of always using the best of ‘n’ signals, the ‘n’ signals are scanned in a fixed sequence until one is found to be above a predetermined threshold.

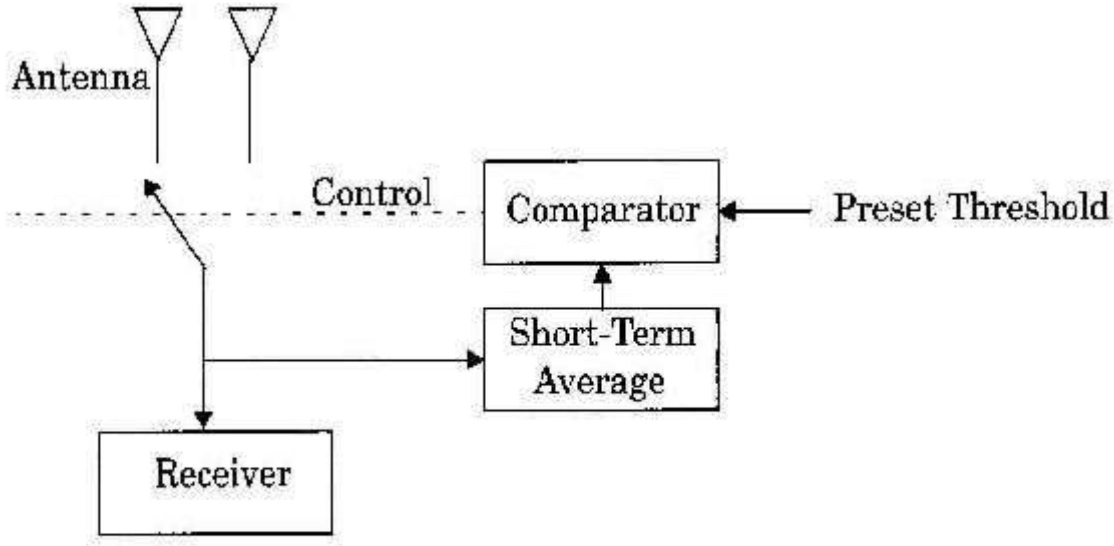


Figure 7.13 Basic form of scanning diversity.

Maximal Ratio Combining

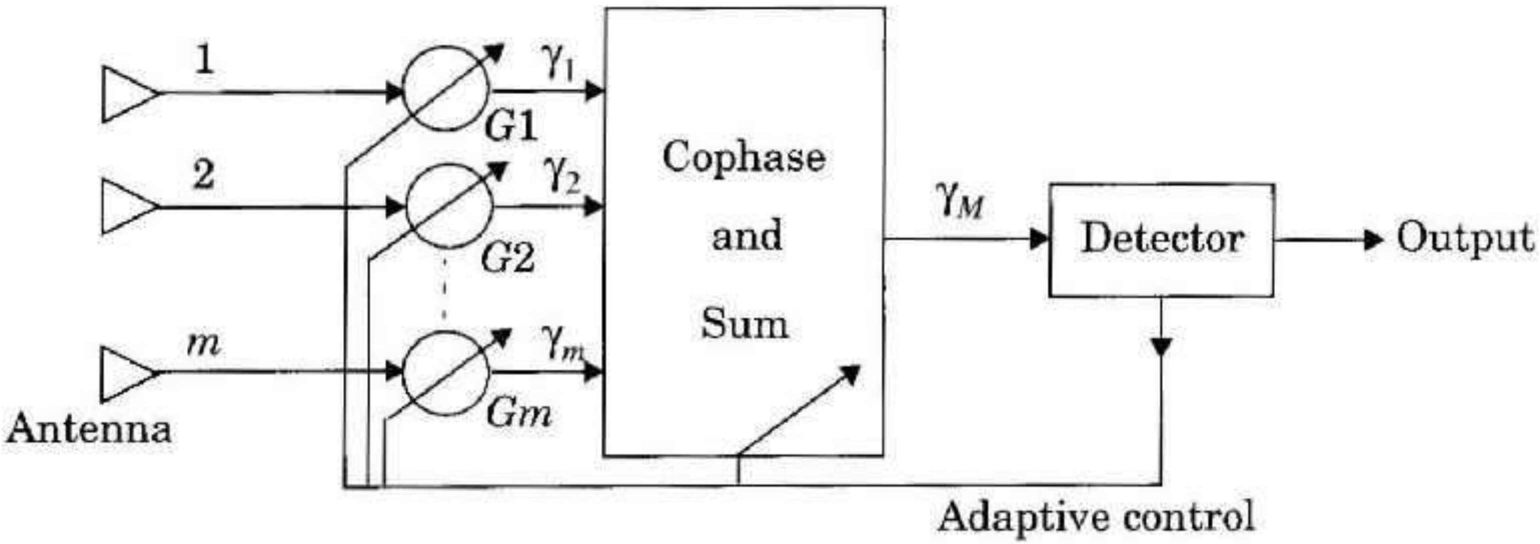


Figure 7.14 Maximal ratio combiner.

Maximal Ratio Combining

- The signals from all of the M branches are weighted according to their individual signal voltage to noise power ratios and then summed up
- Individual signals must be co-phased before being summed which generally requires an individual receiver and the phasing circuit for each antenna element
- MRC produces an output SNR which is equal to the sum of the individual SNRs.

Equal Gain Diversity

- Similar to MRC except that there will be an omission of the weighted circuits

-The branch weights are all set to unity but the signals from **each branch are co-phased to provide an equal gain** combining diversity

-Allows the receiver to exploit the signals that are simultaneously received on each branch.

CHANNEL STATE INFORMATION(CSI)

- Represents the properties of a communication link between the transmitter and receiver.
- CSI describes how a signal propagates from the transmitter to the receiver
- Represents the combined effect of scattering, fading and power decay with distance
- It is usually estimated at the receiving end and then quantized and fed back to the transmitter side

CHANNEL STATE INFORMATION(CSI)- TYPES

INSTANTANEOUS CSI

- Short term CSI
- Current conditions of the channel are known which can be viewed as knowing the impulse response of the digital filter.
- Optimize the received signal for spatial multiplexing or to achieve low bit error rates

CHANNEL STATE INFORMATION(CSI)-TYPES

STATISTICAL CSI

- Long term CSI
- Statistical characterization of the channel are known .
- Include the type of fading distribution, the average channel gain, line-of-sight component and spatial correlation

MIMO-CSI Transmission

Algorithms used for MIMO transmission can be categorized based on the amount of CSI

Full CSI at the transmitter and Full CSI at the receiver:

- Both TX and RX have full and perfect knowledge of the channel
- Results in highest possible capacity

Average CSI at the transmitter and Full CSI at the receiver:

- Rx has full information about the instantaneous channel state, but Tx knows only the average CSI

MIMO-CSI Transmission

No CSI at the transmitter and Full CSI at the receiver:

- Without any feedback this is achieved
- Transmitter does not use any CSI while Receiver learns the instantaneous channel state from a training sequence

No CSI at the transmitter and No CSI at the receiver:

- The channel capacity is high when neither the TX nor the RX have CSI

MIMO-CSI Transmission

Noisy CSI:

- When we **assume Full CSI at the receiver**, implies that receiver has learned the channel state perfectly.
- Any received training sequence will get affected by an additive noise as well as quantization noise
- receiver processes the signal based on the observed channel H_{obs}, while the signal pass through the actual channel, H_{act}
- $H_{act} = H_{obs} + \Delta$
- Δ - changes due to noise

MIMO capacity on fading channels

The capacity increase can be seen by comparing MIMO systems with SISO, SIMO, and MISO systems

– SISO:capacity is given by Shannon's classical formula:

$$C = B\log_2(1 + snr \cdot |h|^2)$$

Where B is the BW and h is the fading gain

– SIMO (with M transmitting antennas), the capacity is given by

$$C = B\log_2(1 + snr \cdot \sum_{n=1}^{m} |h_n|^2)$$

–MISO (with M transmitting antennas), the capacity is given by

$$C = B\log_2(1 + \left(\frac{snr}{N}\right) \cdot \sum_{n=1}^{N} |h_n|^2)$$

MIMO capacity on fading channels

- The capacity for MIMO systems can have the following forms (Assuming Tx antennas = Rx antennas = N):

A) If the channel is not known at the transmitter:

$$C = N \log_2\left(1 + \left(\frac{E_s}{N\sigma^2}\right) \cdot |h_n|^2\right)$$

- Where E_s is the total power, σ^2 is noise level of AWGN
- Hence the power is equally shared by each channel
- The capacity grows linearly with the number of antennas

B) If the channel is known at the transmitter

$$C = \sum_{n=1}^{N} \left[\log_2\left(1 + \left(\frac{E_n}{\sigma^2}\right) \cdot |h_n|^2\right) \right]$$

MIMO capacity on fading channels

- No CSI at transmitter and Full CSI at Receiver

Special Cases:

$N_T = N_R = N$

(i) All transfer functions are identical

-All antenna elements are very close together

Capacity, $C = \log 2(1+N\gamma)$

(ii) All transfer functions are Different

-All antenna elements are spaced far apart

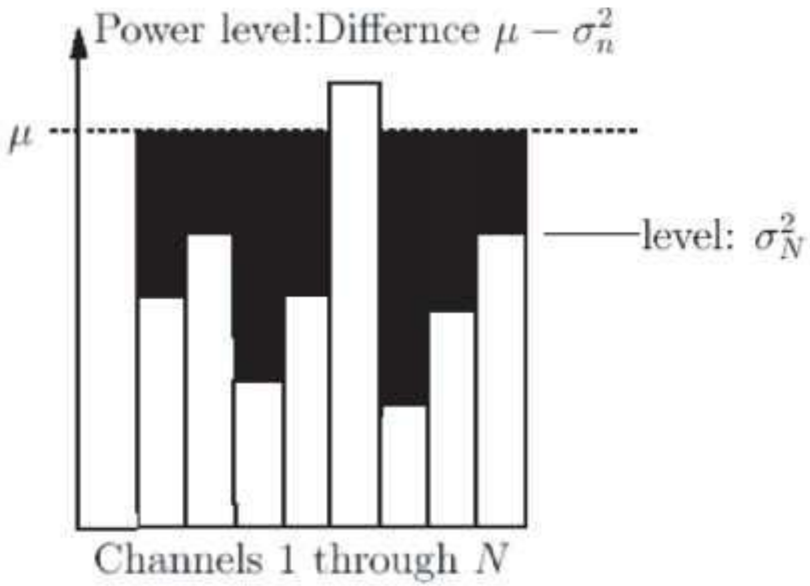
Capacity, $C = N \log 2(1+\gamma)$

Capacity increases linearly with the number of antenna elements

MIMO capacity on fading or Frequency selective channels

- With the channel known at the transmitter and receiver, the total power allocation the each channel will be based on water filling.

–Water filling: Strong Sub-channel, with low noise power level will be assigned with a higher signal power.



Where $\sigma_N^2 = \sigma^2 / |h_n^2|$

MIMO capacity on fading or Frequency selective channels

- Imagine a number of connected vessels
- Bottom of each vessel is a block of concrete with height equal to noise power of corresponding sub channel
- Amount of water poured is proportional to the total transmit power
- Water filling makes sure that energy is not wasted on sub-channels that have poor SNR.
- Some part of the vessel will not be covered by water. These parts correspond to sub channel with strong noise

Capacity in flat fading or non fading Channels

Three cases:

- Fading statistics known
- No CSI at TX and perfect CSI at RX
- Perfect CSI at TX and RX

Capacity in flat fading or non fading Channels

Two types

- Ergodic (Shannon) capacity
- Outage Capacity

Ergodic capacity

- Expected value of the capacity, taken over all realizations of the channel.

Outage Capacity

- It is the minimum transmission rate that is achieved over a certain fraction of time

Capacity in flat fading or non fading Channels

No CSI at TX and perfect CSI at the RX

Capacity, $C = \int B\log_2(1+\gamma)\, p(\gamma)d\gamma$

Full CSI at TX and RX

For fixed transmit power, same capacity will be available only when the receiver knows fading.

- Fading reduces capacity

-Transmit power and transmission rate can be adapted

Channel capacity

$$C = S \int B\log_2(1+(p(\gamma)\,\gamma/s))\, p(\gamma)d\gamma$$