UNIT-III

Equalization: Fundamentals of equalization, Training a generic adaptive equalizer, Equalizers in a communication receiver, survey of equalization techniques, Linear equalizers, Nonlinear equalization, Decision feedback equalization (DFE), Maximum likelihood sequence estimation (MLSE) equalizer.

Diversity Techniques: Practical space diversity considerations: Selection diversity, feedback or scanning diversity, maximum ratio combining (MRC), equal gain combining (EGC), Polarization diversity, Frequency diversity, Time diversity, Rake receiver.

Source: Theodore S. Rappaport, Wireless Communications Principles and Practice, 2nd Edition, Pearson Education, 2003.
W.C.Y. Lee, Mobile Cellular Communications, 2nd Edition, Mc-Graw Hill, 1995
G Sasibhusan Rao, Mobile Cellular Communications, Pearson Education, 2013.

1. Introduction

- The mobile radio channel is particularly dynamic due to multipath fading and Doppler spread.
  - **Multipath fading -> time dispersion, ISI**
  - **Doppler spread -> dynamical fluctuation**

These effects have a strong negative impact on the bit error rate of any modulation.

- Mobile radio channel impairments cause the signal at the receiver to distort or fade significantly.
- **Mobile communication systems require signal processing techniques that improve the link performance in hostile mobile radio environments.**
- **Equalization, diversity, and channel coding** are three techniques which can be used independently or in jointly to improve received signal quality.

(i) Equalization

- **If the modulation bandwidth exceeds the coherence bandwidth of the radio channel, ISI occurs and modulation pulses are spread in time.**

- **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.

(ii) Diversity

- Diversity is used to compensate for fading channel impairments, and is usually implemented by using two or more receiving antennas.
- **Usually employed to reduce the depth and duration of the fades experienced by a receiver in a flat fading (narrowband) channel.**

  Without increasing the transmitted power or bandwidth.
- **Can be employed at both base station and mobile receivers.**
- **Types of diversity:**

  .antenna polarization diversity

  .frequency diversity

  .time diversity.

  For example, CDMA systems often use a RAKE receiver, which provides link improvement through time diversity
- **Spatial diversity is the most common one.**

  While one antenna sees a signal null, one of the other antennas may see a signal peak.

- The most common diversity technique is called spatial diversity, whereby multiple antennas are strategically spaced and connected to a common receiving system.

- The three techniques of equalization, diversity, and channel coding are used to improve radio link performance (i.e. to minimize the instantaneous bit error rate)
- but the approach, cost, complexity, and effectiveness of each technique varies widely in practical wireless communication systems.

2. Fundamentals of Equalization

- **Intersymbol interference (ISI)**
  - caused by multipath propagation (time dispersion) ;
  - cause bit errors at the receiver;
  - the major obstacle to high speed data transmission over mobile radio channels.

- **Equalization**
  - a technique used to combat ISI;
  - can be any signal processing operation that minimizes ISI;
  - usually track the varying channel adaptively.
  - 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.

Operating modes of an adaptive equalizer

- The general operating modes of an adaptive equalizer include training and tracking.

- **Training (first stage)**
  - **A known fixed-length training sequence is sent by the transmitter so that the receiver's equalizer may average to a proper setting.**
  - **The training sequence is designed to permit an equalizer at the receiver to acquire the proper filter coefficients in the worst possible channel conditions**

    The training sequence is typically a pseudorandom binary signal or a fixed, prescribed bit pattern.

    Immediately following the training sequence, the user data is sent.
  - **The time span over which an equalizer converges is a function of**
    1. **the equalizer algorithm**
    2. **the equalizer structure**
    3. **the time rate of change of the multipath radio channel.**

  Equalizers require periodic retraining in order to maintain effective ISI cancellation.

- **Tracking (second stage)**

  Immediately following the training sequence, the user data is sent.

  - **As user data are received, the adaptive algorithm of the equalizer tracks the changing channel and adjusts its filter characteristics over time.**

  - **commonly used in digital communication systems**

    where user data is segmented into short time blocks.

  - **TDMA wireless systems are particularly well suited for equalizers.**

    data in fixed-length time blocks,

    training sequence usually sent at the beginning of a block

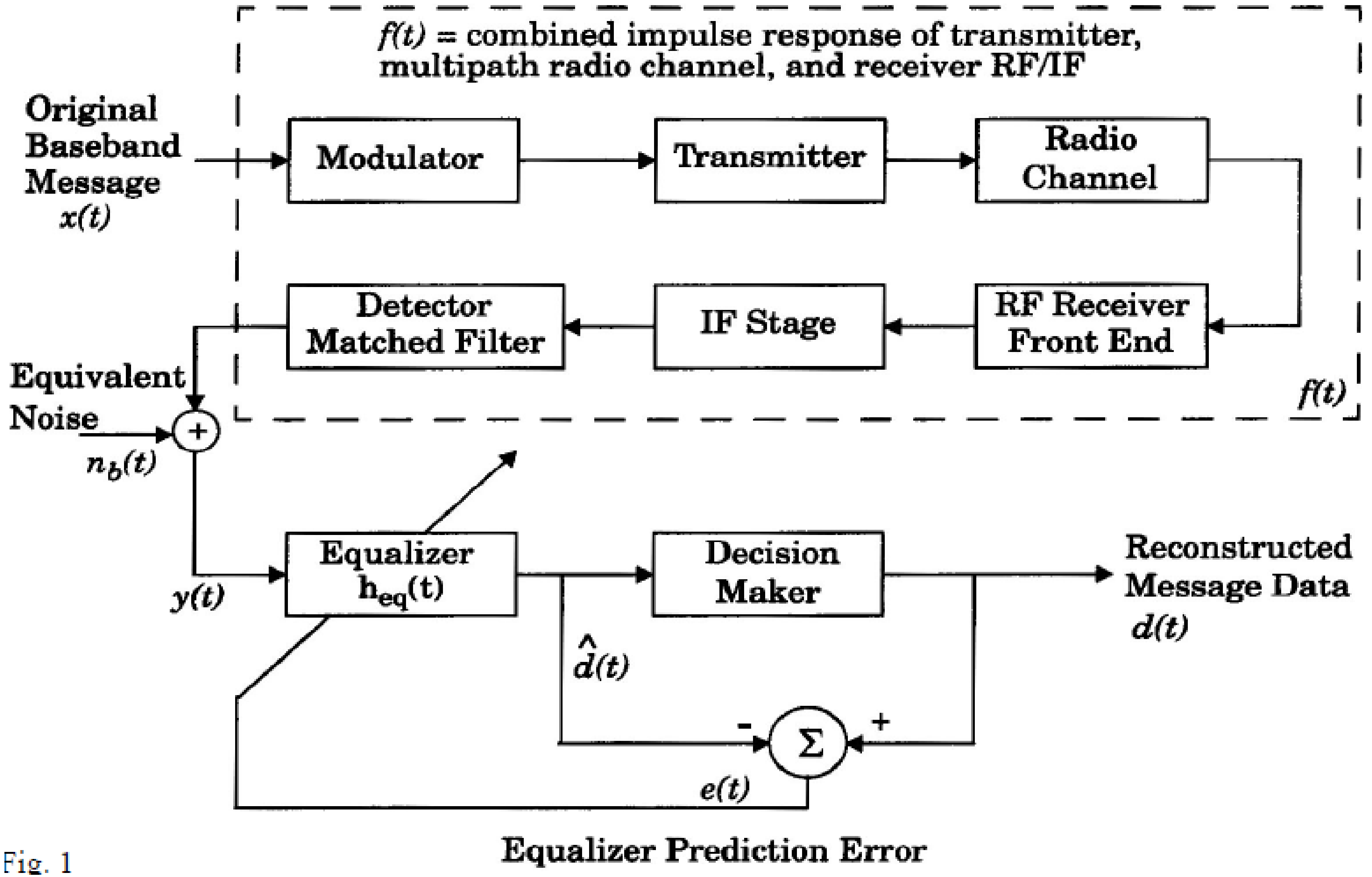


Fig. 1

Fig. Simplified communications system using an adaptive equalizer at the receiver.

- Equalizer can be implemented at baseband or at IF in a receiver.
- If $x(t)$ is the original information signal, and $f(t)$ is the combined complex baseband impulse response of the transmitter, channel, and the RF/IF sections of the receiver.
- Then the signal received by the equalizer may be expressed as:

$$y(t) = x(t) \otimes f^*(t) + n_b(t)$$

$f^*(t)$: complex conjugate of $f(t)$

$n_b(t)$: baseband noise at the input of the equalizer

$h_{eq}(t)$: impulse response of the equalizer

- If the impulse response of the equalizer is $h_{eq}(t)$ then the output of the equalizer is: $\hat{d}(t) = y(t) \otimes h_{eq}(t)$

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

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

where

$g(t) = f^*(t) \otimes h_{eq}(t)$: is the combined impulse response of the transmitter, channel, RF/IF sections of the receiver, and the equalizer.

- The complex baseband impulse response of a transversal filter equalizer is given by

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

where c_n are the complex filter coefficients of the equalizer.

- The desired output of the equalizer is original source data $x(t)$.

- Assume that $n_b(t) = 0$. Then, in order to force $\hat{d}(t) = x(t)$, the $g(t)$ must be equal to:

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

The goal of equalization is to satisfy this equation

- In the frequency domain, the above equation can be expressed as

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

This indicates that an equalizer is actually an inverse filter of the channel.

where $H_{eq}(f)$ and $F(f)$ are Fourier transforms of $h_{eq}(t)$ and $f(t)$, respectively.

- If the channel is frequency selective, the equalizer enhances the frequency components with small amplitudes and attenuates the strong frequencies in the received frequency spectrum in order to provide a flat, composite, received frequency response and linear phase response.

- For a time-varying channel, an adaptive equalizer is designed to track the channel variations so that equation $H_{eq}(f)F^*(-f) = 1$ is approximately satisfied.

3. A Generic Adaptive Equalizer

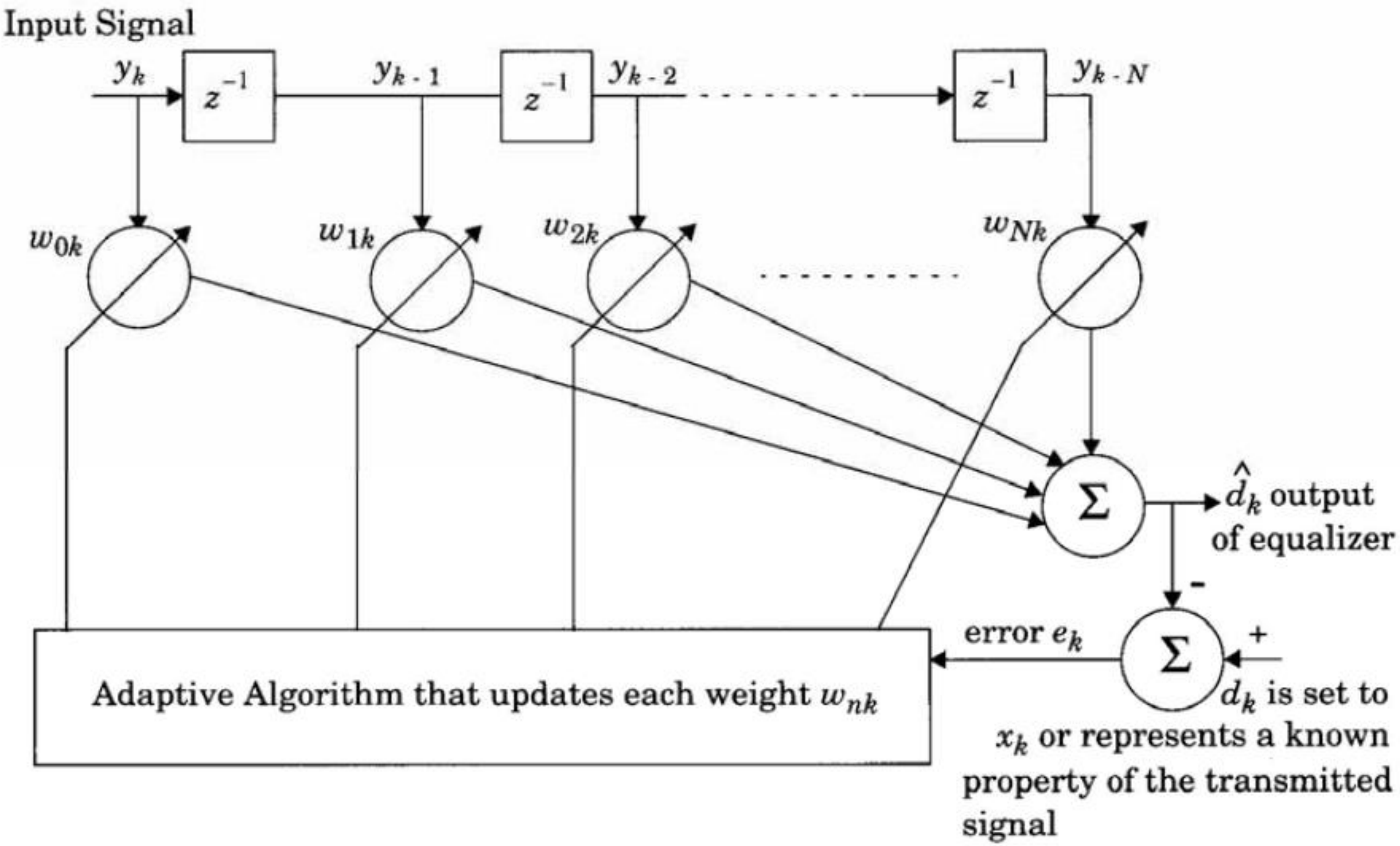


Figure: A basic linear equalizer during training.

- An adaptive equalizer is a time-varying filter which must constantly be retuned.
- In Figure there is a single input y_k at any time instant. The value of y_k depends upon the instantaneous state of the radio channel and the specific value of the noise. So, y_k is a random process.
- The adaptive equalizer structure shown above is called a transversal filter.

- **A transversal filter with**
  - **N delay elements**
  - **N+1 taps**
  - **N+1 tunable complex multipliers**
  - **N+1 weights:**
- **These weights are updated continuously by the adaptive algorithm**

  either on a sample by sample basis or on a block by block basis.
- **The adaptive algorithm is controlled by the error signal e_k.**

  e_k is derived by comparing the output of the equalizer with some signal which is either an exact scaled replica of the transmitted signal x_k or which represents a known property of the transmitted signal.

- The least mean squares (LMS) algorithm searches for the optimum or near-optimum filter weights by performing the following iterative operation:

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

where

Previous error = Previous desired output - Previous actual output

- The constant may be adjusted by the algorithm to control the variation between filter weights on successive iterations.

- This process is repeated rapidly in a programming loop while the equalizer attempts to converge.

- Upon reaching convergence, the adaptive algorithm freezes the filter weights until the error signal exceeds an acceptable level or until a new training sequence is sent.

- **Techniques used to minimize the error**
  - **gradient**
  - **steepest decent algorithms**

- **Based on classical equalization theory, the most common cost function is MSE**

  MSE----mean square error (MSE) between the desired signal and the output of the equalizer

  Denoted by $E[e(k) \cdot e^{*}(k)]$

Blind algorithms

- more recent class of adaptive algorithms
- able to exploit characteristics of the transmitted signal and do not require training sequences.

  provide equalizer convergence without burdening the transmitter with training overhead

  able to acquire equalization through property restoral techniques of the transmitted signal,

- Two techniques:
  - the constant modulus algorithm (CMA)

    used for constant envelope modulation

    forces the equalizer weights to maintain a constant envelope on the received signal
  - spectral coherence restoral algorithm (SCORE).

    exploits spectral redundancy or cyclostationarity in the transmitted signal

- The input signal to the equalizer is a vector y_k which is given by

$$\boldsymbol{y}_k = [y_k \quad y_{k-1} \quad y_{k-2} \quad \ldots \; y_{k-N}]^T \tag{1}$$

- The output of the adaptive equalizer is a scalar given by

$$\hat{d}_k = \sum_{n=0}^{N} w_{nk} \; y_{k-n} \tag{2}$$

- A weight vector can be written as

$$\boldsymbol{w}_k = [w_{0k} \quad w_{1k} \quad w_{2k} \quad \ldots \; w_{Nk}]^T \tag{3}$$

- Using (1) and (3), (2) may be written in vector notation as

$$\hat{d}_k = \boldsymbol{y}_k^T \; \boldsymbol{w}_k = \boldsymbol{w}_k^T \; \boldsymbol{y}_k \tag{4}$$

- It follows that when the desired equalizer output is known (i.e., $d_k = x_k$), the error signal e_k is given by

$$e_k = d_k - \hat{d}_k = x_k - \hat{d}_k \tag{5}$$

and from (4)

$$e_k = x_k - \boldsymbol{y}_k^T \, \boldsymbol{w}_k = x_k - \boldsymbol{w}_k^T \, \boldsymbol{y}_k \tag{6}$$

- The mean square error $|e_k|^2$ at time instant k is given by

$$|e_k|^2 = x_k^2 + \boldsymbol{w}_k^T \, \boldsymbol{y}_k \boldsymbol{y}_k^T \, \boldsymbol{w}_k - 2x_k \boldsymbol{y}_k^T \, \boldsymbol{w}_k \tag{7}$$

- Taking the expected value of $|e_k|^2$ over k yields

$$\boldsymbol{E}|e_k|^2 = \boldsymbol{E}[x_k^2] + \boldsymbol{w}_k^T \, \boldsymbol{E}[\boldsymbol{y}_k \boldsymbol{y}_k^T] \, \boldsymbol{w}_k - 2\boldsymbol{E}[x_k \boldsymbol{y}_k^T] \, \boldsymbol{w}_k \tag{8}$$

- The cross-correlation vector ***p*** between the desired response and the input signal is defined as

$$\boldsymbol{p} = \boldsymbol{E}[x_k \boldsymbol{y}_k] = \boldsymbol{E}[x_k y_k \quad x_k y_{k-1} \quad x_k y_{k-2} \quad \ldots \quad x_k y_{k-N}]^T \tag{9}$$

- The input correlation matrix is defined as the (N+1) x (N+1) square matrix ***R*** where

$$\boldsymbol{R} = \boldsymbol{E}[x_k \boldsymbol{y}_k^*] = \begin{bmatrix} y_k^2 & y_k y_{k-1} & y_k y_{k-2} & \ldots & y_k y_{k-N} \\ y_{k-1} y_k & y_{k-1}^2 & y_{k-1} y_{k-2} & \ldots & y_{k-1} y_{k-N} \\ \ldots & \ldots & \ldots & \ldots & \ldots \\ y_{k-N} y_k & y_{k-N} y_{k-1} & y_{k-N} y_{k-2} & \ldots & y_{k-N}^2 \end{bmatrix} \tag{10}$$

- The matrix ***R*** is sometimes called the input covariance matrix.
- The major diagonal of ***R*** contains the mean square values of each input sample, and the cross terms specify the autocorrelation terms resulting from delayed samples of the input signal.

- If x_k *and* $\boldsymbol{y}_k$ are stationary, then the elements in ***R*** and ***p*** are second order statistics which do not vary with time. Using equations (9) and (10), (8) may be rewritten as

$$\boldsymbol{Mean\ Sqare\ Error} = \boldsymbol{\xi} = \boldsymbol{E}\left[x_k^2\right] + \boldsymbol{w}^T\boldsymbol{Rw} - \boldsymbol{2p}^T\boldsymbol{w} \qquad (11)$$

- By minimizing $\boldsymbol{Mean\ Sqare\ Error}(\boldsymbol{\xi})$ in terms of the weight vector $\boldsymbol{w}_k$, it becomes possible to adaptively tune the equalizer to provide a flat spectral response (minimal ISI) in the received signal.

4. Equalizers in a Communications Receiver

$$y(t) = x(t) \otimes f^*(t) + n_b(t)$$

- Since the received signal $y(t)$ includes channel noise $n_b(t)$, an equalizer is unable to achieve perfect performance.
- Thus, there is always some residual ISI and some small tracking error.
- Therefore, the instantaneous combined frequency response will not always be flat, resulting in some finite prediction error.

- Because adaptive equalizers are implemented using digital logic, it is most convenient to represent all time signals in discrete form.
- Let T represent some increment of time between successive observations of signal states.
- Letting t = t_n=nT, where n is an integer.
- The output of the equalizer $\hat{d}(t)$ can be represented in discrete form as

$$\hat{d}(t) = x(t) \otimes g(t) + n_b(t) \otimes h_{eq}(t)$$

$$\hat{d}(n) = x(n) \otimes g(n) + n_b(n) \otimes h_{eq}(n)$$

- The prediction error $e(n)$ is

$$e(n) = d(n) - \hat{d}(n) = d(n) - [x(n) \otimes g(n) + n_b(n) \otimes h_{eq}(n)]$$

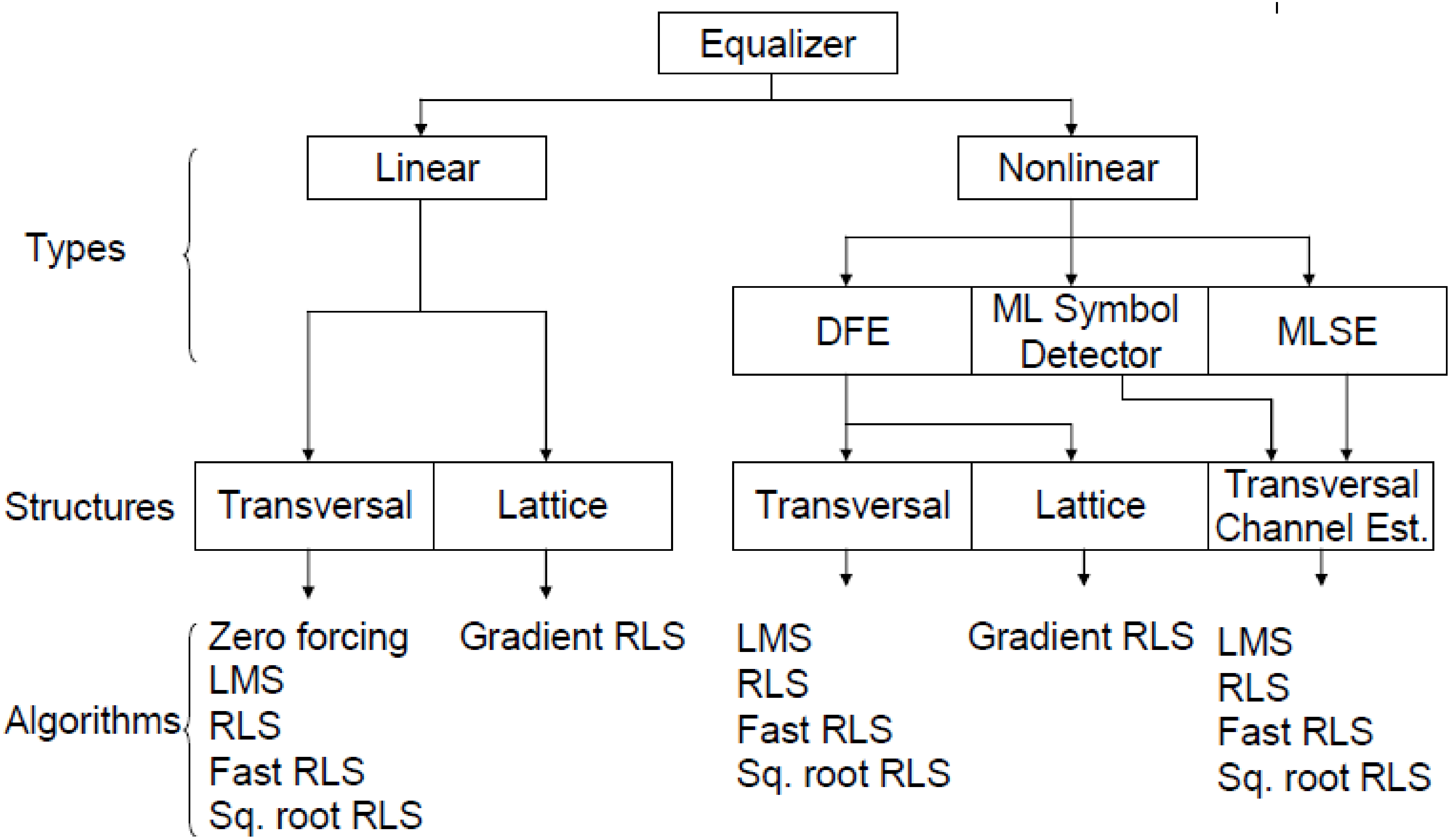
- The mean square error $|e_k|^2$ is one of the most important measures of how well an equalizer works.
- Minimizing the mean square error tends to reduce the bit error rate.
- For wireless communication links, it would be best to minimize the instantaneous probability of error $\boldsymbol{P_e}$ instead of MSE.
- But minimizing $\boldsymbol{P_e}$ generally results in nonlinear equations much more difficult to solve in real-time.

5. Survey of Equalization Techniques

- **Equalization techniques can be subdivided into two general categories:**
  - **linear equalization**
    - *The output of the decision maker is not used in the feedback path to adapt the equalizer.*
  - **nonlinear equalization**
    - *The output of the decision maker is used in the feedback path to adapt the equalizer.*

- **Many filter structures are used to implement linear and nonlinear equalizers**

- **For each structure, there are numerous algorithms used to adapt the equalizer.**


Equalizer
Linear
Nonlinear
Types
DFE
ML Symbol Detector
MLSE
Structures
Transversal
Lattice
Transversal
Lattice
Transversal Channel Est.
Algorithms
Zero forcing
LMS
RLS
Fast RLS
Sq. root RLS
Gradient RLS
LMS
RLS
Fast RLS
Sq. root RLS
Gradient RLS
LMS
RLS
Fast RLS
Sq. root RLS

- The most common equalizer structure is a linear transversal equalizer (LTE).

- A linear transversal filter is made up of tapped delay lines, with the tappings spaced a symbol period (*Ts*) apart, as shown in below Figure.

- Assuming that the delay elements have unity gain and delay *Ts*.

- The transfer function of a linear transversal equalizer can be written as a function of the delay operator $exp(-jwTs)$ or z^{-1}

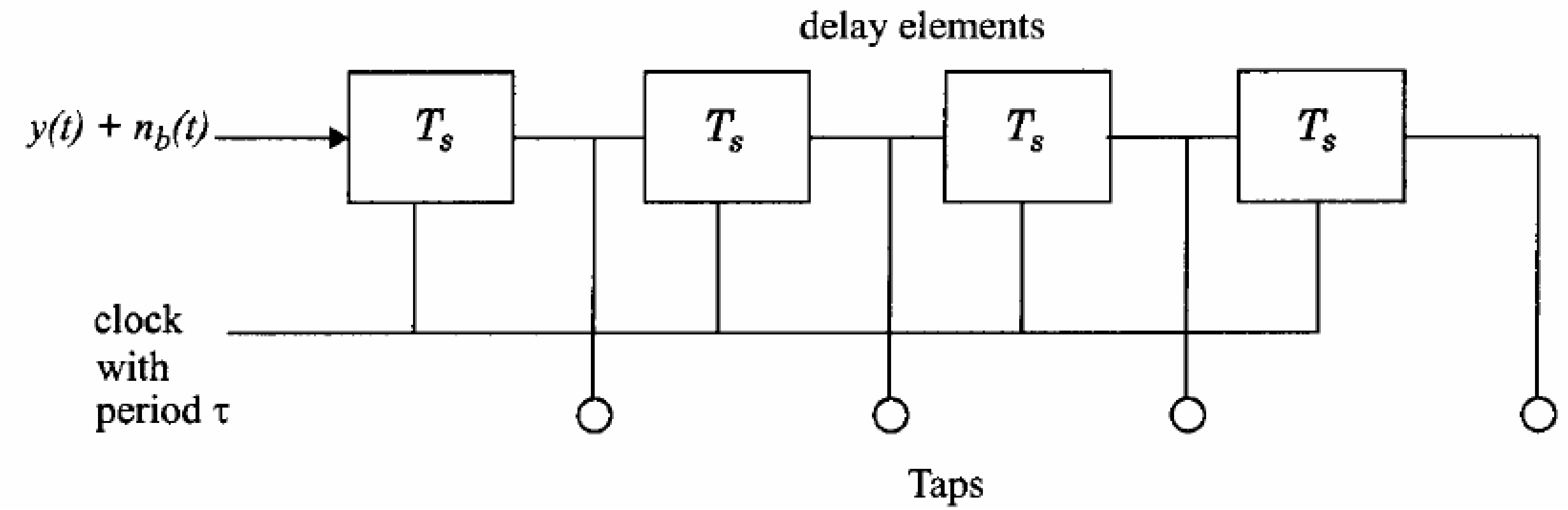


Fig. Basic linear transversal equalizer structure

Types of LTE structures:

1. finite impulse response (FIR) filter or transversal filter:
   - The equalizer uses only feedforward taps, and
   - Its transfer function of the equalizer filter is a polynomial in z^{-1}.
   - This filter has many zeroes but poles only at z = 0.

2. Infinite impulse response (FIR) filter or tapped delay line filter:
   - The equalizer has both feedforward and feedback taps,
   - Its transfer function is a rational function of z^{-1} ,
   - It has both poles and zeros.
   - Since IIR filters tend to be unstable when used in channels where the strongest pulse arrives after an echo pulse (i.e., leading echoes), they are rarely used.

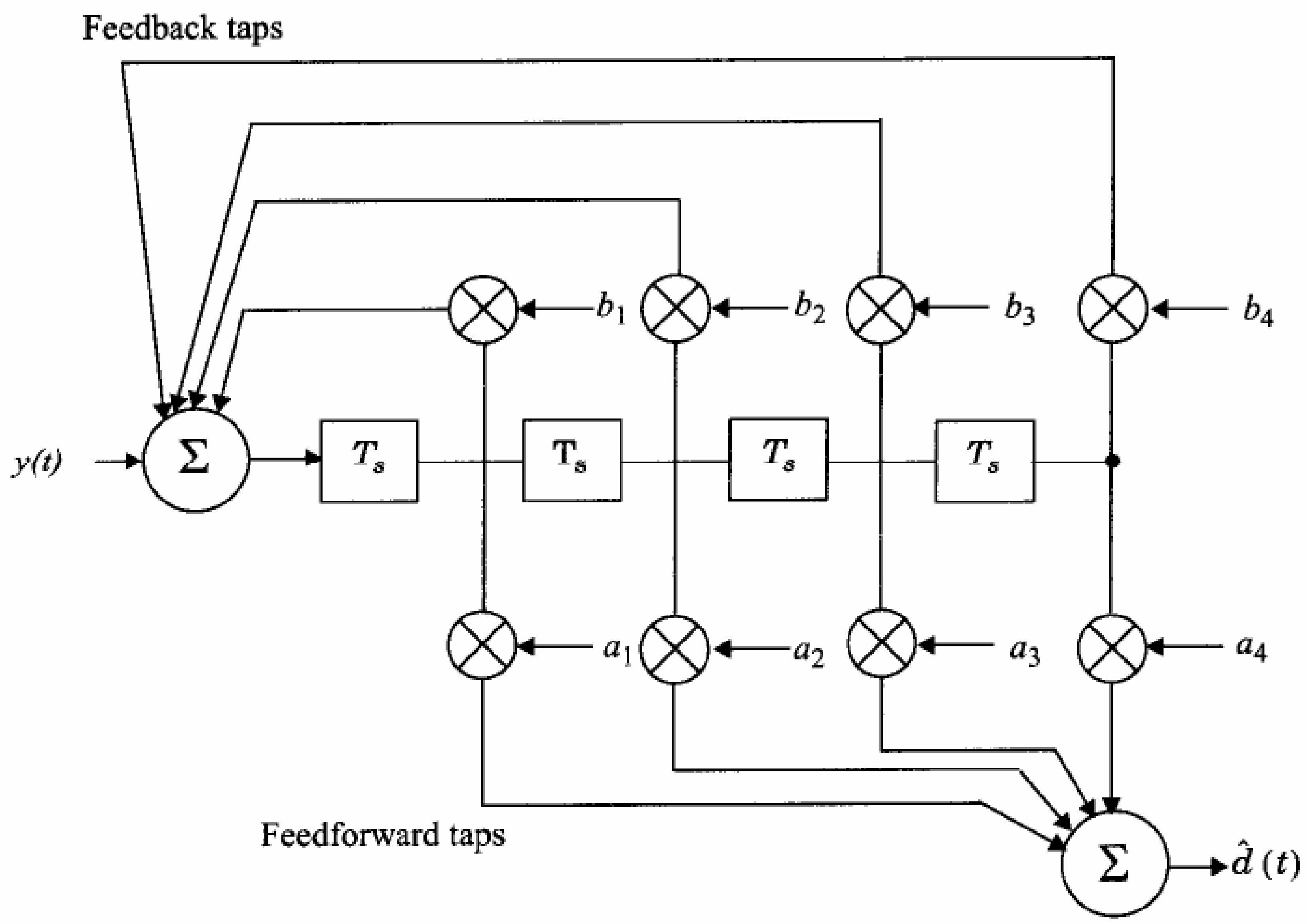


Figure: Tapped delay line filter with both feedforward and feedback taps (IIR)

6. Linear Equalizers

- A linear equalizer can be implemented as an FIR filter or transversal filter.
- This type of equalizer is the simplest type available.
- In this equalizer, the current and past values of the received signal are linearly weighted by the filter coefficient and summed to produce the output.

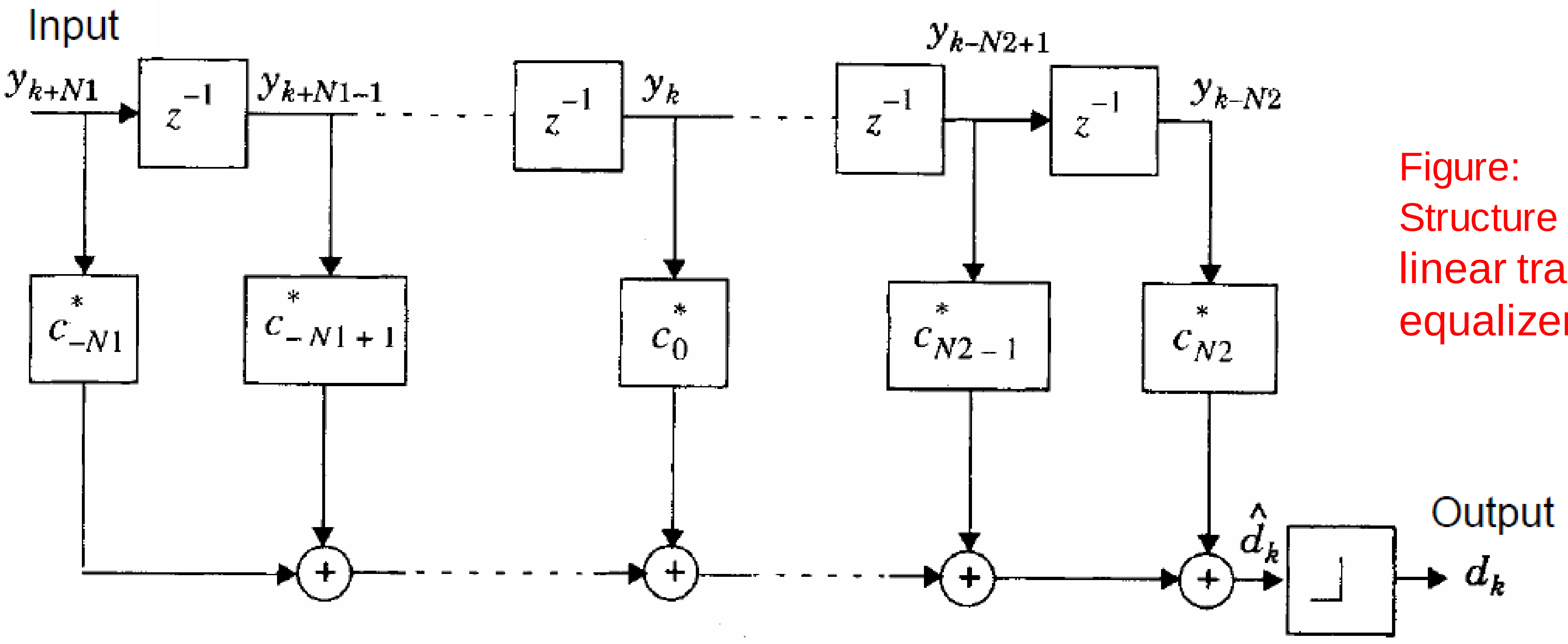


Figure: Structure of a linear transversal equalizer.

This type of equalizer is the simplest.

- 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.
- Implementation is 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) is:

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

where

$c_n{}^*$ is the complex filter coefficients or tap weights,
$\hat{d}_k$ is the output at time index k,
y_i is the input received signal at time $t_o + iT$
t_o is the equalizer starting time
$N = N_1 + N_2 + 1$ is the number of taps
N_1 the number of taps used in the forward path of the equalizer
N_2 the number of taps used in the reverse path of the equalizer

- The minimum mean squared error $E[|e(n)|^2]$ that a linear transversal equalizer can achieve is:

$$E[|e(n)|^2] = \frac{T}{2\pi} \int_{-\pi/T}^{\pi/T} \frac{N_o}{|F(e^{jwT})|^2 + N_o} dw$$

where

$F(e^{jwT})$ is the frequency response of the channel

N_o is the noise spectral density.

- The linear equalizer can also be implemented as a lattice filter,

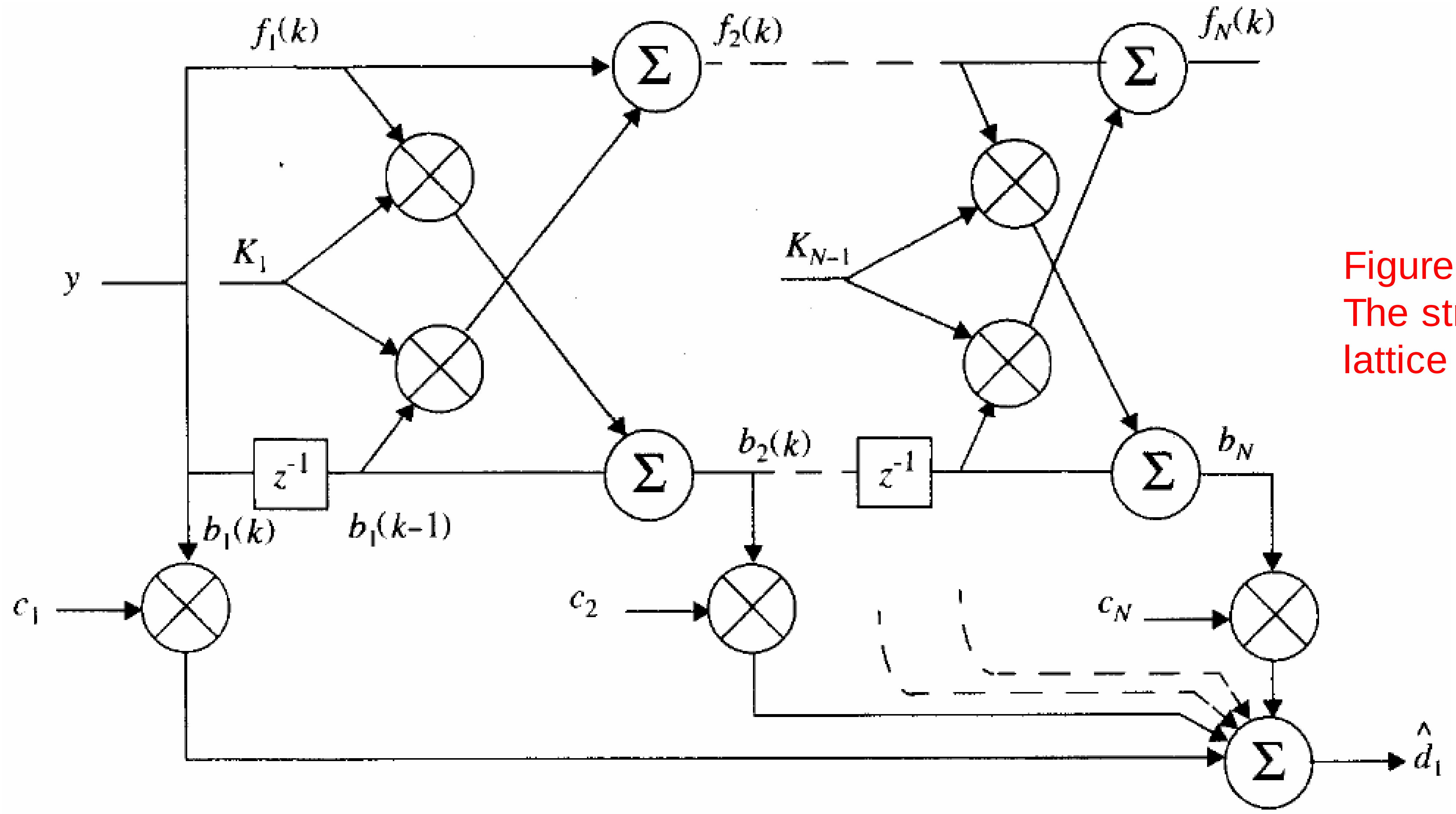


Figure:
The structure of a lattice equalizer

- The input signal y_k is transformed into a set of *N* intermediate forward and backward error signals, $f_n(k)$ and $b_n(k)$ respectively, which are used as inputs to the tap multipliers and are used to calculate the updated coefficients.
- Each stage of the lattice is then characterized by the following recursive equations

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

$$f_n(k) = y(k) - \sum_{i=1}^{n} K_i y(k-i) = f_{n-1}(k) + K_{n-1}(k)\, b_{n-1}(k-1)$$

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

where $K_n(k)$ is the reflection coefficient for the *n*th stage of the lattice.

- The backward error signals b_n, are then used as inputs to the tap weights, and the output of the equalizer is given by

$$\hat{d}_n = \sum_{n=1}^{N} c_n(k)\, b_n(k)$$

- Advantages of the lattice equalizer:
  - numerical stability
  - faster convergence

- Disadvantages of the lattice equalizer:
  - When the channel becomes more time dispersive, the length of the equalizer can be increased by the algorithm without stopping the operation of the equalizer.
  - More complicated than a linear transversal equalizer.

7. Nonlinear Equalization

- Linear equalizers do not perform well on channels which have deep spectral nulls in the passband.

- Nonlinear equalizers are used in applications where the channel distortion is too severe.

- Three very effective nonlinear equalizers:
  - **Decision Feedback Equalization (DFE)**
  - **Maximum Likelihood Symbol Detection**
  - **Maximum Likelihood Sequence Estimation (MLSE)**

7.1. Decision Feedback Equalization (DFE)

- Basic idea:

  Once an information symbol has been detected, the ISI that it induces on future symbols can be estimated and subtracted out before detection of subsequent symbols.

- DFE Can be realized in either the direct transversal form or as a lattice filter.

- The direct form 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.

- The equalizer has $N_1 + N_2 + 1$ taps in the feed forward filter and N_3 taps in the feedback filter.

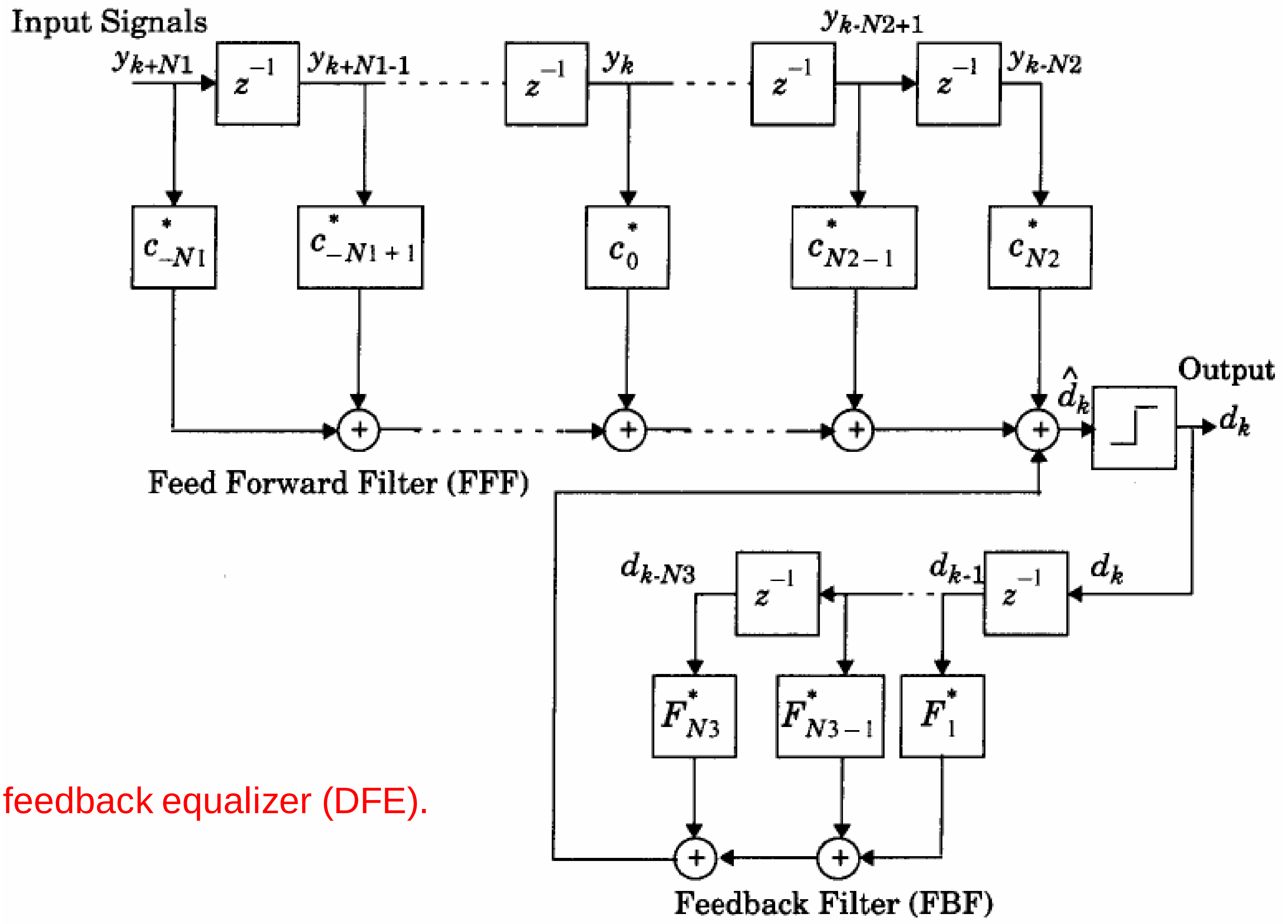


Figure:
Decision feedback equalizer (DFE).

- The equalizer output can be expressed as:

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

where
$c_n{}^*$ are the tap gains to the forward filter
y_n are the inputs to the forward filter
$f_i{}^*$ are tap gains for the feedback filter
d_i $(i < k)$ is the previous decision made on the detected signal.

- That is, once $\hat{d}_k$ is obtained using above equation, d_k is decided from it.

- Then, d_k along with previous decisions $d_{k-1}, d_{k-2}, \ldots.$ are fed back into the equalizer, and $\hat{d}_{k+1}$ is obtained using above equation.

- The minimum mean squared error that a DFE can achieve is:

$$E\left[|e(n)|^2\right]_{min} = exp\left\{\frac{T}{2\pi}\int_{-\pi/T}^{\pi/T} ln\left[\frac{N_o}{|F(e^{jwT})|^2 + N_o}\right] dw\right\}$$

- It can be seen that the minimum MSE for a DFE is always smaller than that of an LTE.

- An LTE is well behaved when the channel spectrum is comparatively flat.

- If the channel is severely distorted or exhibits nulls in the spectrum, the performance of an LTE deteriorates and the mean squared error of a DFE is much better than an LTE.

- **So, a DFE is more appropriate for severely distorted wireless channels.**

- The lattice implementation of the DFE is equivalent to a transversal DFE having a feed forward filter of length N_1 and a feedback filter of length N_2, where $N_1 > N_2$.
- Another form of DFE is called a predictive DFE.

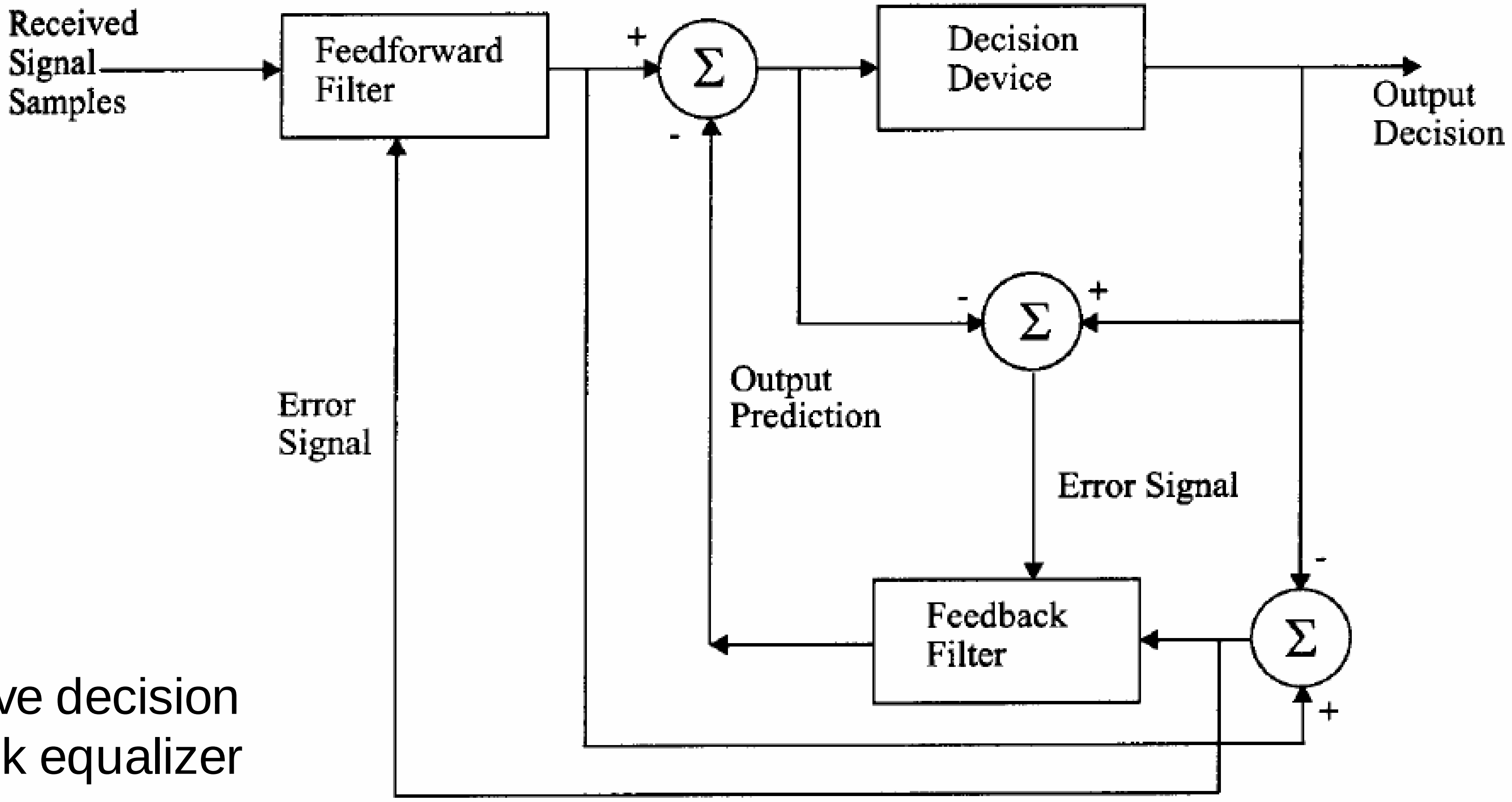


Figure:
Predictive decision feedback equalizer

- It consists of a feedforward filter (FFF) as in the conventional DFE.
- However, the feedback filter (FBF) is driven by an input sequence formed by the difference of the output of the detector and the output of the feed forward filter.
- Hence, the FBF here is called a noise predictor because it predicts the noise and the residual ISI contained in the signal at the FFF output and subtracts from it the detector output after some feedback delay.
- The FBF in the predictive DFE can also be realized as a lattice structure.
- The RLS lattice algorithm can be used in this case to yield fast convergence.

7.2 Maximum Likelihood Sequence Estimation (MLSE) equalizer

- MLSE tests all possible data sequences (rather than decoding each received symbol by itself) and chooses the data sequence with the maximum probability as the output.

- MLSE uses various forms of the classical maximum likelihood receiver structure.

- First proposed by Forney using a basic MLSE estimator structure and implementing it with the Viterbi algorithm

- **Drawback:** MLSE usually has a large computational requirement especially when the delay spread of the channel is large.

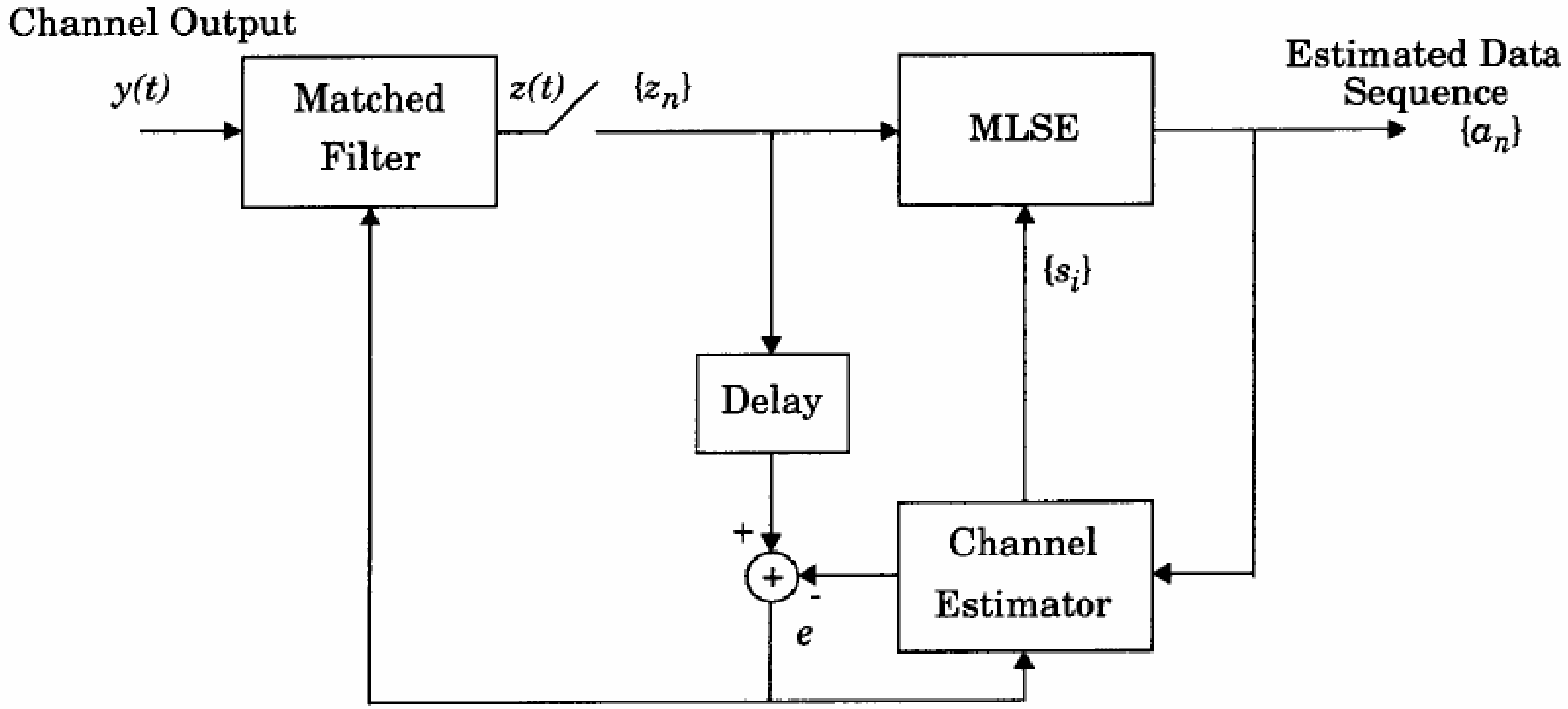


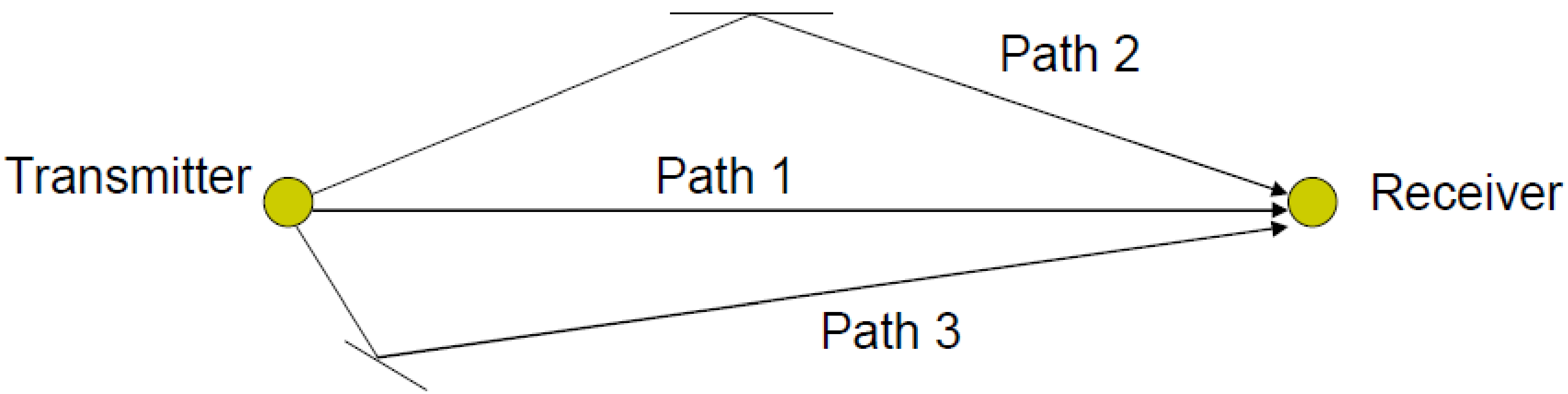
Figure: The structure of a maximum likelihood sequence equalizer(MLSE) with an adaptive matched filter

- MLSE requires knowledge of the channel characteristics in order to compute the matrics for making decisions.
- MLSE also requires knowledge of the statistical distribution of the noise corrupting the signal.

Part-II
Diversity Techniques

1. Introduction to 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.
- Diversity exploits (achieves) the random nature of radio propagation by finding independent signal paths for communication so as to boost the instantaneous SNR at the receiver.
- So, in diversity receivers, If one radio path undergoes a deep fade, another independent path may have a strong signal.



Two types of diversity:

I. **Microscopic diversity for mitigating small scale fading**

II. **Macroscopic diversity for mitigating large scale fading**

I. Microscopic diversity:

- Small-scale fades **represents deep and rapid amplitude fluctuations over distances of just a few wavelengths.**
- **These fades are** caused by multiple reflections from the surroundings in the vicinity of the mobile.
- These fades results in a Rayleigh fading distribution of signal strength over small distances.

- **Microscopic diversity techniques can exploit the rapidly changing signal.**
  - For example, use two antennas at the receiver (separated by a fraction of a meter), one may receive a null while the other receives a strong signal.
  - So, by selecting the best signal at all times, a receiver can mitigate small-scale fading effects. This is called antenna diversity or space diversity.
  - **Examples: Rake receiver, MIMO transmission**

II. Macroscopic diversity:

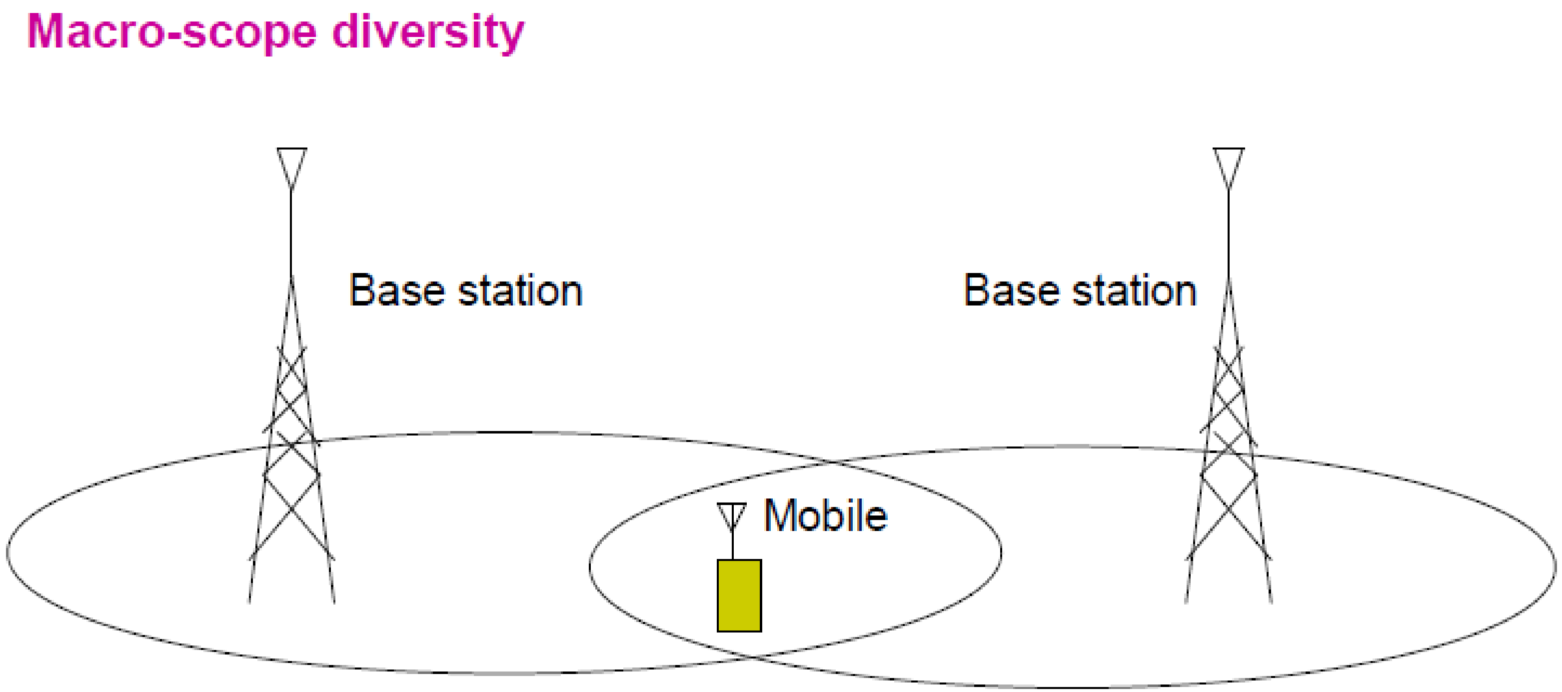
- **Large-scale fading** is caused by shadowing due to variations in both the terrain profile and the nature of the surroundings.

- In deeply shadowed conditions, the received signal strength at a mobile can drop well below that of free space.

- Large-scale fading can be represented with log-normally distributed with a standard deviation of about 10 dB in urban environments.

- By selecting a base station which is not shadowed, the mobile can improve substantially the average SNR ratio on the forward link. This is called Macroscopic diversity.

- Macroscopic diversity is also useful at the base station receiver.
- By using base station antennas that are sufficiently separated in space, the base station is able to improve the reverse link by selecting the antenna with the strongest signal from the mobile.



2. Practical space diversity considerations

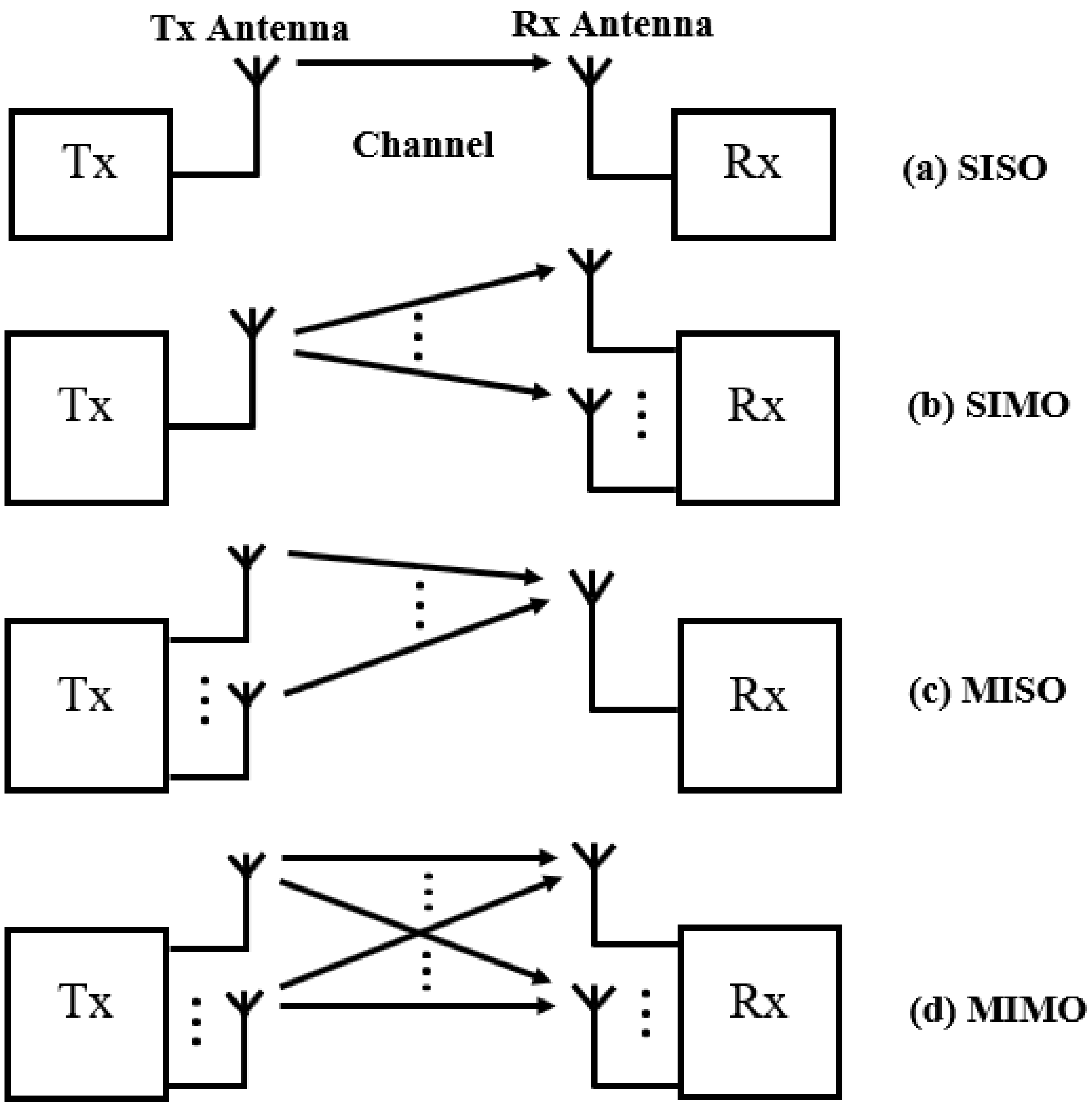
- In urban and indoor environments, there is no clear line-of-sight (LOS) between transmitter and receiver.
- So, the signal is reflected along multiple paths before finally being received.
- Each of these reflected signals can introduce phase shifts, time delays, attenuations, and distortions that can destructively interfere with one another at the aperture of the receiving antenna.
- Antenna diversity is especially effective at mitigating these multipath situations.

- **Antenna diversity**, also known as **space diversity** or **spatial diversity**, is any one of several wireless diversity schemes that uses two or more antennas to improve the quality and reliability of a wireless link.

- Space diversity (also known as antenna diversity), is one of the most popular forms of diversity used in wireless systems.

- The signals received from spatially separated antennas on the mobile would have essentially uncorrelated envelopes for antenna separations of one-half wavelength or more.

- Space diversity can be used at either the mobile or base station, or both.


Tx Antenna
Rx Antenna
Channel
Tx
Rx
(a) SISO
Tx
Rx
(b) SIMO
Tx
Rx
(c) MISO
Tx
Rx
(d) MIMO

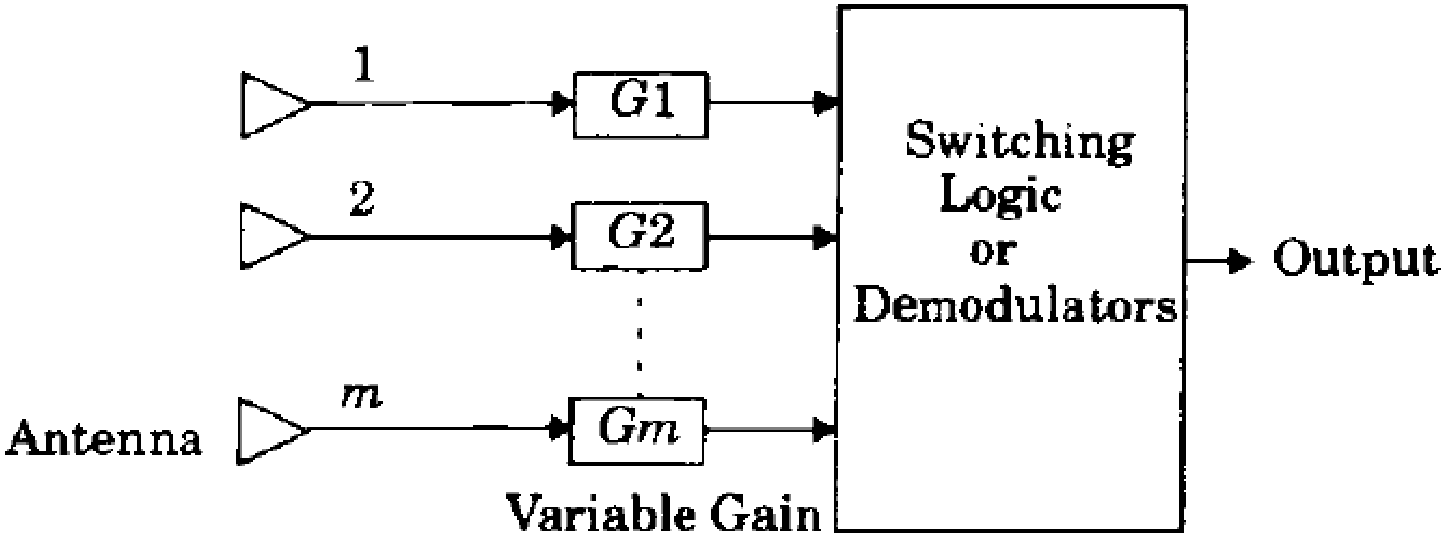


Fig. Generalized block diagram for space diversity

- Space diversity reception methods can be classified into four categories:

    i. Selection diversity

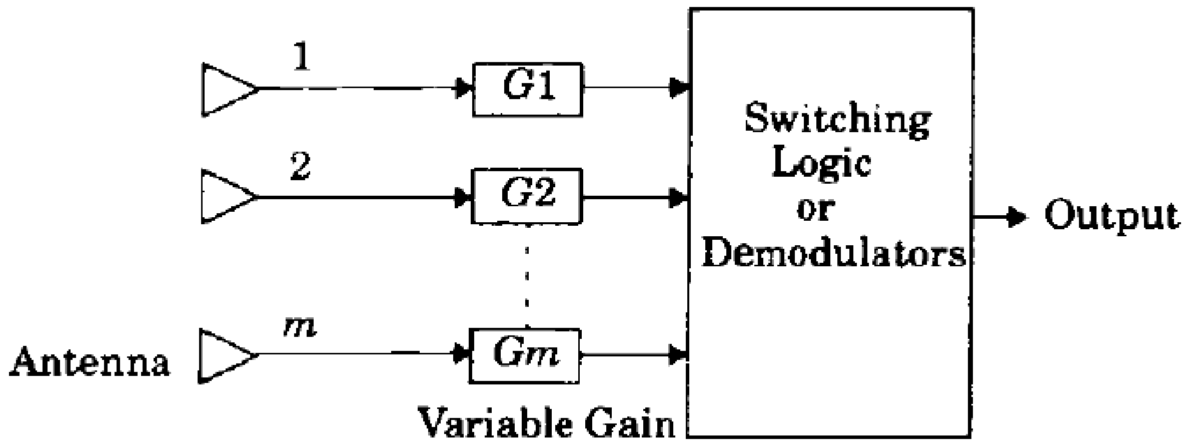
    ii. Feedback diversity

    iii. Maximal ratio combining

    iv. Equal gain diversity

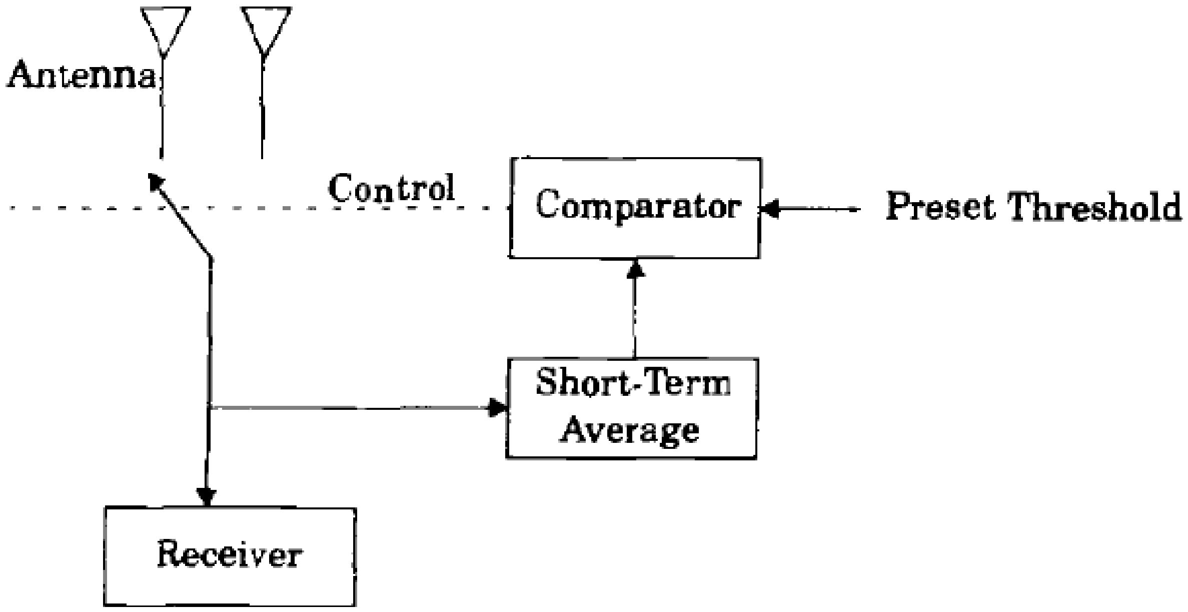
i. Selection Diversity:

- The simplest diversity technique.
- 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.



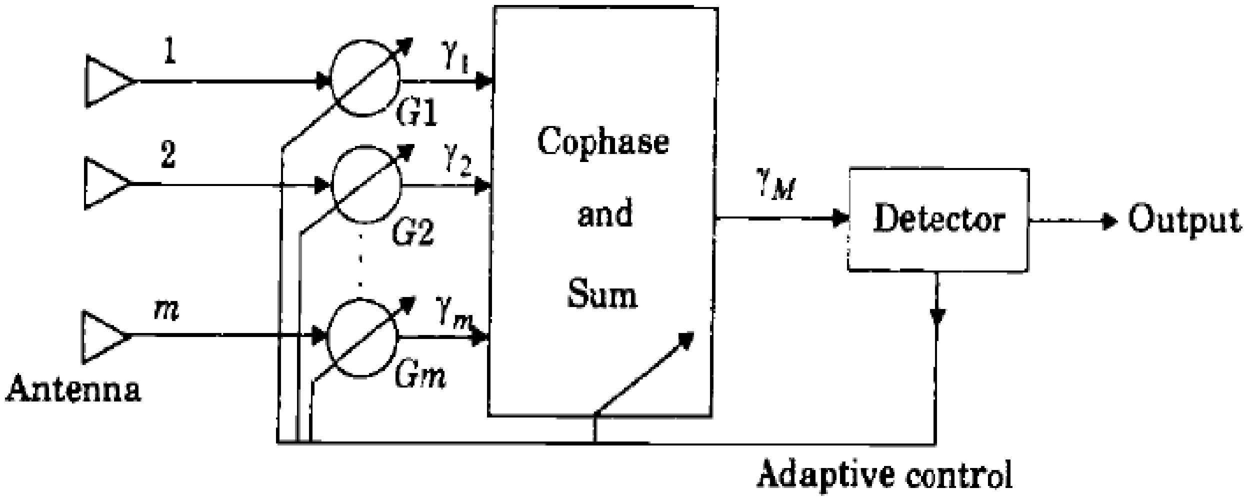
ii. Feedback or Scanning Diversity:

- In this diversity, 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.



iii. 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.
- The individual signals must be co-phased before being summed which requires an individual receiver and phasing circuit for each antenna element.



iv. Equal Gain Combining:

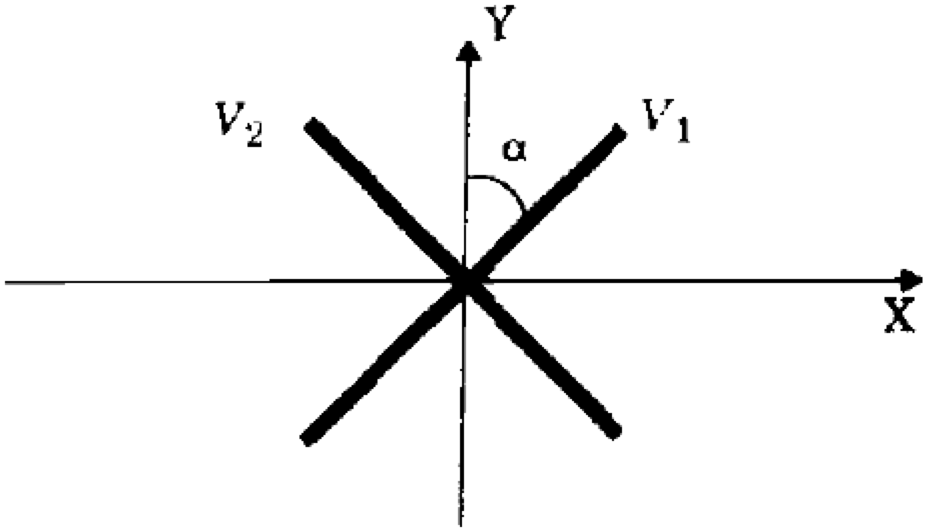
- In this method, the branch weights are all set to unity but the signals from each branch are co-phased to provide equal gain combining diversity.
- This allows the receiver to exploit signals that are simultaneously received on each branch.

3. Polarization Diversity

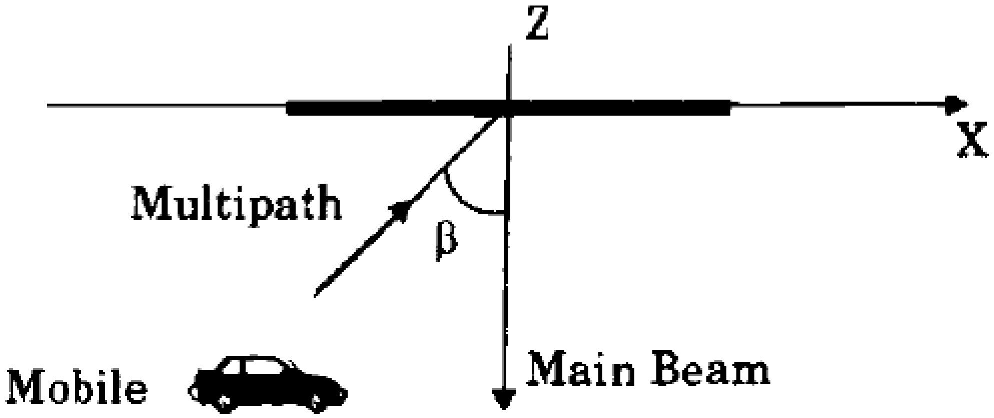
- At the base station, spatial diversity is less practical than at the mobile.
- Polarization diversity only provides two diversity branches it does allow the antenna elements to be co-located.
- The decorrelation for the signals in each polarization is caused by multiple reflections in the channel between the mobile and base station antennas.
- Multiple versions of a signal are transmitted and received via antennas with different polarization.
- A diversity combining technique is applied on the receiver side.

Theoretical Model for Polarization Diversity:

- It is assumed that the signal is transmitted from a mobile with vertical (or horizontal) polarization.
- It is received at the base station by a polarization diversity antenna with 2 branches.
- As seen in the figure, a polarization diversity antenna is composed of two antenna elements V_1 and V_2, which make $a \pm \alpha$ angle (polarization angle) with the Y axis.
- A mobile station is in the direction of offset angle β from the main beam direction of the diversity antenna.



(a) x-y plane



(b) x-z plane

- Some of the vertically polarized signals transmitted are converted to the horizontal polarized signal because of multipath propagation.
- The signal arriving at the base station can be expressed as

$$x = r_1 \cos(\omega t + \phi_1)$$
$$y = r_2 \cos(\omega t + \phi_2)$$

where *x* and *y* are signal levels which are received when β=0.

- The received signal values at elements V_1 and V_2 can be written as:

$$V_1 = (r_1 a\ cos\emptyset_1 + r_2 b\ cos\emptyset_2)coswt - (r_1 a\ sin\emptyset_1 + r_2 b\ sin\emptyset_2)sinwt$$
$$V_2 = (-r_1 a\ cos\emptyset_1 + r_2 b\ cos\emptyset_2)coswt - (-r_1 a\ sin\emptyset_1 + r_2 b\ sin\emptyset_2)sinwt$$

where $a = sin\alpha\ cos\beta$
$b = cos\alpha$

- The correlation coefficient *p* can be written as

$$\rho = \left(\frac{\tan^2(\alpha)\cos^2(\beta) - \Gamma}{\tan^2(\alpha)\cos^2(\beta) + \Gamma} \right)^2$$

where

$$\Gamma = \frac{\langle R_2^2 \rangle}{\langle R_1^2 \rangle}$$

Here **Γ** is the cross-polarization discrimination of the propagation path between a mobile and a base station.

where

$$R_1 = \sqrt{r_1^2 a_2 + r_2^2 b_2 + 2 r_1 r_2 ab \cos(\phi_1 + \phi_2)}$$

$$R_1 = \sqrt{r_1^2 a_2 + r_2^2 b_2 - 2 r_1 r_2 ab \cos(\phi_1 + \phi_2)}$$

- The correlation coefficient ***p*** is determined by three factors:
  - Polarization angle $\boldsymbol{\alpha}$;
  - Offset angle $\boldsymbol{\beta}$ from the main beam direction of the diversity antenna
  - The cross-polarization discrimination $\boldsymbol{\Gamma}$.
- The correlation coefficient ***p*** generally becomes higher as offset angle $\boldsymbol{\beta}$ becomes larger.
- Also, ***p*** generally becomes lower as polarization angle $\boldsymbol{\alpha}$ increases. This is because the horizontal polarization component becomes larger as $\boldsymbol{\alpha}$ increases.
- Because antenna elements V_1 and V_2 are polarized at $\pm\boldsymbol{\alpha}$ to the vertical, the received signal level is lower than that received by a vertically polarized antenna.
- The average value of signal loss L , relative to that received using vertical polarization is given by

$$L = a^2/\Gamma + b^2$$

4. Frequency diversity

- Frequency diversity transmits information on more than one carrier frequency.

- Frequencies separated by more than the coherence bandwidth of the channel will not experience the same fads.

- Frequency diversity is employed in microwave line-of-sight links which carry several channels in a frequency division multiplex mode (FDM).

- Due to tropospheric propagation and resulting refraction, deep fading sometimes occurs.

5. 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.

6. RAKE Receiver

- In CDMA spread spectrum systems, the chip rate is typically much greater than the flat fading bandwidth of the channel.

- CDMA spreading codes are designed to provide very low correlation (in general correlation is not good) between successive chips purposefully.

- 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.

- 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.

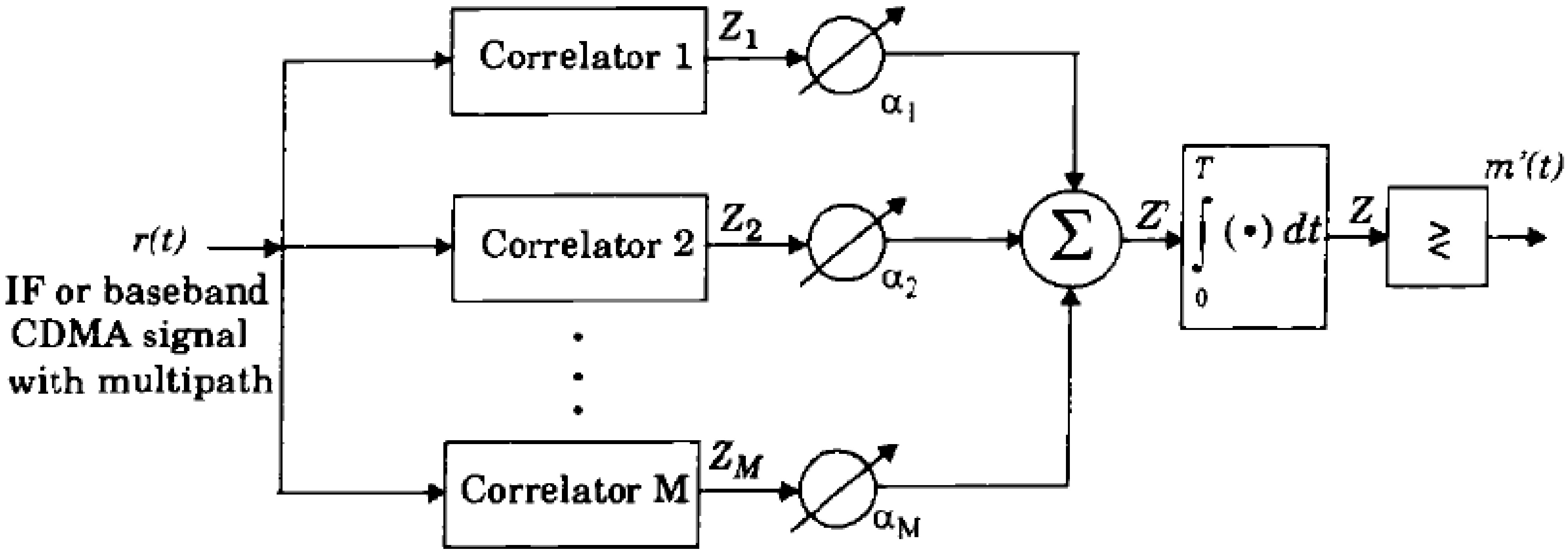


Fig. An M-branch (M-finger) RAKE receiver implementation. Each correlator detects a time shifted version of the original CDMA transmission, and each finger of the RAKE correlates to a portion of the signal which is delayed by at least one chip in time from the other fingers.

- The RAKE receiver is essentially a diversity receiver designed specifically for CDMA, where the diversity is provided by the multipath components are practically uncorrelated from one another when their relative propagation delays exceed a chip period.

- 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.

- Assume *M* correlators are used in a CDMA receiver to capture the *M* strongest multipath components.

- A weighting network is used to provide a linear combination of the correlator output for bit detection.

- Correlator 1 is synchronized to the strongest multipath m_1. Multipath component m_2 arrives τ_1 later than component m_1.

- The second correlator is synchronized to m_2. It correlates strongly with m_2 but has low correlation with m_1 .

- In a RAKE receiver, if the output from one correlator is corrupted by fading, the others may not be corrupted by fading, and the corrupted signal may be discounted through the weighting process.

- Decisions based on the combination of the M separate decision statistics offered by the RAKE provide a form of diversity which can overcome fading and thereby improve CDMA reception.

- The outputs of the *M* correlators are denoted as Z_1, Z_2 ,... and Z_M. They are weighted by α_1, α_2, ... α_M, respectively.

- The weighting coefficients are based on the power or the SNR from each correlator output. If the power or SNR is small out of a particular correlator, it will be assigned a small weighting factor.

- The overall signal Z' is given by

$$Z' = \sum_{m=1}^{M} \alpha_m Z_m \qquad \text{where} \quad \alpha_m = \frac{{Z_m}^2}{\sum_{m=1}^{M} {Z_m}^2}$$