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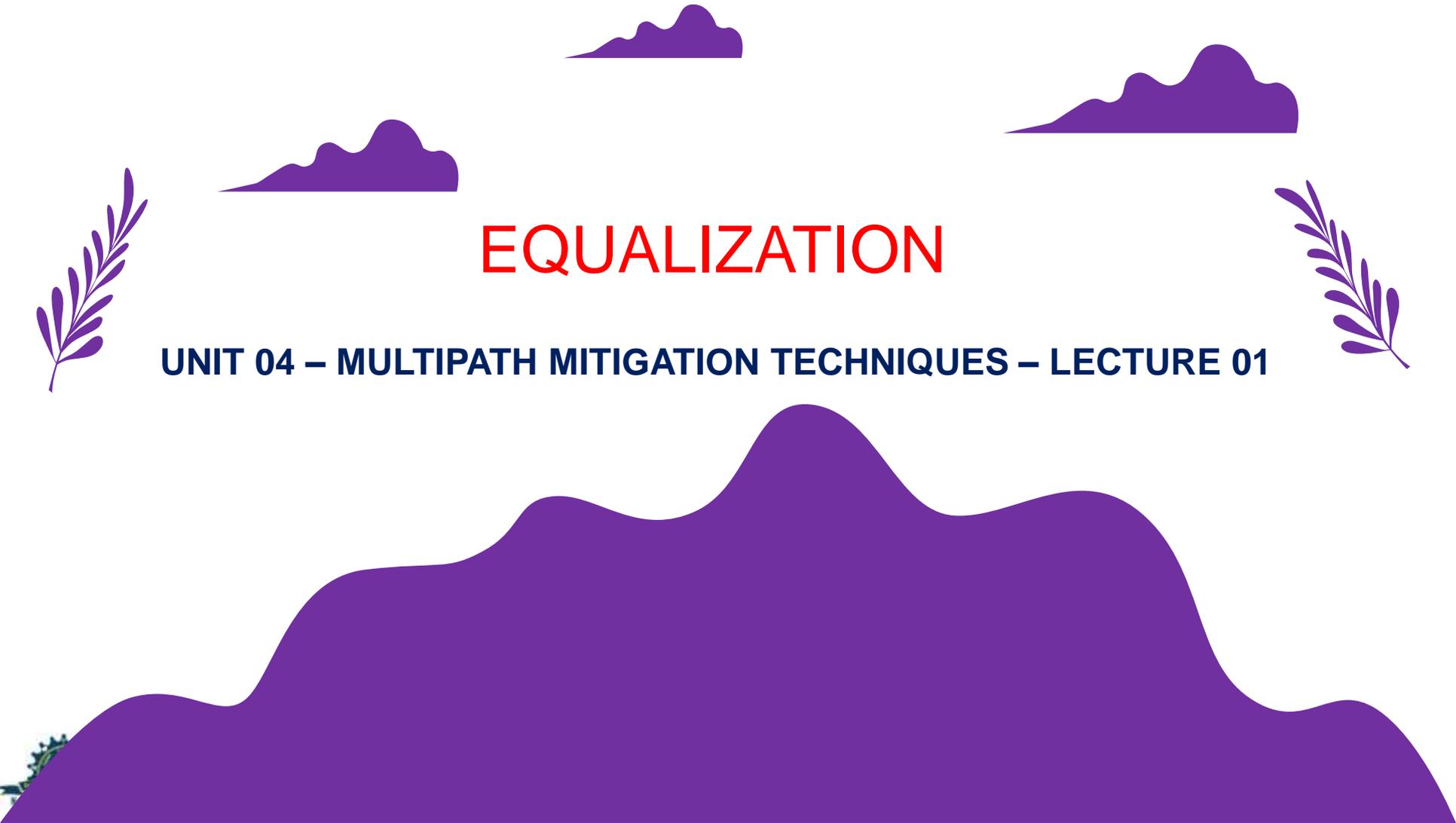
IV YEAR / VIII SEMESTER

EC6801– WIRELESS COMMUNICATION

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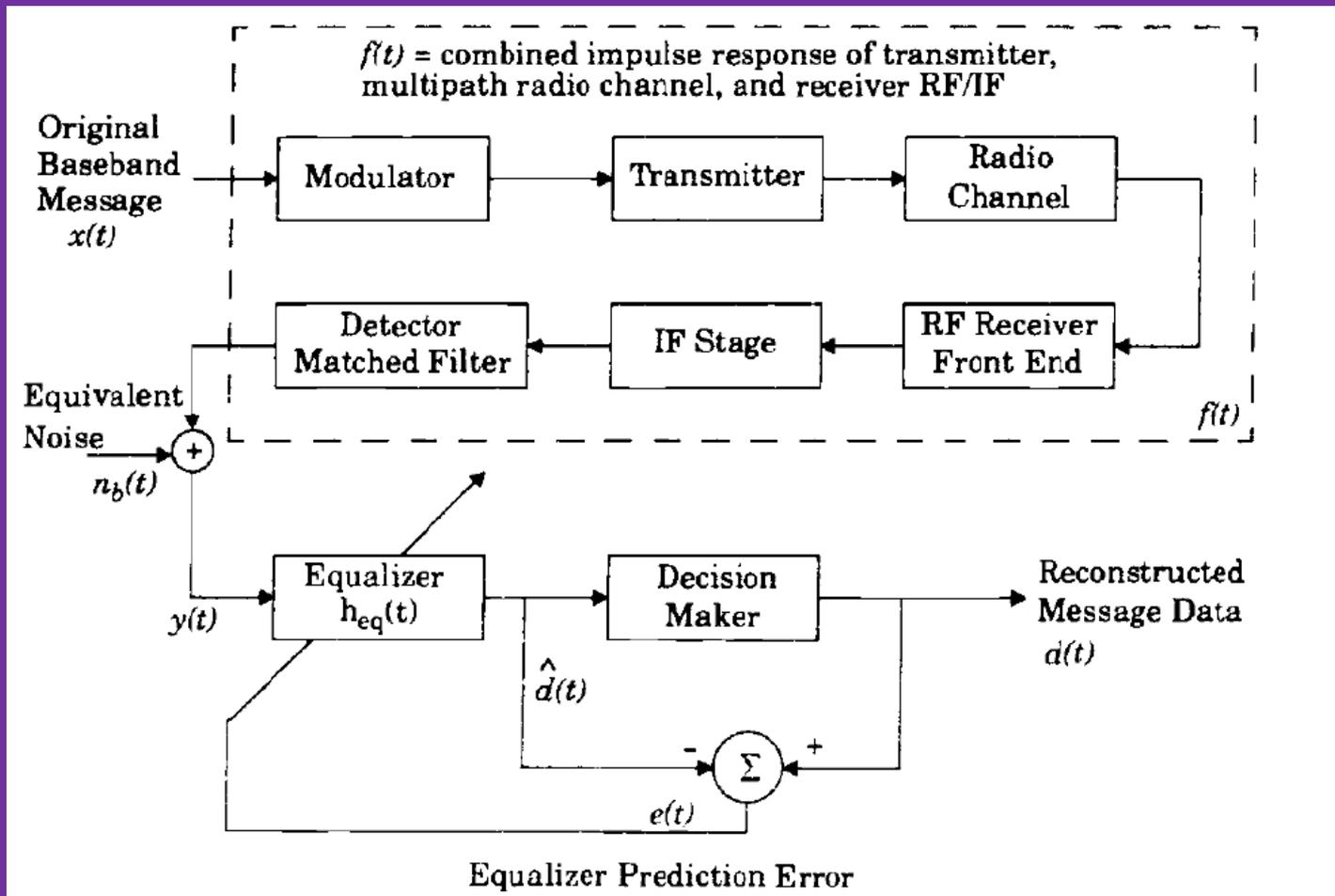
EQUALIZATION

UNIT 04 – MULTIPATH MITIGATION TECHNIQUES – LECTURE 01

INTRODUCTION

- Equalization, diversity, and channel coding are three techniques which can be used independently or in tandem to improve received signal quality.

- **Equalization** compensates for intersymbol interference (ISI) created by multipath within time dispersive channels.

- An equalizer within a receiver compensates for the average range of expected channel amplitude and delay characteristics. Equalizers must be adaptive since the channel is generally unknown and time varying.

- **Diversity** is another technique used to compensate for fading channel impairments, and is usually implemented by using two or more receiving antennas. As with an equalizer, the quality of a mobile communications link is improved without increasing the transmitted power or bandwidth.

- **Channel coding** improves mobile communication link performance by adding redundant data bits in the transmitted message. At the baseband portion of the transmitter, a channel coder maps a digital message sequence into another specific code sequence containing a greater number of bits than originally contained in the message. The coded message is then modulated for transmission in the wireless channel.

Fundamentals of Equalization

- Intersymbol interference ISI caused by multipath in bandlimited (frequency selective) time dispersive channels distorts the transmitted signal, causing bit errors at the receiver has been recognized as the major obstacle to high speed data transmission over mobile radio channels.
- Equalization is a technique used to combat intersymbol interference.
- In a broad sense, the term can be used to describe any signal processing operation that minimizes ISI.
- In radio channels, a variety of adaptive equalizers can be used to cancel interference while providing diversity.
- Since the mobile fading channel is random and time varying, equalizers must track the time varying characteristics of the mobile channel, and thus are called adaptive equalizers.

$f(t)$ = combined impulse response of transmitter, multipath radio channel, and receiver RF/IF

Equalizer Prediction Error

Generic Adaptive Equalizer

- An adaptive equalizer is a time-varying filter which must constantly be retuned.
- The basic structure of an adaptive equalizer has subscript k is used to denote a discrete time index another notation is introduced to represent discrete time events
- The value of Yk depends upon the instantaneous state of the radio channel and the specific value of the noise.
- As such, Yk is a random process.
- The weights of the filter are described by their physical location in the delay line structure, and have a second subscript, k, to explicitly show they vary with time.
- These weights are updated continuously by the adaptive algorithm, either on a sample by sample basis (i.e., whenever k is incremented by 1) or on a block by block basis (i.e., whenever a specified number of samples have been clocked into the equalizer).

New weights = Previous weights + (constant) × (Previous error) × (Current input vector)

Previous error = Previous desired output − Previous actual output

Input Signal
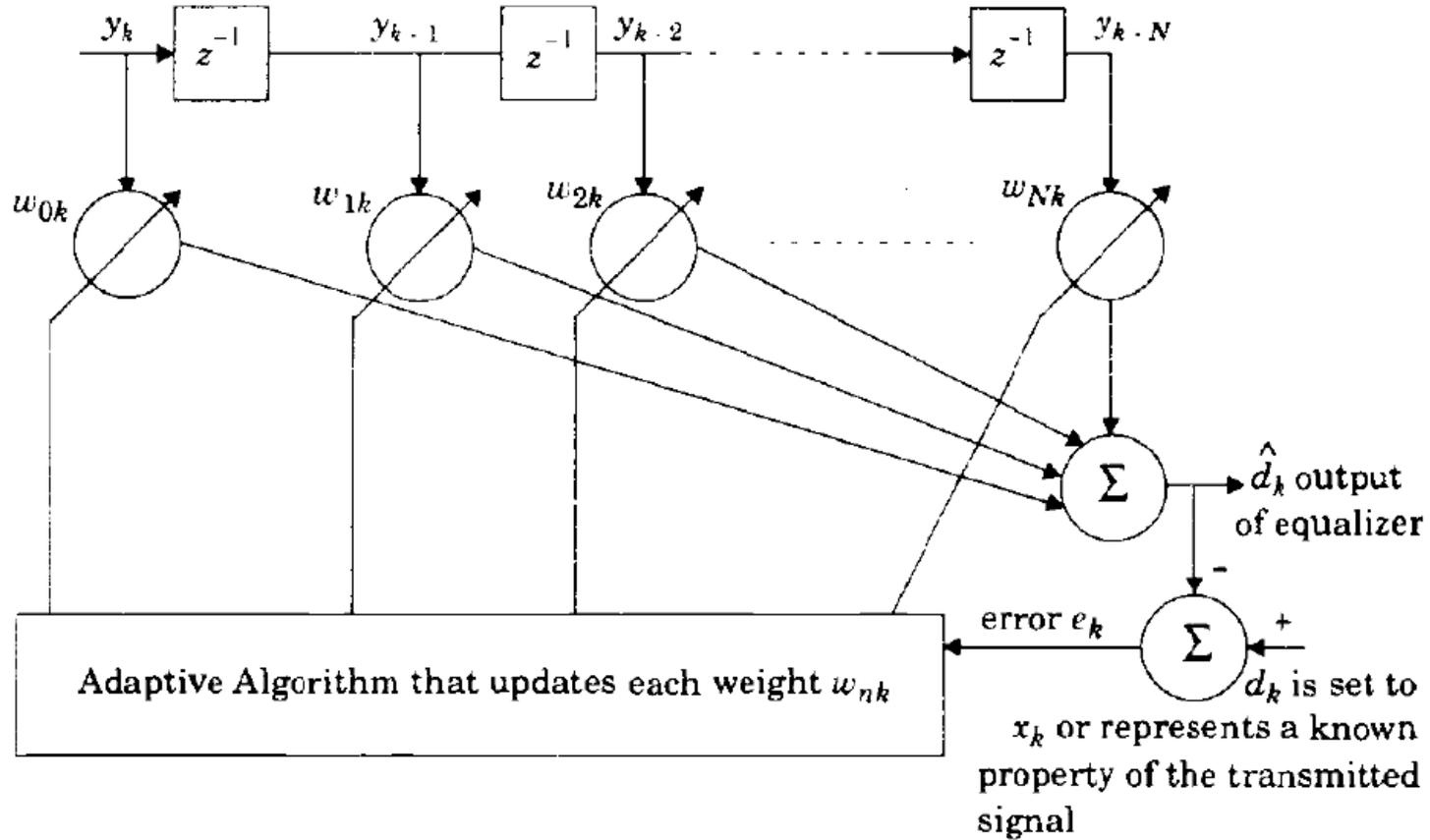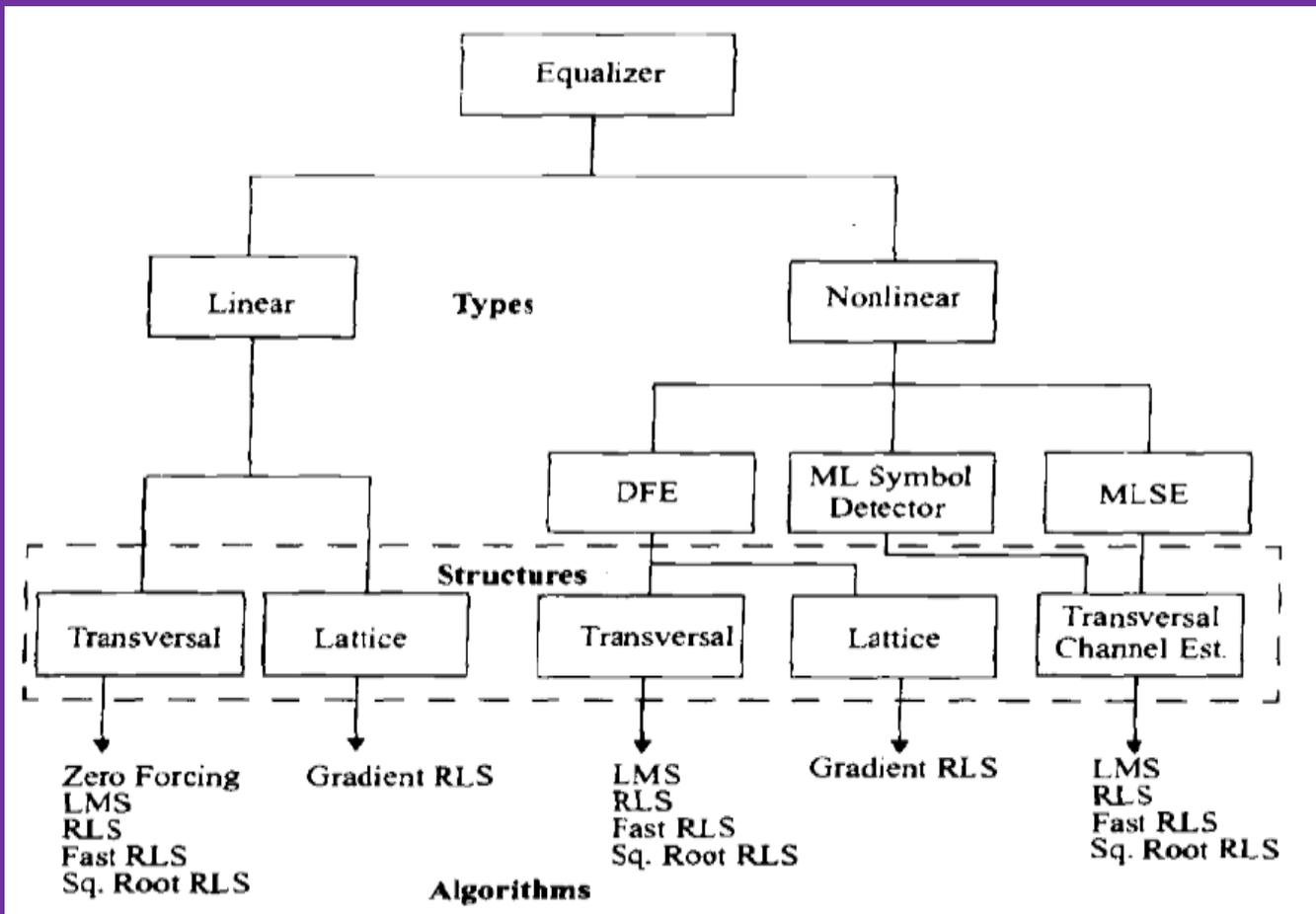
Linear Equalizers

- A linear equalizer can be implemented as an FIR filter, otherwise known as the transversal filter.
- This type of equalizer is the simplest type available. In such an equalizer, the current and past values of the received signal are linearly weighted by the filter coefficient and summed to produce the output.
- If the delays and the tap gains are analog, the continuous output of the equalizer is sampled at the symbol rate and the samples are applied to the decision device.
- The implementation is, however, usually carried out in the digital domain where the samples of the received signal are stored in a shift register.
- The output of this transversal filter before decision making (threshold detection)

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} (c_n{}^*) y_{k-n}$$

$$E[|e(n)|^2] = \frac{T}{2\pi} \int_{-\pi/T}^{\pi/T} \frac{N_0}{\left|F(e^{j\omega T})\right|^2 + N_0} d\omega$$
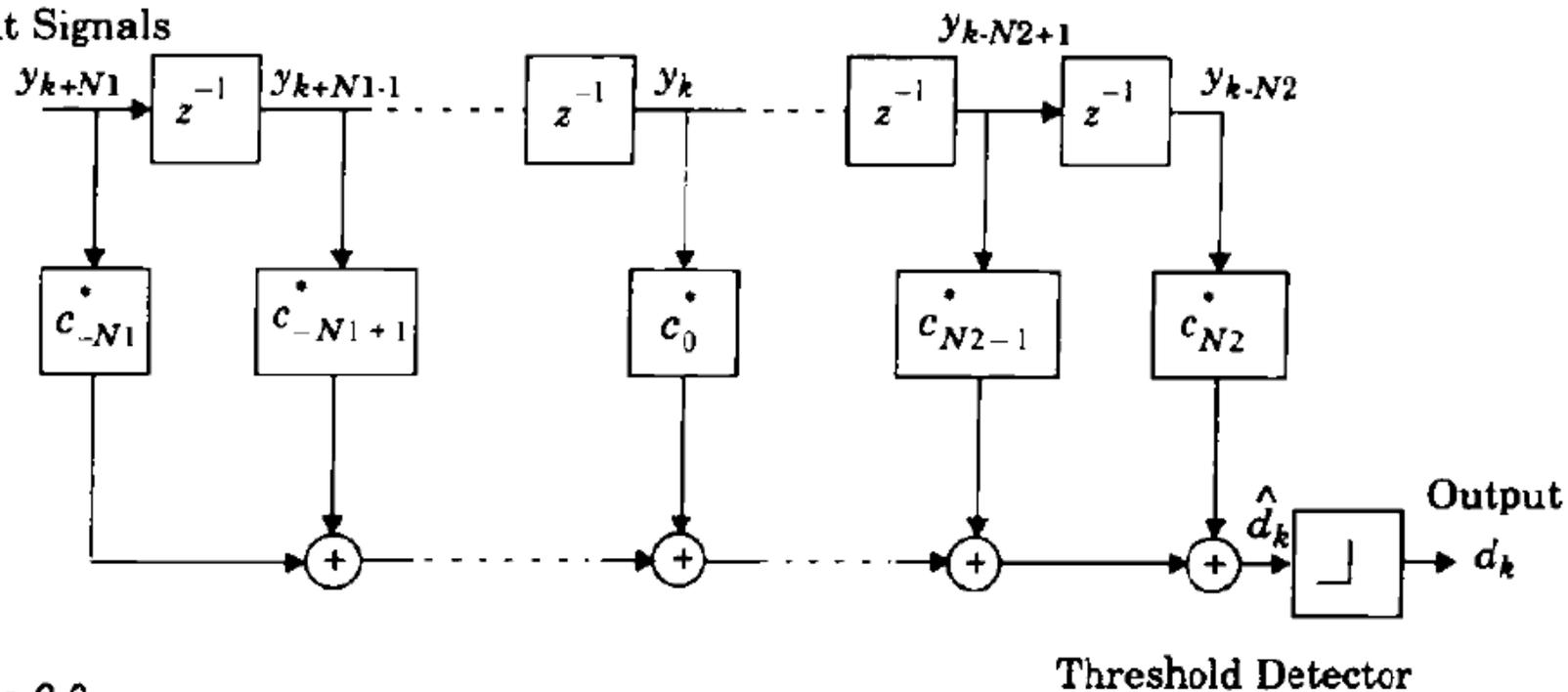
Figure 6.6

Nonlinear Equalization

- Nonlinear equalizers are used in applications where the channel distortion is too severe for a linear equalizer to handle.
- Linear equalizers do not perform well on channels which have deep spectral nulls in the pass band.
- In an attempt to compensate for the distortion, the linear equalizer places too much gain in the vicinity of the spectral null, thereby enhancing the noise present in those frequencies.
- Three very effective nonlinear methods have been developed which offer improvements over linear equalization techniques.
  1. Decision Feedback Equalization (DFE)
  2. Maximum Likelihood Symbol Detection
  3. Maximum Likelihood Sequence Estimation (MLSE)

Decision Feedback Equalization (OFE)

- The basic idea behind decision feedback equalization is that once an information symbol has been detected and decided upon, the ISI that it induces on future symbols can be estimated and subtracted out before detection of subsequent symbols.
- The DFE can be realized in either the direct transversal form or as a lattice filter.
- It consists of a feedforward filter (FFF) and a feedback filter (FBF). The FBF is driven by decisions on the output of the detector, and its coefficients can be adjusted to cancel the ISI on the current symbol from past detected symbols.

$$\hat{d}_k = \sum_{n=-N_1}^{N_2} c_n^* y_{k-n} + \sum_{i=1}^{N_3} F_i d_{k-i}$$

$$E\left[|e(n)|^2\right]_{min} = \exp\left\{\frac{T}{2\pi} \int_{-\pi/T}^{\pi/T} \ln\left[\frac{N_0}{|F(e^{j\omega T})|^2 + N_0}\right] d\omega\right\}$$
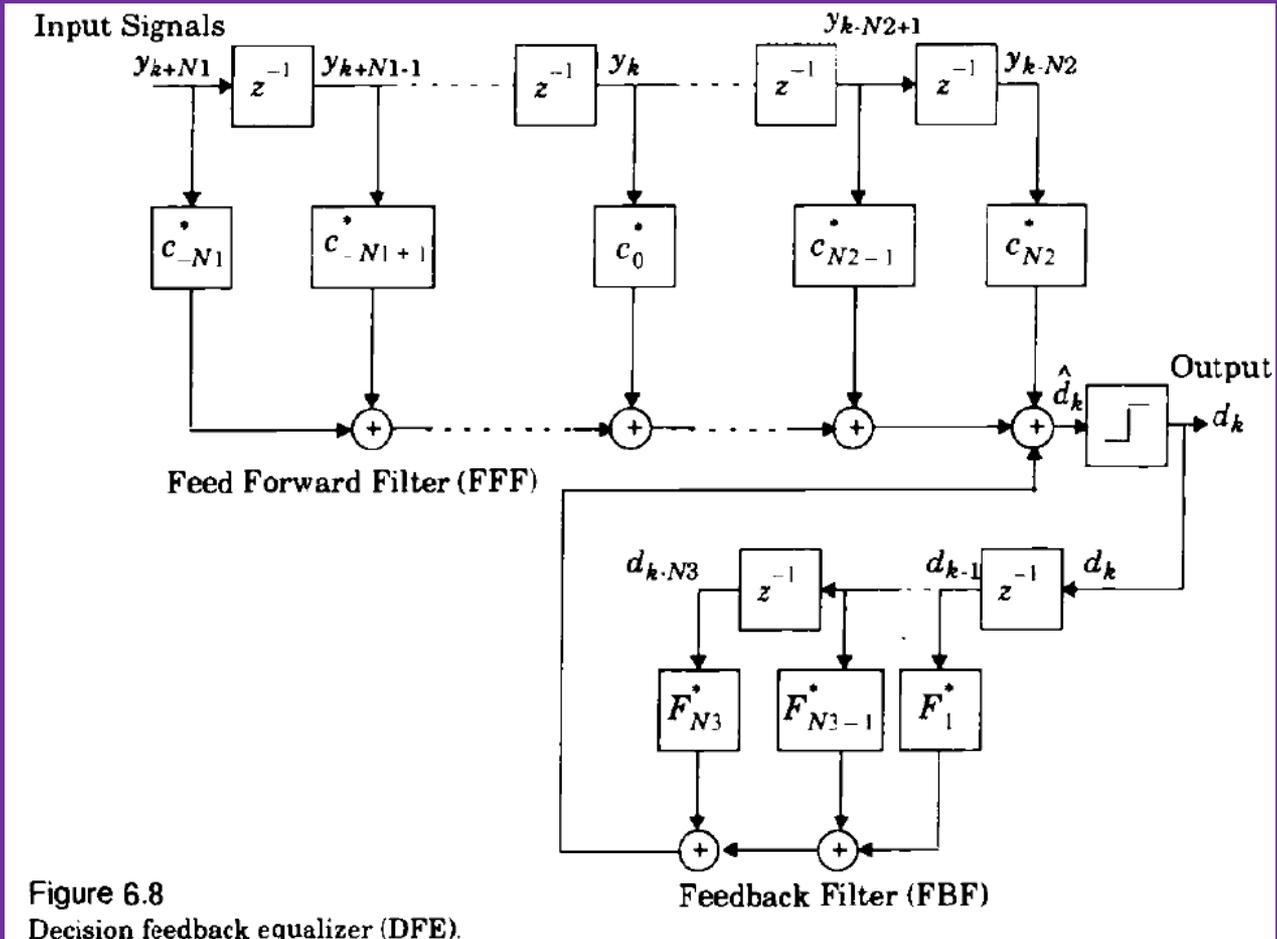
Figure 6.8
Decision feedback equalizer (DFE).

Maximum Likelihood Sequence Estimation (MLSE) Equalizer

- The MSE-based linear equalizers described previously are optimum with respect to the criterion of minimum probability of symbol error when the channel does not introduce any amplitude distortion.

- Yet this is precisely the condition in which an equalizer is needed for a mobile communications link.

- This limitation on MSE-based equalizers led researchers to investigate optimum or nearly optimum nonlinear structures.

- These equalizers use various forms of the classical maximum likelihood receiver structure. Using a channel impulse response simulator within the algorithm, the MLSE tests all possible data sequences and chooses the data sequence with the maximum probability as the output.

- This algorithm was recognized to be a maximum likelihood sequence estimator (MLSE) of the state sequences of a finite state Markov process observed in memoryless noise.

- It has recently been implemented successfully for equalizers in mobile radio channels.
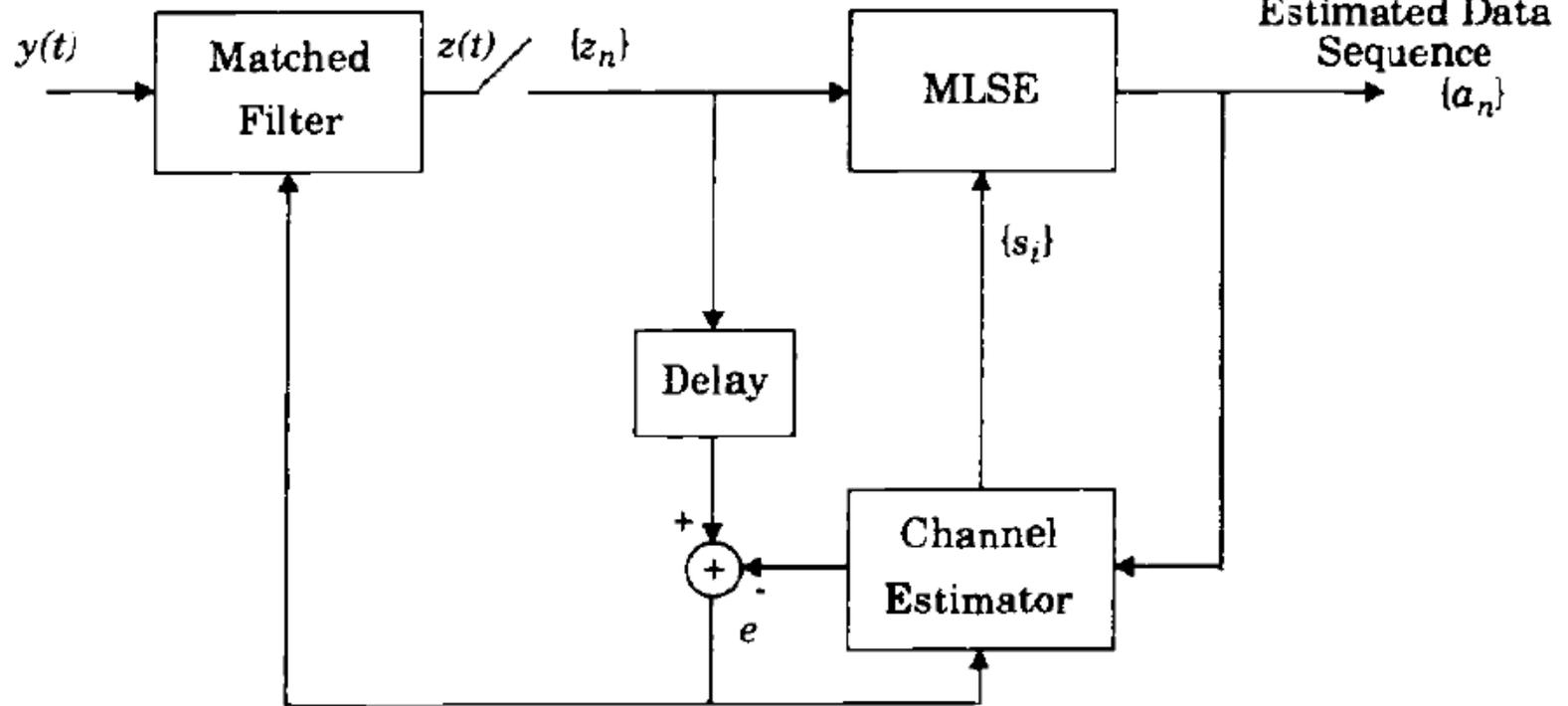
Algorithms for Adaptive Equalization

The performance of an algorithm is determined by various factors which include:

1. **Rate of convergence** — This is defined as the number of iterations required for the algorithm, in response to stationary inputs, to converge close enough to the optimum solution. A fast rate of convergence allows the algorithm to adapt rapidly to a stationary environment of unknown statistics. Furthermore, it enables the algorithm to track statistical variations when operating in a nonstationary environment

2. **Computational complexity** — This is the number of operations required to make one complete iteration of the algorithm.

3. **Numerical properties** — When an algorithm is implemented numerically, inaccuracies are produced due to round-oft' noise and representation errors inthe computer. These kinds of errors influence the stability of the algorithm.

Zero Forcing Algorithm

- In a zero forcing equalizer, the equalizer coefficients are chosen to force the samples of the combined channel and equalizer impulse response to zero at all but one of the spaced sample points in the tapped delay line filter.
- By letting the number of coefficients increase without bound, an infinite length equalizer with zero ISI at the output can be obtained, When each of the delay elements provide a time delay equal to the symbol duration , the frequency response Heq (f) of the equalizer is periodic with a period equal to the symbol rate I / T.
- The combined response of the channel with the equalizer must satisfy Nyquists first criterion where Hch (f) is the folded frequency response of the channel.
- Thus, an infinite length, zero, ISI equalizer is simply an inverse filter which inverts the folded frequency response of the channel.
- This infinite length equalizer is usually implemented by a truncated length version.

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

Least Mean Square Algorithm

- A more robust equalizer is the LMS equalizer where the criterion used is the minimization of the mean square error (MSE) between the desired equalizer output and the actual equalizer output.
- The LMS equalizer maximizes the signal to distortion ratio at its output within the constraints of the equalizer filter length.
- If an input signal has a time dispersion characteristic that is greater than the propagation delay through the equalizer, then the equalizer will be unable to reduce distortion.

The prediction error is given by

$$e_k = d_k - \hat{d}_k = x_k - \hat{d}_k$$

$$e_k = x_k - y_k^T w_k = x_k - w_k^T y_k$$

To compute the mean square error

$$\xi = E[e_k^* e_k]$$

The normal equation

$$R_{NN}\hat{w}_N = p_N$$

Recursive Least Squares Algorithm

- The convergence rate of the gradient-based LMS algorithm is very slow, especially when the eigen values of the input covariance matrix RNN have a very large spread.
- In order to achieve faster convergence, complex algorithms which involve additional parameters are used.
- Faster converging algorithms are based on a least squares approach, as opposed to the statistical approach used in the LMS algorithm.
- This leads to the family of powerful, complex, adaptive signal processing techniques known as recursive least squares (RLS), which significantly improves the convergence of adaptive equalizers.

Least square error based on the time average is

$$J(n) = \sum_{i=1}^{n} \lambda^{n-i} e^*(i,n) e(i,n)$$

The error e (i, n) is

$$e(i,n) = x(i) - y_N^T(i) w_N(n)$$

Optimal tap gain vector of the RLS equalizer

$$R_{NN}(n) = \sum_{i=1}^{n} \lambda^{n-i} y_N^*(i) y_N^T(i)$$

$$p_N(n) = \sum_{i=1}^{n} \lambda^{n-i} x^*(i) y_N(i)$$

1. Initialize $w(0) = k(0) = x(0) = 0$, $\boldsymbol{R}^{-1}(0) = \delta\boldsymbol{I}_{NN}$, where $\boldsymbol{I}_{NN}$ is an $N \times N$ identity matrix, and δ is a large positive constant.

2. Recursively compute the following:

$$\hat{d}(n) = \boldsymbol{w}^T(n-1)\boldsymbol{y}(n)$$

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

$$\boldsymbol{k}(n) = \frac{\boldsymbol{R}^{-1}(n-1)\boldsymbol{y}(n)}{\lambda + \boldsymbol{y}^T(n)\boldsymbol{R}^{-1}(n-1)\boldsymbol{y}(n)}$$

$$\boldsymbol{R}^{-1}(n) = \frac{1}{\lambda}[\boldsymbol{R}^{-1}(n-1) - \boldsymbol{k}(n)\boldsymbol{y}^T(n)\boldsymbol{R}^{-1}(n-1)]$$

$$\boldsymbol{w}(n) = \boldsymbol{w}(n-1) + \boldsymbol{k}(n)e^*(n)$$

Algorithm	Number of Multiply Operations	Advantages	Disadvantages
LMS Gradient DFE	$2N + 1$	Low computational complexity, simple program	Slow convergence, poor tracking
Kalman RLS	$2.5N^2 + 4.5N$	Fast convergence, good tracking ability	High computational complexity
FTF	$7N + 14$	Fast convergence, good tracking, low computational complexity	Complex programming, unstable (but can use rescue method)
Gradient Lattice	$13N - 8$	Stable, low computational complexity, flexible structure	Performance not as good as other RLS, complex programming
Gradient Lattice DFE	$13N_1 + 33N_2 - 36$	Low computational complexity	Complex programming
Fast Kalman DFE	$20N + 5$	Can be used for DFE, fast convergence and good tracking	Complex programming, computation not low, unstable
Square Root RLS DFE	$1.5N^2 + 6.5N$	Better numerical properties	High computational complexity
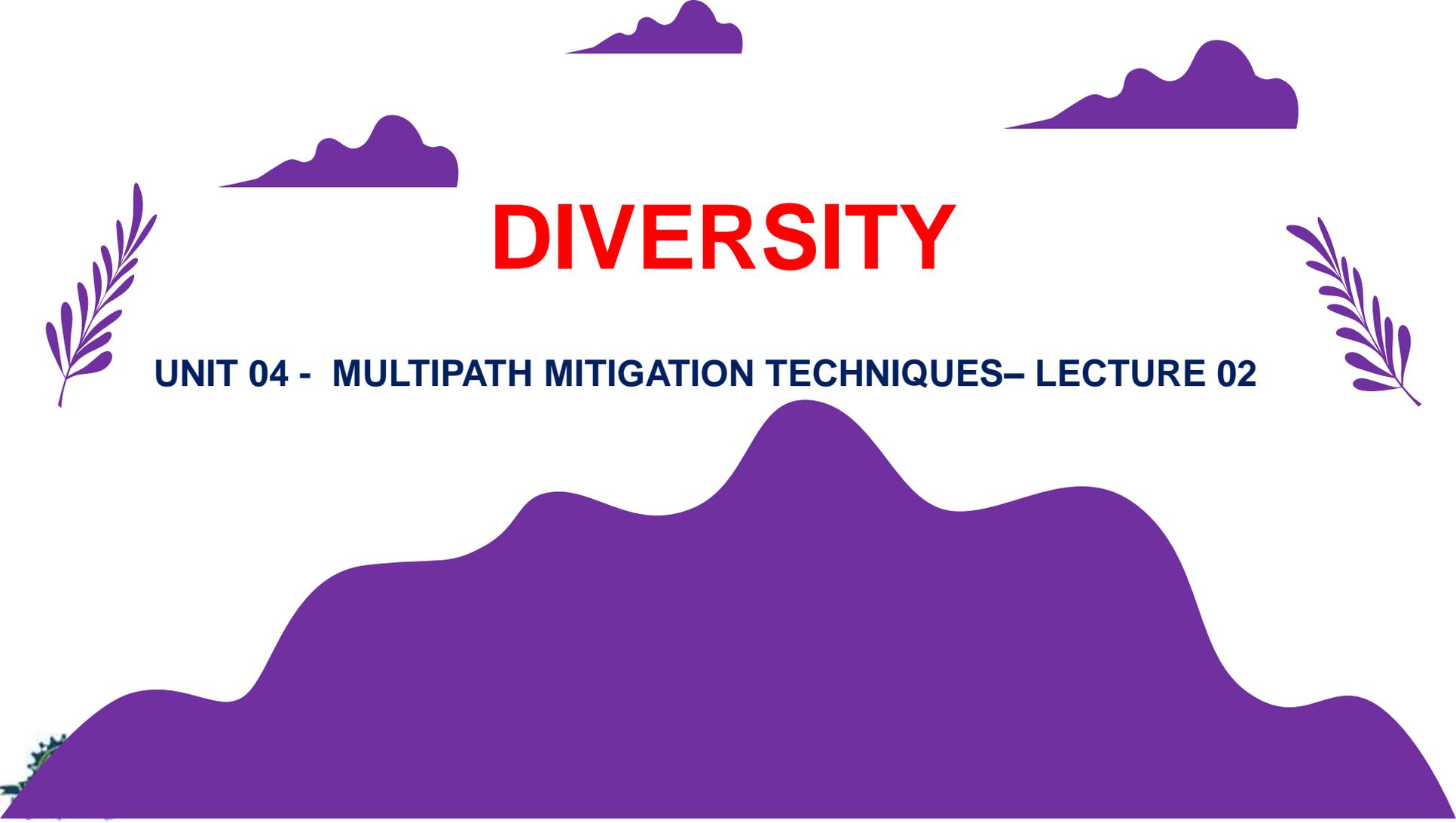
DIVERSITY

UNIT 04 - MULTIPATH MITIGATION TECHNIQUES– LECTURE 02

Diversity Techniques

- Diversity is a powerful communication receiver technique that provides wireless link improvement at relatively low cost. Unlike equalization, diversity requires no training overhead since a training sequence is not required by the transmitter.
- Furthermore, there are a wide range of diversity implementations, many which are very practical and provide significant link improvement with little added cost.
- Diversity exploits the random nature of radio propagation by finding independent (or at least highly uncorrelated) signal paths for communication.
- In virtually all applications, diversity decisions are made by the receiver, and are unknown to the transmitter.
- If one radio path undergoes a deep fade, another independent path may have a strong signal. By having more than one path to select from, both the instantaneous and average SNRs at the receiver may be improved, often by as much as 20 dB to 30 dB.

- There are two types of fading — small-scale and large-scale fading. Small-scale fades are characterized by deep and rapid amplitude fluctuations which occur as the mobile moves over distances of just a few wavelengths.
- These fades are caused by multiple reflections from the surroundings in the vicinity of the mobile.
- Small-scale fading typically results in a Rayleigh fading distribution of signal strength over small distances.
- In order to prevent deep fades from occurring, **microscopic diversity** techniques can exploit the rapidly changing signal.
- Large-scale fading is caused by shadowing due to variations in both the terrain profile and the nature of the surroundings. In deeply shadowed conditions, the received signal strength at a mobile can drop well below that of free space.
- By selecting a base station which is not shadowed when others are, the mobile can improve substantially the average ratio on the forward link. This is called **macroscopic diversity**, since the mobile is taking advantage of large separations between the serving base stations.

Derivation of Maximal Ratio Combining Improvement

- In maximal ratio combining, the voltage signals ri from each of the M diversity branches are co-phased to provide coherent voltage addition and are individually weighted to provide optimal SNR.
- If each branch has gain then the resulting signal envelope applied to the detector is

$$r_M = \sum_{i=1}^{M} G_i r_i$$

Practical Space Diversity Considerations

- Space diversity, also known as antenna diversity, is one of the most popular forms of diversity used in wireless systems.
- Conventional cellular radio systems consist of an elevated base station antenna and a mobile antenna close to the ground. The existence of a direct path between the transmitter and the receiver is not guaranteed and the possibility of a number of scatterers in the vicinity of the mobile suggests a Rayleigh fading signal.
- The concept of antenna space diversity is also used in base station design. At each cell site, multiple base station receiving antennas are used to provide diversity reception.
- However, since the important scatterers are generally on the ground in the vicinity of the mobile, the base station antennas must be spaced considerably far apart to achieve decorrelation.
- Separations on the order of several tens of wavelengths are required at the base station. Space diversity can thus be used at either the mobile or base station, or both.

Space diversity reception methods can be classified into four categories
1. Selection diversity
2. Feedback diversity
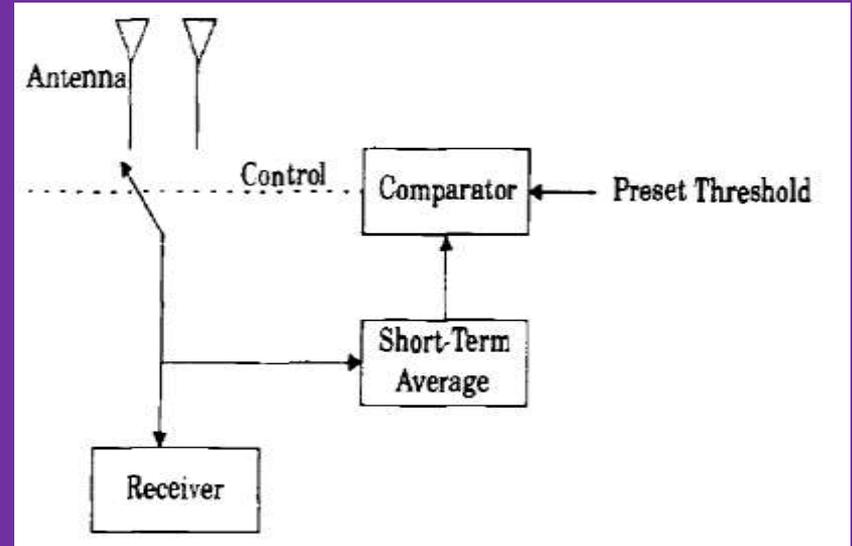3. Maximal ratio combining
4. Equal gain diversity
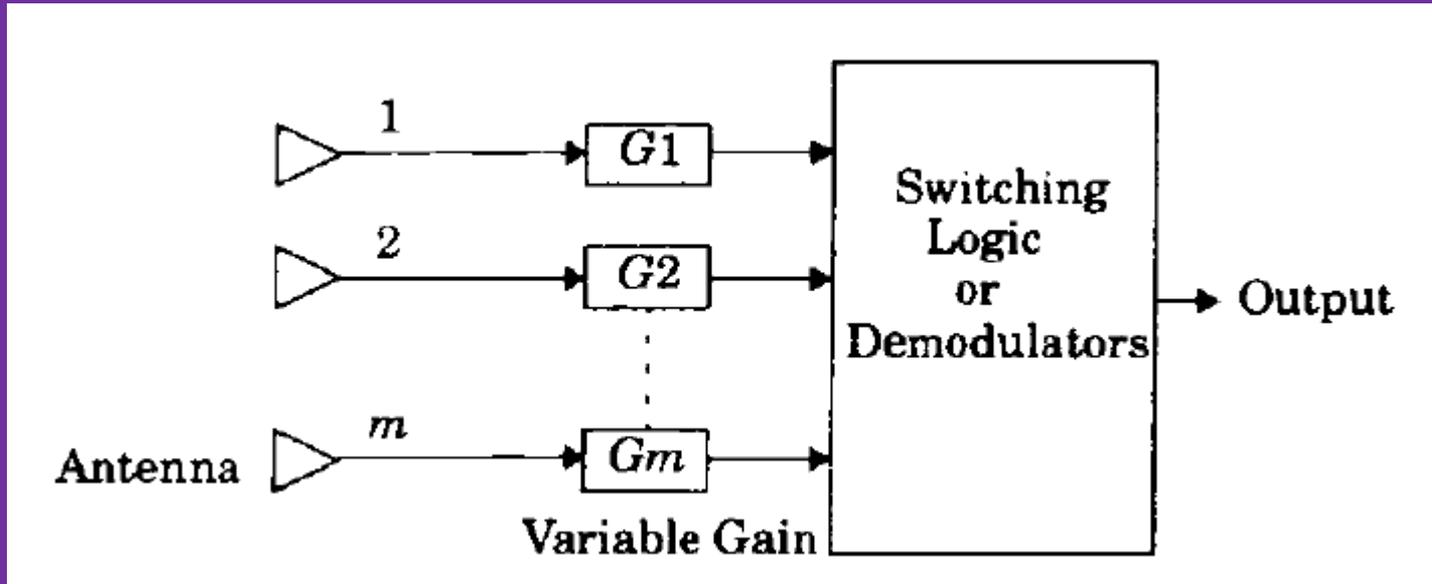
Selection Diversity

- Selection diversity is the simplest diversity technique
- Here m demodulators are used to provide m diversity branches whose gains are adjusted to provide the same average SNR for each branch.
- The receiver branch having the highest instantaneous SNR is connected to the demodulator.
- The antenna signals themselves could be sampled and the best one sent to a single demodulator.
- In practice, the branch with the largest (S + N) /N is used, since it is difficult to measure SNR.
- A practical selection diversity system cannot function on a truly instantaneous basis, but must be designed so that the internal time constants of the selection circuitry are shorter than the reciprocal of the signal fading rate.

Feedback or Scanning Diversity

- Scanning diversity is very similar to selection diversity except that instead of always using the best of M signals, the M signals are scanned in a fixed sequence until one is found to be above a predetermined threshold.
- This signal is then received until it falls below threshold and the scanning process is again initiated.
- The resulting fading statistics are somewhat inferior to those obtained by the other methods but the advantage with this method is that it is very simple to implement — only one receiver is required.

Maximal Ratio Combining

- In this method, the signals from all of the M branches are weighted according to their individual signal voltage to noise power ratios and then summed.
- Here, the individual signals must be co-phased before being summed (unlike selection diversity) which generally requires an individual receiver and phasing circuit for each antenna element.
- Maximal ratio combining produces an output SNR equal to the sum of the individual SNRs
- Thus, it has the advantage of producing an output with an acceptable SNR even when none of the individual signals are themselves acceptable.
- This technique gives the best statistical reduction of fading of any known linear diversity combiner.
- Modern DSP techniques and digital receivers are now making this optimal form of diversity practical.
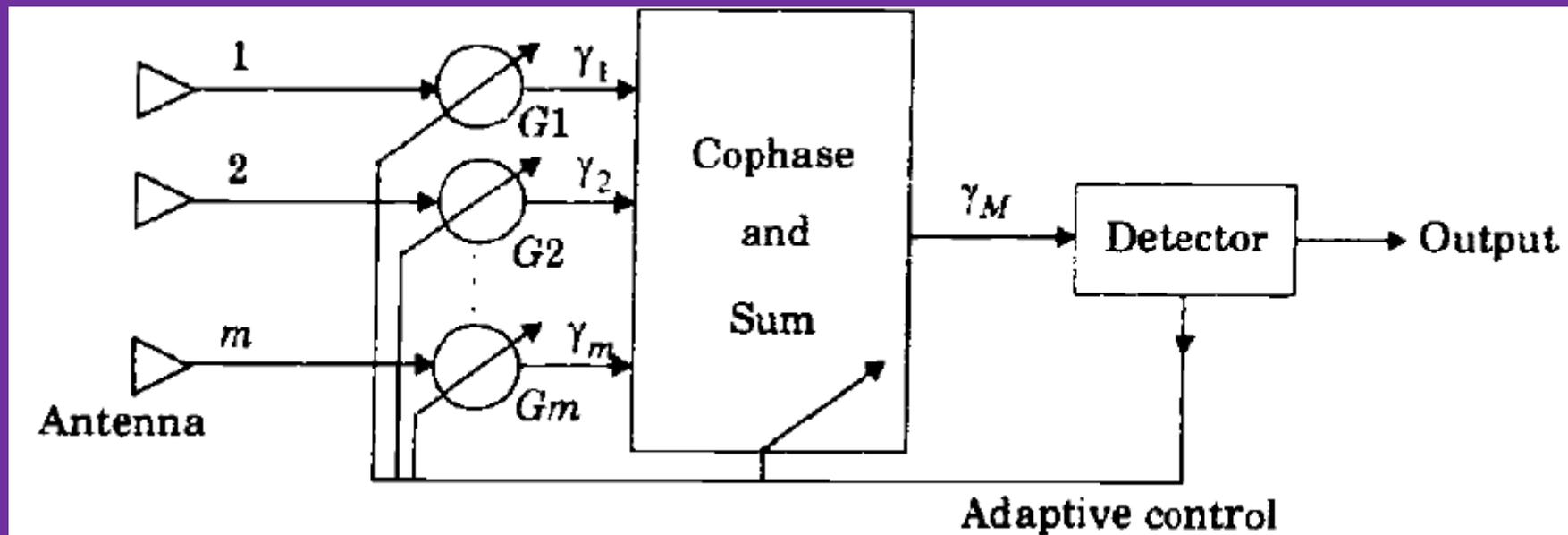
Equal Gain Combining

- In certain cases, it is not convenient to provide for the variable weighting capability required for true maximal ratio combining.
- In such cases, the branch weights are all set to unity but the signals from each branch are co-phased to provide equal gain combining diversity.
- This allows the receiver to exploit signals that are simultaneously received on each branch.
- The possibility of producing an acceptable signal from a number of unacceptable inputs is still retained, and performance is only marginally inferior to maximal ratio combining and superior to selection diversity.

Polarization Diversity

- At the base station, space diversity is considerably less practical than at the mobile because the narrow angle of incident fields requires large antenna spacings
- The comparatively high cost of using space diversity at the base station prompts the consideration of using orthogonal polarization to exploit polarization diversity.
- The decorrelation for the signals in each polarization is caused by multiple reflections in the channel between the mobile and base station antennas.
- The reflection coefficient for each polarization is different, which results in different amplitudes and phases for each, or at least some, of the reflections.
- After sufficient random reflections, the polarization state of the signal will be independent of the transmitted polarization.
- In practice, however, there is some dependence of the received polarization on the transmitted polarization.

(a) x-y plane

(b) x-z plane
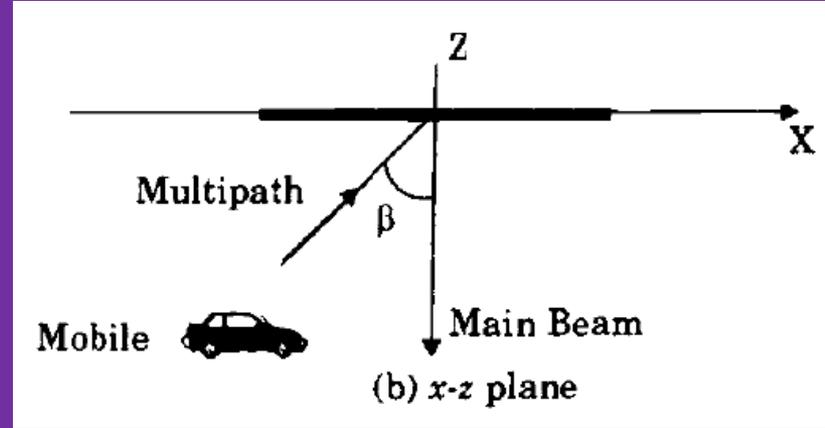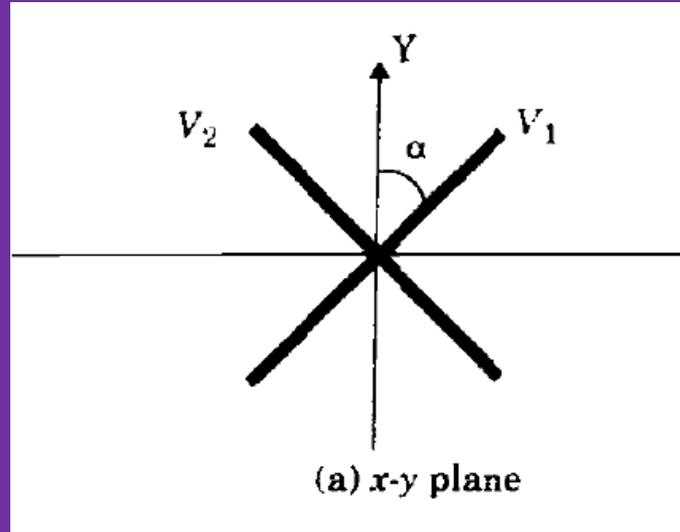
Frequency diversity

- Frequency diversity transmits information on more than one carrier frequency.
- The rationale behind this technique is that frequencies separated by more than the coherence bandwidth of the channel will not experience the same fades.
- Theoretically, if the channels are uncorrelated, the probability of simultaneous fading will be the product of the individual fading probabilities .
- Frequency diversity is often employed in microwave line-of-sight links which carry several channels in a frequency division multiplex mode (FDM).
- When diversity is needed, the appropriate traffic is simply switched to the backup frequency.
- This technique has the disadvantage that it not only requires spare bandwidth but also requires that there be as many receivers as there are channels used for the frequency diversity.
- However, for critical traffic, the expense may be justified.

Time Diversity

- Time diversity repeatedly transmits information at time spacings that exceed the coherence time of the channel, so that multiple repetitions of the signal will be received with independent fading conditions, thereby providing for diversity.
- One modem implementation of time diversity involves the use of the RAKE receiver for spread spectrum CDMA, where the multipath channel provides redundancy in the transmitted message

RAKE Receiver

- In CDMA spread spectrum systems , the chip rate is typically much greater than the flat fading bandwidth of the channel. Whereas conventional modulation techniques require an equalizer to undo the intersymbol interference between adjacent symbols, CDMA spreading codes are designed to provide very low correlation between successive chips.

- Thus, propagation delay spread in the radio channel merely provides multiple versions of the transmitted signal at the receiver.

- If these multipath components are delayed in time by more than a chip duration, they appear like uncorrelated noise at a CDMA receiver, and equalization is not required.

- However, since there is useful information in the multipath components. CDMA receivers may combine the time delayed versions of the original signal transmission in order to improve the signal to noise ratio at the receiver.

- A RAKE receiver does just this — it attempts to collect the time-shifted versions of the original signal by providing a separate correlation receiver for each of the multipath signals.
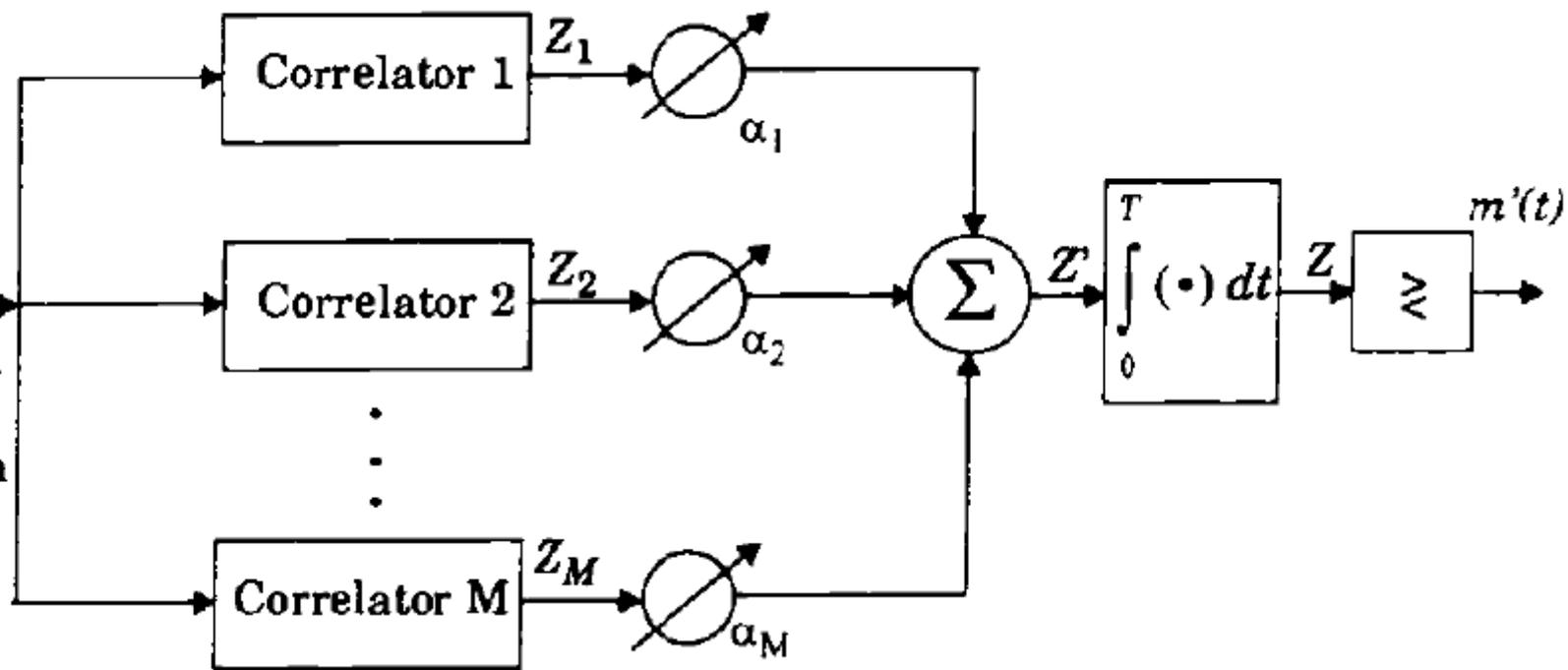
- A RAKE receiver utilizes multiple correlators to separately detect the M strongest multipath components.
- The outputs of each correlator are weighted to provide a better estimate of the transmitted signal than is provided by a single component.
- Demodulation and bit decisions are then based on the weighted outputs of the M correlators.
- In outdoor environments, the delay between multipath components is usually large and, if the chip rate is properly selected, the low autocorrelation properties of a CDMA spreading sequence can assure that multipath components will appear nearly uncorrelated with each other

The overall signal Z' is given by

$$Z' = \sum_{m=1}^{M} \alpha_m Z_m$$

The weighting coefficient

$$\alpha_m = \frac{Z_m^2}{\sum_{m=1}^{M} Z_m^2}$$

THANK YOU

Does anyone have any questions?

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