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Unit - IV Signal Processing in Wireless Systems

Principles of diversity, Macrodiversity, Microdiversity, Signal Combining Techniques, Transmit diversity, Equalisers - Linear and Decision feedback equalisers, Review of Channel Coding and Speech Coding techniques.

4.1 Principles of diversity :-

* Diversity is a technique to improve the BER. It is a powerful communication receiver technique that provides wireless link improvement at relatively low cost.

* It requires no training overhead since a training sequence is not required by the transmitter.

Definition :-

* The principle of diversity is to ensure that the same information reaches the receiver (Rx) on statistically independent channels.

* Diversity exploits the random nature of radio propagation by finding independent (or at least highly uncorrelated) signal paths for communication.

* 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 SNR at the receiver may be improved, often by as much as 20 dB to 30 dB.

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Correlation Coefficient :- [ρ_{xy}]

The Correlation Coefficient characterizes the Correlation between Signals on different diversity branches. For Signal envelopes of x and y, the Correlation Coefficient is defined as:

$$\rho_{xy} = \frac{E[x \cdot y] - E[x] \cdot E[y]}{\sqrt{\{E[x^2] - E[x]^2\} \cdot \{E[y^2] - E[y]^2\}}}$$

For two statistically independent signals, $E[xy] = E[x]E[y]$. Therefore, the Correlation Coefficient becomes Zero.

Signals are said to be "effectively" decorrelated if ρ is below a certain threshold (typically 0.5 or 0.7)

Joint probability density function (pdf) of field strength $Pdf_{r_1, r_2, \ldots}(r_1, r_2, \ldots)$ is equal to the product of the marginal pdfs for the channel, $Pdf_{r_1}(r_1)$, $Pdf_{r_2}(r_2)$, ... Any Correlation between the fading of the channels decreases the effectiveness of diversity.

Types of diversity :-

Receiver diversity :- [Multiple Antennas at the receiver]

There are two types of diversity :-

* Micro diversity [Combat Small scale fading]
* Macro diversity [Combat Large scale fading]

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4.2 Microdiversity :- [Antenna diversity or Space diversity]

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.

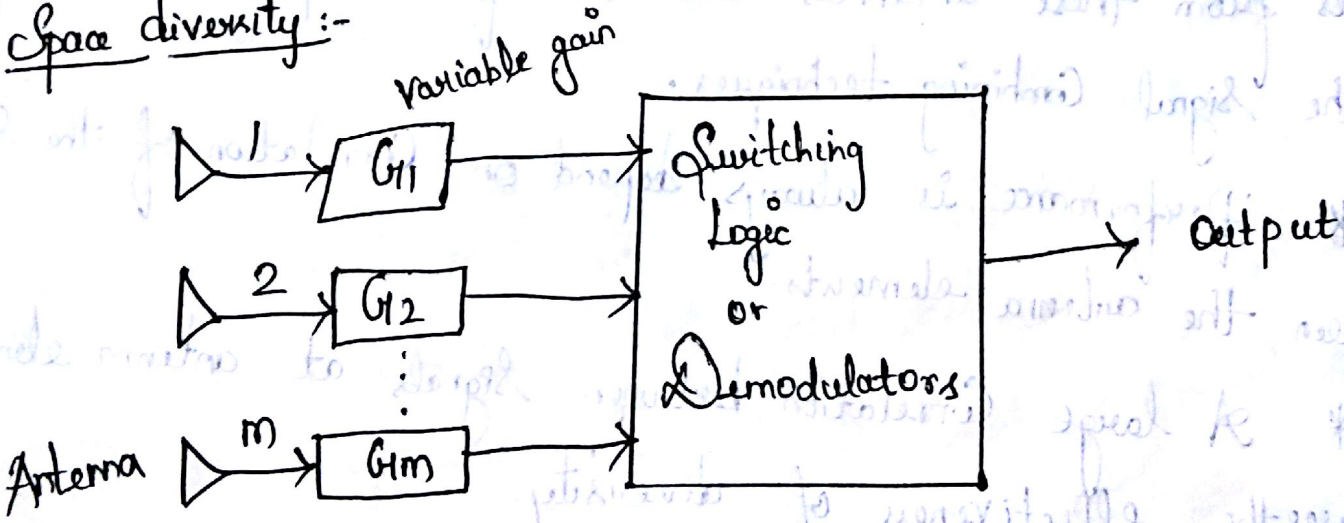
For narrowband signals, small-scale fading typically results in a Rayleigh fading distribution of signal strength over small distances.

In order to prevent deep fades from occurring, microdiversity techniques can exploit the rapidly changing signal.

If two antennas are separated by a fraction of a meter, one may receive a null while the other receives a null, while the other receives a strong signal.

By selecting the best signal at all times, a receiver can mitigate small-scale fading effects.

Space diversity :-



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This diversity is used in base station design. There are five different methods of macrodiversity. They are :-

* Spatial diversity : Several antenna elements separated in space.
* Temporal diversity : Repetition of the transmit signal at different times.
* Frequency diversity : Transmission of the signal on different frequencies.
* Angular diversity : Multiple antennas with different antenna patterns
* Polarization diversity : Multiple antennas receiving different polarizations (i.e vertical or Horizontal)

<u>Spatial diversity :-</u>

* Spatial diversity is the oldest and simplest form of diversity. Hence it is widely used.

* The transmit signal is received at several antenna and signals from these antennas are then further processed according to the signal combining techniques.

* Performance is always depend on correlation of the signals between the antenna elements.

* A large correlation between signals at antenna elements decreases the effectiveness of diversity.

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A first important step in designing diversity antennas is thus to establish a relationship between antenna spacing and the Correlation Coefficient. This relationship is different for BS and MS antennas.

(i) Mobile Station in Cellular and Cordless Systems :-

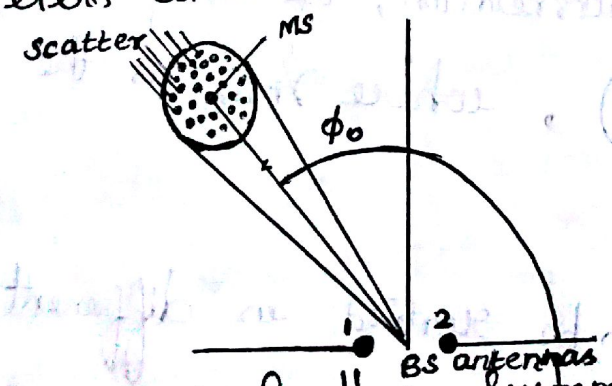
* Here waves are incident from all directions at the MS. Thus points of positive and negative interference of Multipath Component are spaced approximately $\lambda/4$ apart.

* This is therefore the distance that is required for decorrelation of received signals.

* For Wireless LAN, the distances are even smaller.

* For various Cordless and Cellular Systems, distance is about 4cm and In GSM, it is about 8 cm (at 900 MHZ).

Scatterers Concentrated around the mobile station



(ii) Base Station in Cordless Systems and WLANS :-

The angular distribution of incident radiation at indoor BSs is also uniform - i.e., radiation is incident with equal

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strength from all directions.

(iii) Base stations in Cellular Systems :-

For a cellular BS, the assumption of uniform directions of incidence is no longer valid.

Interacting object (IOs) are typically concentrated around the MS. Since all the waves are incident essentially from one direction, the correlation coefficient is higher.

∴ The antenna spacing required to obtain sufficient decorrel-ation increases.

Temporal diversity :-

* Temporal diversity means that the signals are received at different times. Since the channel is time-variant, signals received at different times are uncorrelated.

* For sufficient decorrelation, the temporal distance must be atleast $1/(2\nu_{max})$, where ν_{max} is the maximum Doppler frequency.

Temporal diversity can be realized in different ways:-

(i) Repetition Coding :-

This is the simplest form and it is bandwidth inefficient. The signal is repeated several times, where the

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repetition intervals are long enough to achieve decorrelation.

* Spectral efficiency decreases by a factor that is equal to the number of repetitions.

(ii) Automatic Repeat Request (ARQ) :-

* Here the Rx sends a message to the transmitter to indicate whether it received the data with sufficient quality.

* If this is not the case, then the transmission is repeated (after a wait period that achieves decorrelation).

* It requires a feedback channel.

* The spectral efficiency of ARQ is better than that of repetition coding, since it requires multiple transmissions only when the first transmission occurs in a bad fading state, while for repetition coding, retransmissions occur always.

(iii) Combination of interleaving and coding :-

* It is a more advanced version. Here the different symbols of a codeword are interleaved and transmitted at different times.

* The transmitted codeword can then be reconstructed.

* This method improves BER.

<u>Frequency diversity :-</u>

In frequency diversity, the same signal is transmitted at different frequencies.

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* If these frequencies are spaced apart by more than the Coherence bandwidth of the Channel, then their fading is approximately independent, and the probability is low that the signal is in a deep fade at both frequencies simultaneously.

* The information is spread over a large bandwidth. This spreading can be done by following methods:-

* Compressing the information in time - (i.e) Sending short bursts that occupy a large bandwidth.

* Code Division Multiple Access (CDMA)

* Multi Carrier CDMA and Coded Orthogonal frequency division Multiplexing

* Frequency hopping in Conjuction with Coding:
Different parts of a codeword are transmitted on different carrier frequencies.

<u>Angle Diversity</u>:- [pattern diversity]

* Two Co-located antennas with different patterns "see" differently weighted Multipath Component (MPCs), so that the MPCs interfere differently for the two antennas.

* A fading dip is created when MPCs, which usually come from different directions, interfere destructively.

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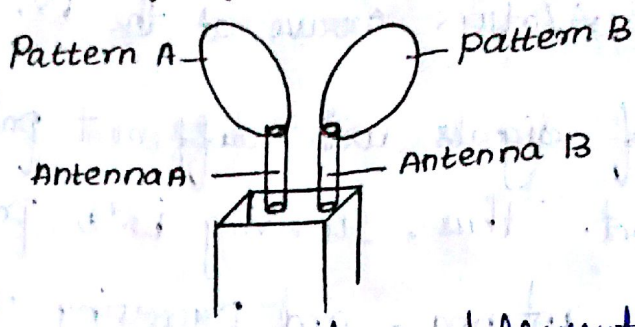
* If some of these waves are attenuated or eliminated, then the location of fading dip changes.

Angle diversity for closely spaced antennas



* Different types of antennas have different patterns. But even identical antennas can have different patterns when mounted close to each other.

* This is due to mutual Coupling:- Antenna B acts as a reflector for antenna A, whose pattern is therefore skewed to the left.

* Analogously, the pattern of antenna B is skewed to the right due to reflections from antenna A. Thus the two patterns are different.

* Angular diversity is used in Conjuction with Spatial diversity. It enhances the decorrelation of Signals at closely spaced antennas.

<u>Polarization diversity :-</u>

* As the reflection and diffraction process depend on Polarization, Horizontally and vertically polarized MPCs propagate

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differently in a wireless channel. Even if the transmit antenna only sends signals with a single polarization, the propagation effects in the channel lead to depolarization so that both polarizations arrive at the Rx.

* The fading of signals with different polarizations is statistically independent. Thus, receiving both polarizations using a dual-polarized antenna, and processing the signals separately, offers diversity.

* This diversity technique can be obtained without any requirement for a minimum distance between antenna elements.

* Consider a more closely situation where the transmit signal is vertically polarized, while the signal is received in both vertical and horizontal polarization.

* In this case, fading of the two received signals is independent, but the average received signal in the two diversity branches is not identical.

* Depending on the environment, the horizontal (i.e. cross-polarized) component is some 3-20 dB weaker than the vertical (co-polarized) component.

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* The diversity order that can be achieved with polarization diversity is upto 6.

4.3 Macrodiversity and Simulcast :-

* This type is used for combating large-scale fading, which is created by shadowing effects.

* Shadowing affects different MPCs almost equally, so that frequency diversity or polarization diversity are not effective. Furthermore, the correlation distances for large-scale fading are on the order of tens or hundreds of meters, so that spatial or temporal diversity cannot be used either.

* If there is a hill between the TX and RX, adding antennas on either the BS or the MS does not help to eliminate the shadowing caused by this hill.

* The only way to eliminate this problem is to use a separate basestation (BS2) that is placed in such a way that the hill is not in the connection line between the MS and BS2. This is called as Macrodiversity.

* The simplest way for macrodiversity is the use of on-frequency repeaters that receive the signal and retransmit an amplified version of it.

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Simulcast :-

* In Simulcast, the same signal is transmitted simultaneously from different Basestations. In cellular communications, the two basestations should be synchronized and transmit the signals intended for a specific user in such a way that the two waves arrive at the Rx almost simultaneously.

* Synchronization can only be obtained if the runtimes from the two BSs to the MS are known.

* Simulcast is widely used for broadcast applications, especially digital TV. In this case, the exact synchronization of all possible Rxs is not possible. Each Rx would require a different timing advance from the Tx.

* It requires a large amount of signaling information. Synchronization information and transmit data have to be transported on landlines to the BSs.

* Synchronization is not required in on-frequency repeaters. Delay dispersion is larger because

(i) Runtime from BS to repeater and repeater to MS is larger,

(ii) Repeater introduces additional delays due to the group delays of electronic components, filters, etc...

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4.4 Transmitter Diversity :-

* In transmit diversity, there are multiple transmit antennas and the transmit power is divided among these antennas.

* It is desirable in systems where more space, power and processing capability is available on the transmit side than on the receive side.

* There are two forms of transmitter diversity :-

(i) Channel gain known at the transmitter.
[Similar to receive diversity]

(ii) Channel gain unknown at the transmitter.
[Alamouti scheme]

With known channel state information [CSI] :-

* Here the Tx knows the channel perfectly. This knowledge might be obtained from feedback from the Rx, or from reciprocity principles.

* The optimum transmission scheme linearly weights signals transmitted from different antenna elements with the complex conjugates of the channel transfer functions

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from the transmit antenna elements to the single receive antenna. This approach is known as maximum ratio transmission.

* At high SNR, the diversity order of transmit diversity with known channel gain (achieves full diversity) is M, where M is the number of branches.

* Thus its performance is similar to receiver diversity.

* Generally channel values can be measured at the receiver using a pilot technique and then feedback to the transmitter.

* In cellular systems with time division, the base station can measure the channel gain and phase on transmissions from mobile to the base station and then use these measurements when transmitting back to the mobile, since under time division the forward and reverse links are reciprocal.

With Unknown Channel State information :-

* In many cases, channel state information (CSI) is not available at the Tx. The transmit weighted copies of the same signal cannot be transmitted from different transmit antennas. Because it is not known that how they would add up at the Rx.

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There are three ways of achieving transmit diversity in this case.

* Delay diversity,
* phase- Sweeping diversity,
* Space time Coding.

Delay diversity :-

* In this method, transmitted signal Components from different antennas are delayed Copies of the same signal. Thus each Component can be distinguished easily by the other Component.

Space time Coding :-

* A more common method that Combines space and time diversity is Alamouti scheme.

* This scheme is developed for a digital Communication System with two antenna transmit diversity.

* This scheme works over two symbol periods and it is assumed that the channel gain is Constant over this time.

* Over the first symbol periods, two different symbols S_1 and S_2 are transmitted simultaneously from antennas 1 and 2 respectively. Over the next symbol period, symbol $-S_2^*$ is transmitted from antenna 1 and symbol S_1^* is transmitted from antenna 2.

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Thus the received symbol period is

$$y_1 = h_1 s_1 + h_2 s_2 + n_1 \quad \text{and}$$

$$y_2 = -h_1 s_2^* + h_2 s_1^* + n_2$$

where n_i $(i=1,2)$ is the AWGN sample at the receiver.

h_i $(i=1,2)$ is the channel gain.

This schemes achieves diversity order of 2.

Phase-sweeping diversity :-

* This scheme is especially useful if there are 2 antennas.
* The same signal is transmitted from both antenna elements.

One of the antenna signals undergoes a time-varying phaseshift.

* Thus temporal variations (time) are introduced.
* Even if the Tx, Rx and the interacting objects (IOs) are stationary, the signal does not remain stuck in a fading dip

<u>Space time Coding :-</u> [For 2 Antenna System]

$$\begin{array}{c|cc} & \text{Antennas} \rightarrow & \\ \text{Time Period} \downarrow & 1 & 2 \\ \hline 1 & s_1 & s_2 \\ 2 & s_2^* & s_1^* \end{array}$$

* represents Conjugate. This is used here to introduce Orthogonality.

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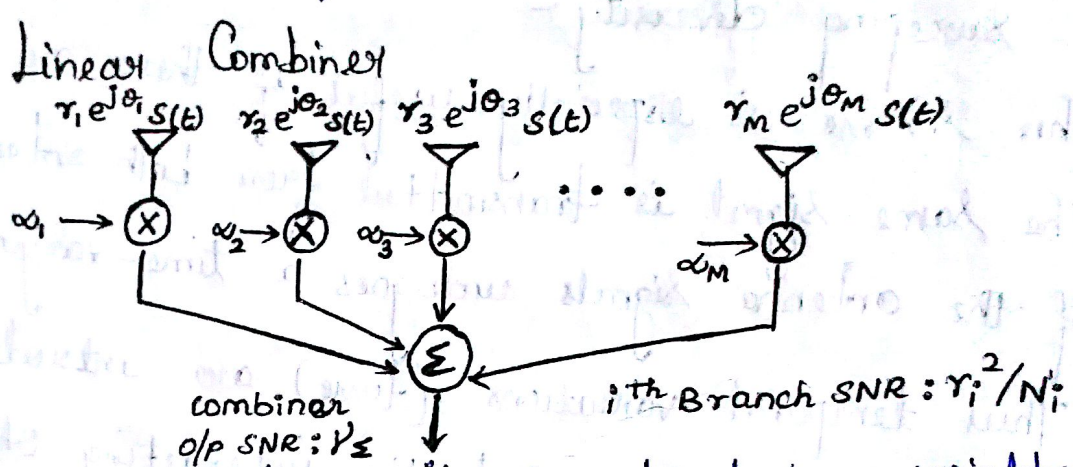
4.5 Signal Combining Techniques :-

* In signal receiver diversity, the independent fading paths associated with multiple receive antennas are combined to obtain a signal that is then passed through a standard demodulator.

* The combining can be done in several ways, which vary in complexity and overall performance.



* The Combiner is <u>linear if the output is a weighted sum of the different fading paths or branches.</u>

Cophasing :-

* Combining more than one branch requires co-phasing, where the phase θ_i of the ith branch is removed through multiplication by $\alpha_i = a_i e^{-j\theta_i}$ for some real valued a_i. This phase removal requires coherent detection of each branch.

* Without co-phasing, the branch signals would not add up coherently in the combiner.

* Thus fading is not removed.

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Array gain [Ag] :-

The SNR increase in the absence of fading is referred to as the array gain. It is defined as the increase in the average Combined SNR $\bar{\gamma}_\Sigma$ over the average branch SNR $\bar{\gamma}$.

(i.e) $$A_g = \frac{\bar{\gamma}_\Sigma}{\bar{\gamma}}$$

Diversity gain :-

For a $\bar{\gamma}_\Sigma$, if symbol error rate and Outage probability decreases because of diversity Combining, the resulting gain is called diversity gain.

There are four Combining methods :-

* Selection diversity,
* Switched diversity,
* Maximal Ratio Combining (MRC),
* Equal gain Combining (EGC).

} → Combining technique

Selection diversity :-

There are two methods of selection diversity :-

* Received signal strength indication driven diversity [RSSI]
* BER-driven selection diversity.

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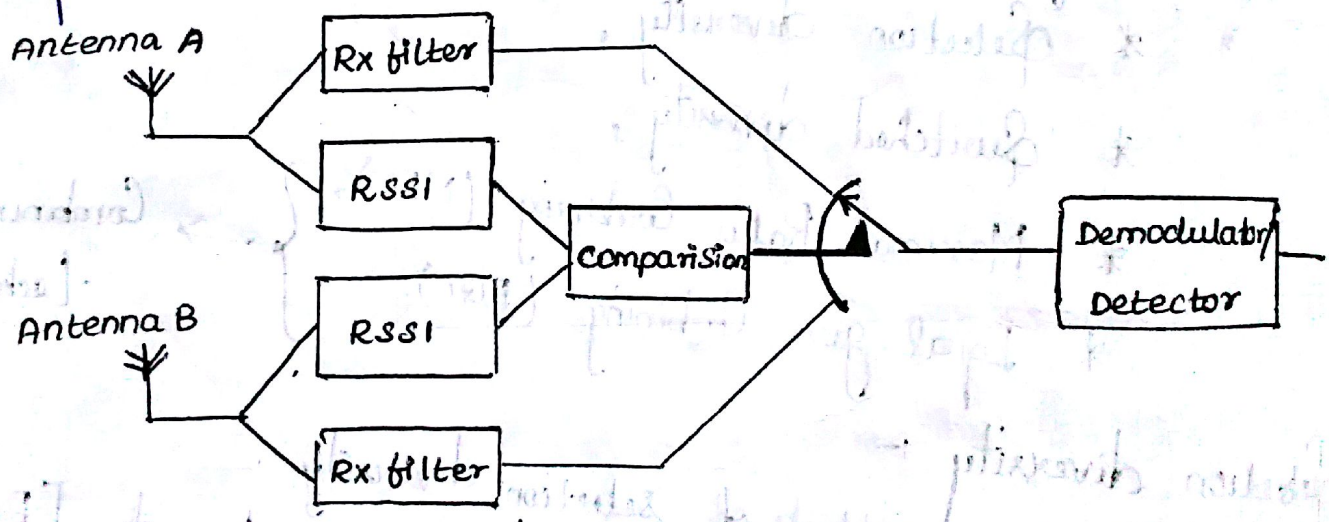
RSSI driven Selection diversity :-

* In this method, the signal with the largest instantaneous power and processes it further.
* This method requires N_r antennas, N_r RSSI sensors, and a N_r-to-1 multiplexer but only one RF chain.
* It is a simple tracking procedure even in fast fading channels.
* In case of AWGN channels, RSSI driven selection diversity is the best of all the diversity methods.
* In case of frequency selective channels, this diversity is suboptimum.



BER-driven Selection diversity :-

* In this scheme, a training sequence [a bit sequence that is known at the Rx] is transmitted.
* The Rx then demodulates the signal from each receive antenna and compare it with the transmit signal.

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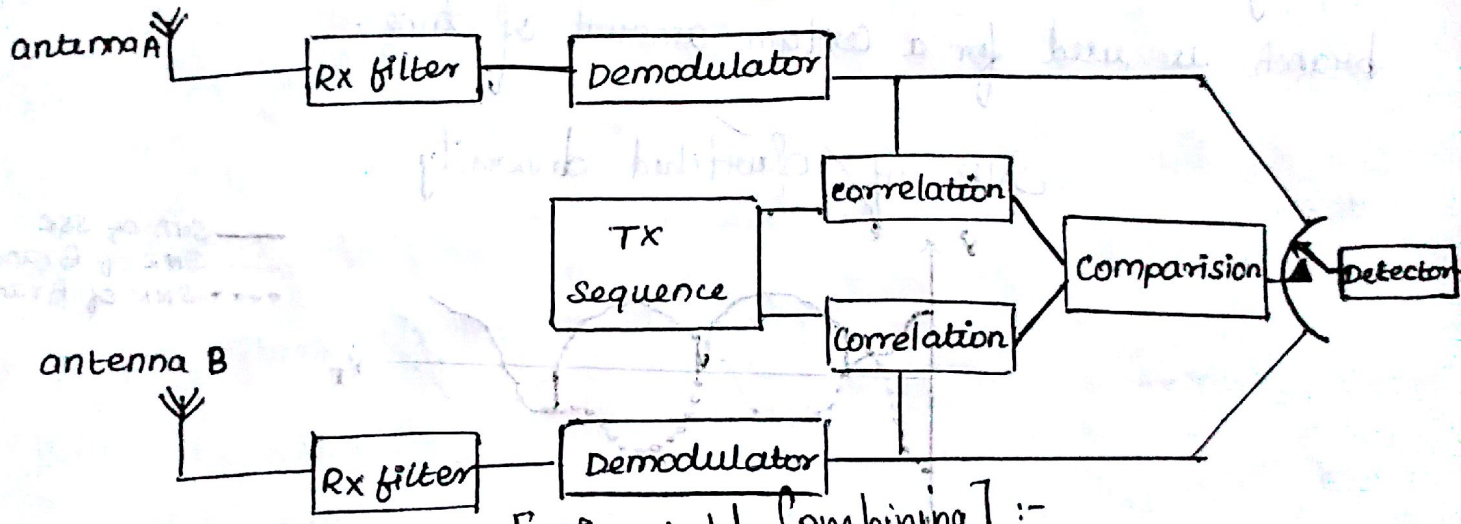
The antenna whose signal results in the smallest BER is considered as best and it is used for subsequent reception of data signals.

Drawbacks :-

* If RX has only one demodulator, it is difficult to evaluate BER of all diversity branches. This is critical if the channel changes quickly
* Since the duration of the training sequence is finite, BER cannot be determined exactly.



Switched diversity [Threshold Combining] :-

* In this method, the selection criterion [BER or SNR] of the active diversity branch is monitored.
* If it falls below a certain threshold, then the RX switches to a different antenna.
* It needs less amount of hardware and increases spectral efficiency.

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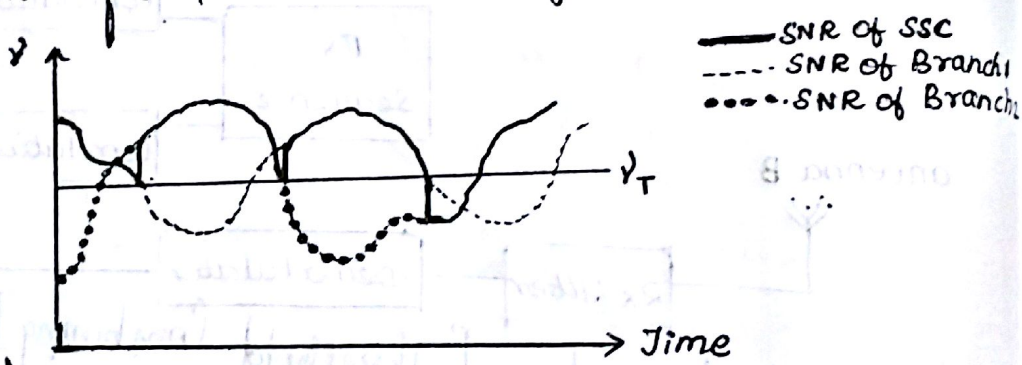
* Cophasing is not required because only one branch output is used at a time. Hence this technique can be used with coherent or differential modulation.

* This method is called as Switch-and-Stay Combining (SSC).

* If both branches have signal quality below the threshold, then the Rx just switches back and forth between the branches. This situation can be avoided by introducing a hysteresis or hold time, so that the new diversity branch is used for a certain amount of time.

SNR of Switched diversity



Combining Diversity :-

* Selection diversity wastes signal energy by discarding $(N_r - 1)$ copies of the received signal.

* This drawback is avoided by combining diversity, which exploits all available signal copies.

* There are two ways of doing combining :-

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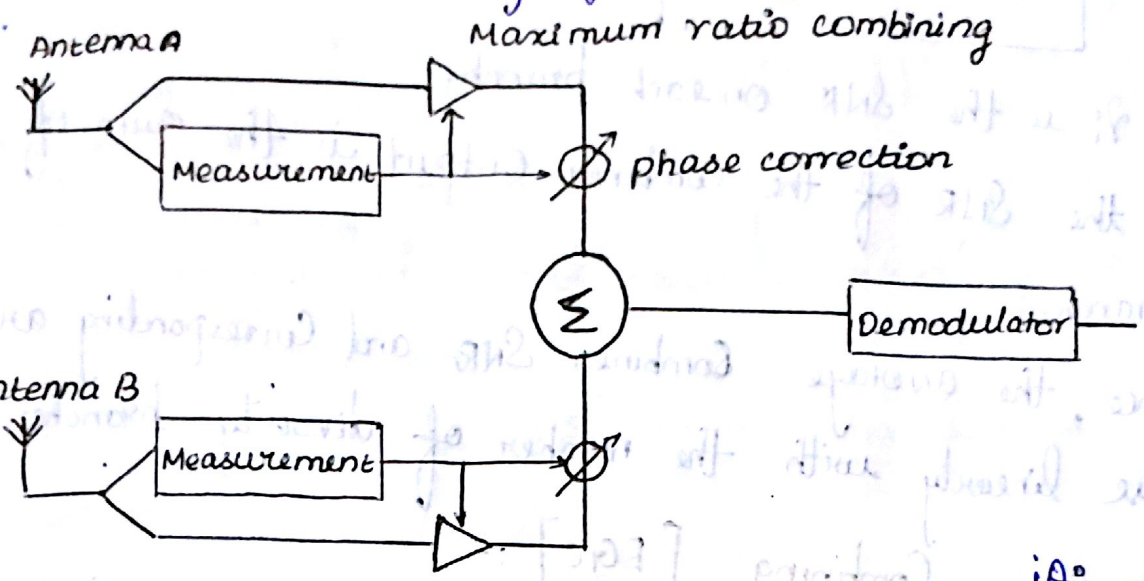
Maximal Ratio Combining :- (MRC)

* In MRC, each signal copy is multiplied by a (Complex) weight, and then added up (based on SNR).

* This is the optimum way of combining different diversity branches.



* The signals are cophased and so $a_i = a_i e^{-j\theta_i}$ where θ_i is the phase of the incoming signal on the ith branch.

Thus the envelope of the combiner output will be

$$r = \sum_{i=1}^{M} a_i r_i$$

where M is the number of branches

Assuming the same noise PSD $N_0/2$ in each branch yields a total noise PSD $N_{tot}/2$ at the combiner output.

$$N_{tot}/2 = \sum_{i=1}^{M} a_i^2 N_0/2$$

Thus the output SNR of the combiner is

$$\gamma_\Sigma = \frac{r^2}{N_{tot}} = \frac{\left(\sum_{i=1}^{M} a_i \gamma_i\right)^2}{\sum_{i=1}^{M} a_i^2 N_0}$$

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(i.e) $$\gamma_\Sigma = \frac{1}{N_0} \frac{\left(\sum_{i=1}^{M} a_i r_i\right)^2}{\sum_{i=1}^{M} a_i^2}$$

$$\boxed{\gamma_\Sigma = \sum_{i=1}^{M} \gamma_i}$$

where γ_i is the SNR on each branch.

Thus the SNR of the Combiner output is the sum of SNRs of each branch.

Hence, the average Combiner SNR and Corresponding array gain increase linearly with the number of diversity branches M.

Equal Gain Combining [EGC] :-

* MRC requires Knowledge of the time-varying SNR on each branch, which can be difficult to measure.

* EGC is a simple technique which Co-phases the Signals on each branch and then Combines them with equal weighting $\alpha_i = e^{-j\theta_i}$. The SNR of the Combiner output is

$$\gamma_\Sigma = \frac{1}{N_0 M} \left[\sum_{i=1}^{M} r_i\right]^2$$

Here, equal noise PSD on each branch is $N_0/2$.

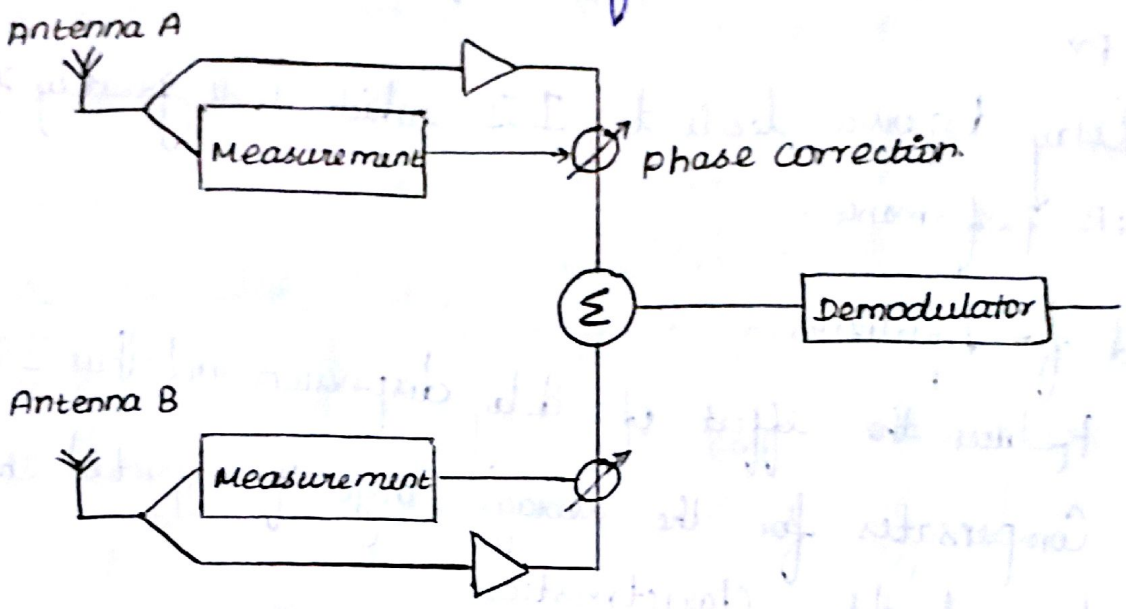
$M \rightarrow$ Number of branches.

EGC performs worse than MRC by only a factor $\pi/4$ (in terms of mean SNR).

The mean SNR of the Combiner Output is

$$\bar{\gamma}_{\Sigma} = \bar{\gamma}\left(1 + (M-1)\cdot \pi/4\right)$$

where $\bar{\gamma}$ is the mean SNR of all branches.



Equal Gain Combining

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Equalizers :-

* Equalizer compensates for intersymbol interference (ISI) created by multipath components.

* Wireless channels can exhibit delay dispersion (i.e). Multipath components (MPCs) can have different runtimes from Tx to Rx.

* Delay dispersion leads to ISI which will greatly reduce the BER performance.

Need for Equalizers :-

* Reduces the effect of delay dispersion and thus ISI

* Compensates for the average range of expected channel amplitude and delay characteristics.

Design of Equalizer :-

* 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 <u>adaptive equalizers</u>.

* The general operating modes of an adaptive equalizer include <u>training and tracking</u>.

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* First, a known, fixed-length training sequence is sent by the transmitter so that the receiver's equalizer may adapt to a proper setting for minimum BER detection.

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

* This training sequence is transmitted to estimate the channel impulse response.

* Immediately following this training sequence, the user data is sent and the adaptive equalizer at the receiver utilizes a recursive algorithm to evaluate the channel and estimate filter coefficients to compensate for the distortion created by multipath.

* The training sequence is designed to permit an equalizer at the receiver to acquire the proper filter coefficients in the worst possible channel conditions so that when the training sequence is finished, the filter coefficients are near the optimal values for reception of user data.

* As user data are received, the adaptive algorithm of the equalizer tracks the changing channel.

* As a consequence, the adaptive equalizer is continually changing its filter characteristics over time.

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* When an equalizer has been properly trained, it is said to have converged.

* Equalizers require periodic retraining in order to maintain effective ISI cancellations and are commonly used in digital communication systems where user data is segmented into short time blocks or time slots.

* The goal of an equalizer is to reverse distortions by the channel.

Consider the original signal be $s(t)$. It is sent through a wireless channel with the impulse response $h(t)$, received and sent through an equalizer with impulse response $e(t)$.

Assume that the transmit signal uses pulse amplitude modulation (PAM).

$$s(t) = \sum_i c_i g(t - iT)$$

where the c_i are the complex transmit symbols.

It is required that:

$$H(\omega)\, G(\omega)\, E(\omega) = \text{Const}$$

where $E(\omega)$, $H(\omega)$ and $G(\omega)$ are the Fourier transforms of $e(t)$, $h(t)$ and $g(t)$ respectively.

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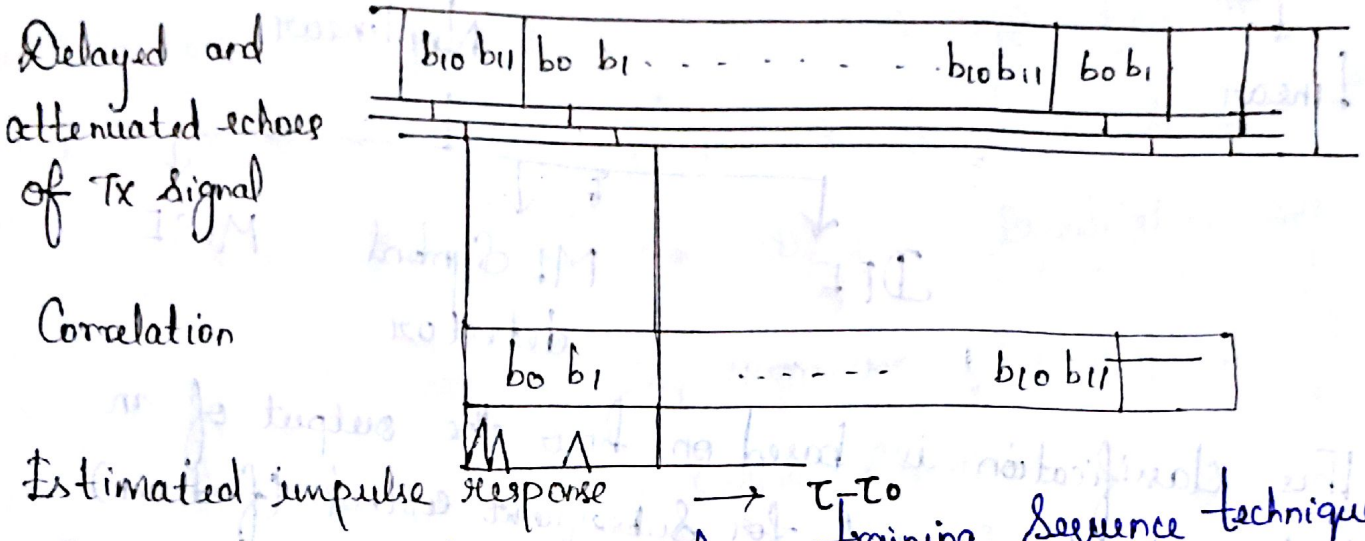
In time domain,

$$\eta(t) = h(t) * g(t)$$

where $\eta(t)$ is the Convolution of the channel impulse response $h(t)$ with the basis pulse $g(t)$.

Principle of Channel estimation by Correlation



Channel estimation by means of a training sequence technique has several drawbacks :-

* A reduction in Spectral efficiency :-
  Training sequence does not Convey any payload information.
  eg :- GSM uses 26 bits in every 148-bit frame for the training sequence.

* Sensitivity to noise :-
  To increase Spectral efficiency, training sequence must be short. This implies that the training sequence is sensitive to noise

* Outdated estimates :-
  If the channel changes after transmission of the training

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Sequence, the Rx cannot detect this variation. Use of an outdated channel estimate leads to decision errors.

Types of Equalizers :-

- Equalizer
  - Linear
  - Nonlinear
    - DFE
    - ML Symbol detector
    - MLSE

This classification is based on how the output of an adaptive equalizer is used for subsequent control (feedback) of the equalizer.

Generally the analog signal $\hat{d}(t)$ is processed by the decision making device in the receiver.

The decision maker determines the value of the digital data bit being received and applies a slicing or thresholding operation in order to determine the value of $d(t)$.

Linear equalizer :-

If $d(t)$ is not used in the feedback path to adapt the equalizer, the equalization is said to be linear.

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Non-Linear Equalizer :-

If d(t) is fed back to change the subsequent outputs of the equalizer, the equalization is said to be non-linear.

Algorithms for Adaptive Equalization :-

Since an adaptive equalizer compensates for an unknown and time-varying channel, it requires a specific algorithm to update the equalizer coefficients and track the channel variations.

Generally there are three different equalization algorithms:-

* Zero Forcing Algorithm,
* [LMSA] Least Mean Square Algorithm,
* [RMSA] Recursive Mean Square Algorithm.

} → For MMSE criterion

MMSE → Minimum mean Square Error.

Zero forcing Algorithm :-

* This was developed by Lucky for wireline communication.
* In this, 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 NT spaced sample points in the tapped delay line filter.

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* By letting the number of Coefficients increase without bound, an infinite length equalizer with Zero ISI at the o/p Can be obtained.

* When each of the delay elements provide a time delay equal to Symbol duration T, frequency response $H_{equ}(f)$ of the equalizer is periodic with a period equal to the Symbol rate $1/T$.

The Combined response of channel with equalizer must Satisfy Nyquist's first Criterion

$$H_{ch}(f)\, H_{equ}(f) = 1 \quad , \quad |f| \le \frac{1}{2T}$$

where $H_{cf}(f) \rightarrow$ folded frequency response of channel.

* Thus an, infinite length, Zero ISI equalizer is an inverse filter which inverts $H_{cf}(f)$.

* Disadvantages :-

* Inverse filter may excessively amplify noise at frequencies where the folded channel Spectrum has high attenuation

* Hence it is not often used for wireless links.

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The Mean Square error Criterion [MSE] :-

* The ultimate goal of an equalizer is minimization of BER.
* MSE is a better criterion which minimizes the MSE between the transmit signal and the output of equalizer.
* Filter is designed that minimizes :-

$$MSE = E\{|\epsilon_i|^2\} = E\{\epsilon_i \epsilon_i^*\}$$

Filter Coefficients C_{opt} are designed as

$$C_{opt} = R^{-1} P \longrightarrow (A)$$

where $R = E[u_i^* u_i^T]$ is the Correlation matrix of the received signal, $P = E[u_i^* c_i]$, the Cross Correlation between the received and transmit signal.

To find the optimum equalizer weights, eqn (A) has to be solved. This requires on the order of $(2K+1)^3$ Complex operations. For this purpose, the following two iterative algorithms have been developed. They are LMS and RMS.

The quality of an iterative algorithm is determined by the following criteria :-

* Convergence rate,
* Computational effort per iterations,
* Misadjustement.

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Least Mean Square Algorithm :- [LMS]

* Also known as Stochastic gradient method.
* A more robust algorithm.

Steps :-

(i) Initialize the weights with values e_0

(ii) Compute the approximation for the gradients of the MSE. Since the true gradient is an expected value, it cannot be computed. It is computed by using estimates R & p :-

$$\hat{R}_n = u_n^* u_n^T$$

$$\hat{P}_n = u_n^* c_n$$

where $n \rightarrow$ iterations. The gradient is estimated as :-

$$\hat{\nabla}_n = -2\hat{P}_n + 2\hat{R}_n e_n$$

(iii) Compute an updated estimate of e by adjusting weights in the direction of the negative gradient :-

$$e_{n+1} = e_n - \mu \hat{\nabla}_n$$

where μ is a user defined parameter.

The value of μ should be

$$0 < \mu < \frac{2}{\lambda_{max}}$$

λ_{max} - Largest Eigen value of R

(iv) If the stop criterion is fulfilled [Relative change in weight vector falls below a predefined threshold], the algorithm

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has Converged. otherwise return to step 2.

Advantages:- Low Computational Complexity, Simple Program.

Disadvantages:- poor tracking, Slow Convergence.

Here, when μ is Small, Convergence is slow.

Recursive least Square Algorithm [RLS]

* The problem of slow Convergence in LMS Can be Solved by RLS.

Steps:-

1) Initialize $w(0) = K(0) = x(0) = 0$, $R^{-1}(0) = \delta I_{NN}$, where I_{NXN} is an NXN identity matrix and δ is a large positive Constant.

2) Recursively Compute the following:-

$$\hat{d}(n) = W^T(n-1)\, y(n)$$

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

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

$$R^{-1}(n) = \frac{1}{\lambda}\left[R^{-1}(n-1) - K(n)\, y^T(n)\, R^{-1}(n-1)\right]$$

$$W(n) = W(n-1) + K(n)\, e^*(n)$$

where λ is the weighting Coefficient.

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Advantages :- Fast Convergence, Good tracking ability

Disadvantages :- High Computational Complexity

Linear Equalizer :-

Definition :-

If an estimated value is not used in the feedback path to adapt the equalizer, the equalization is said to be linear. A linear equalizer can be implemented by the following 2 ways :-

* Transversal filter [An FIR filter],
* Lattice filter.

* Linear Transversal filter [LTE] is the most Common and Simplest type available. In this type, the Current and past values of the received Signal are linearly weighted by the filter Coefficients and Summed up to produce output.

* LTE is made of tapped delay lines with tappings Spaced a Symbol period (T_s) apart.

The transfer function of LTE is

$$H(z) = \exp(-j\omega T_s) \text{ or } z^{-1}$$

* This filter has many Zeros but poles only at $z=0$. This is called as a finite impulse response filter (FIR).

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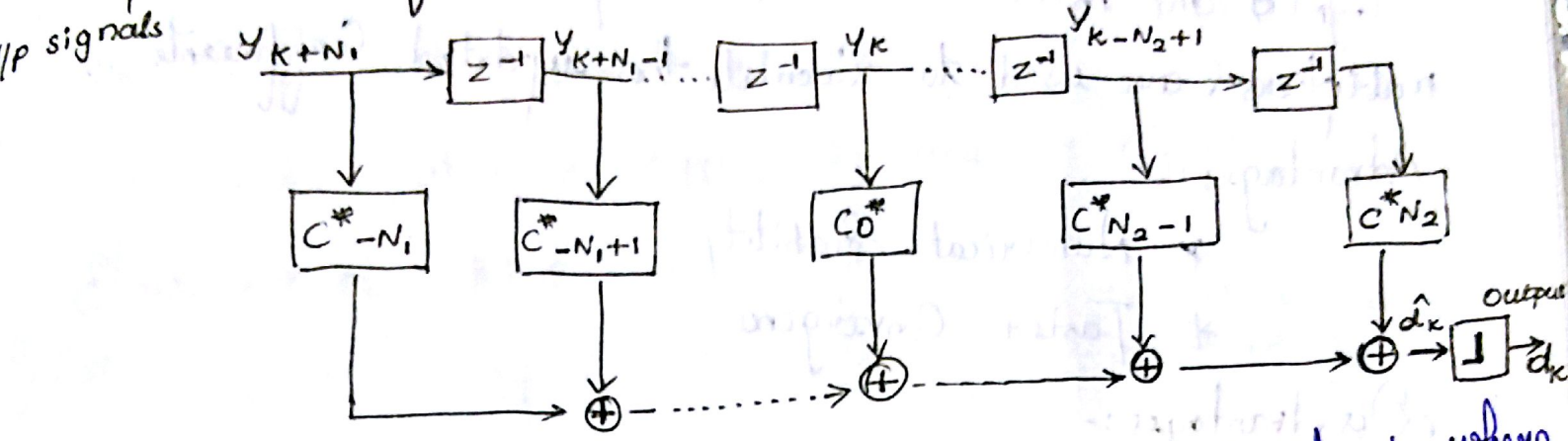
Structure of LTE :-



Usually this implementation is carried out in digital domain where samples of the received signals are stored in a shift register. The output of this filter before a decision is made :-

$$\hat{d}_K = \sum_{n=-N_1}^{N_2} (c_n^*)\, y_{K-n}$$

where C_n^* - represents complex filter coefficients

$y_i \rightarrow$ I/p received signal at time $t_0 + iT$

$t_0 \rightarrow$ Starting time

$N = N_1 + N_2 + 1 \longrightarrow$ Number of taps.

$N_1 \longrightarrow$ No. of taps in forward portion

$N_2 \rightarrow$ No. of taps in reverse portion.

In lattice filter structure, 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.

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$f_n(k)$ and $b_n(k)$ are used as inputs to the tap multipliers & are used to Calculate the updated Coefficients.

Advantages :-

* Numerical Stability
* Faster Convergence

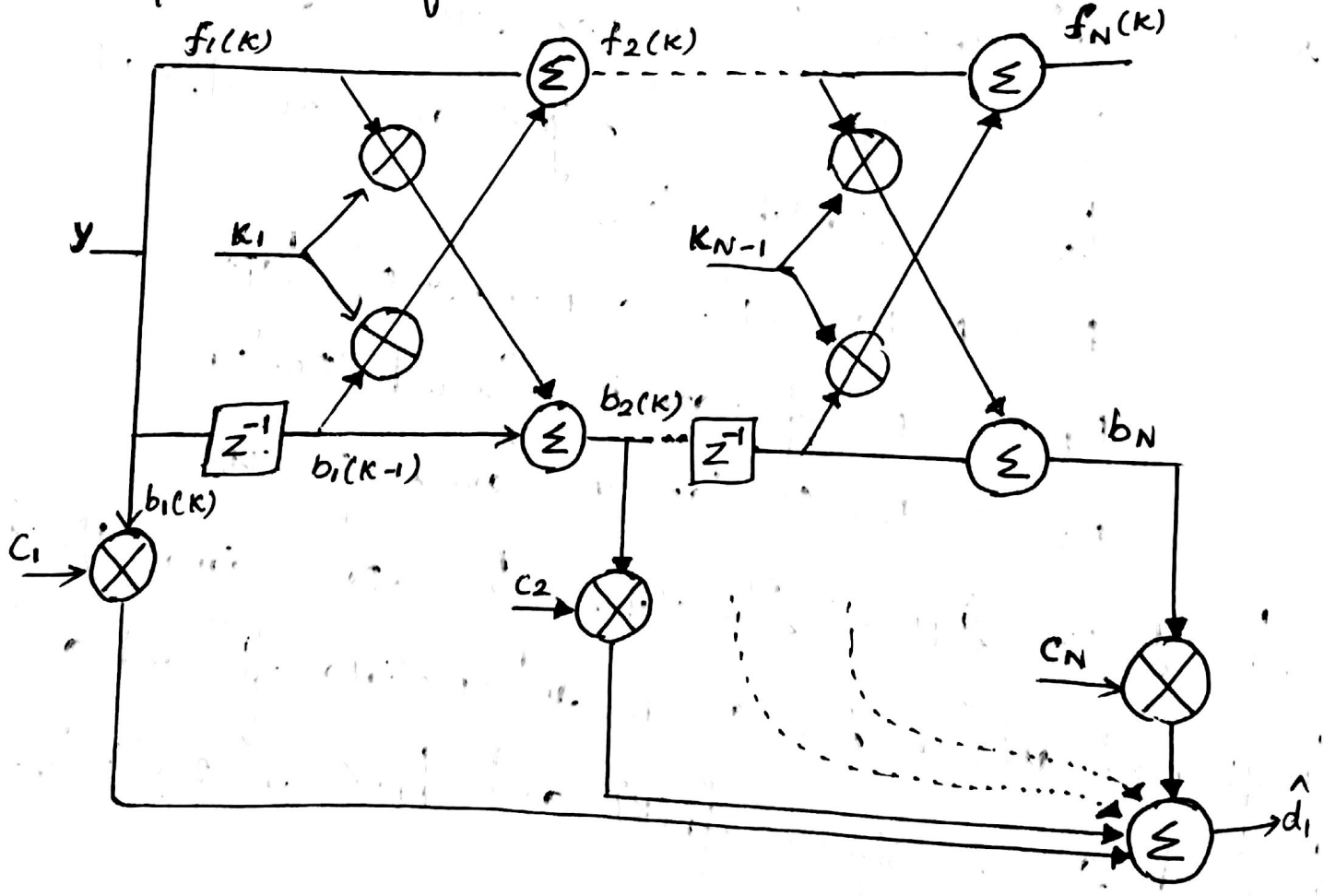
Disadvantages :-

* More Complicated Structure

The output of this equalizer is given by,

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

The Structure of a lattice equalizer



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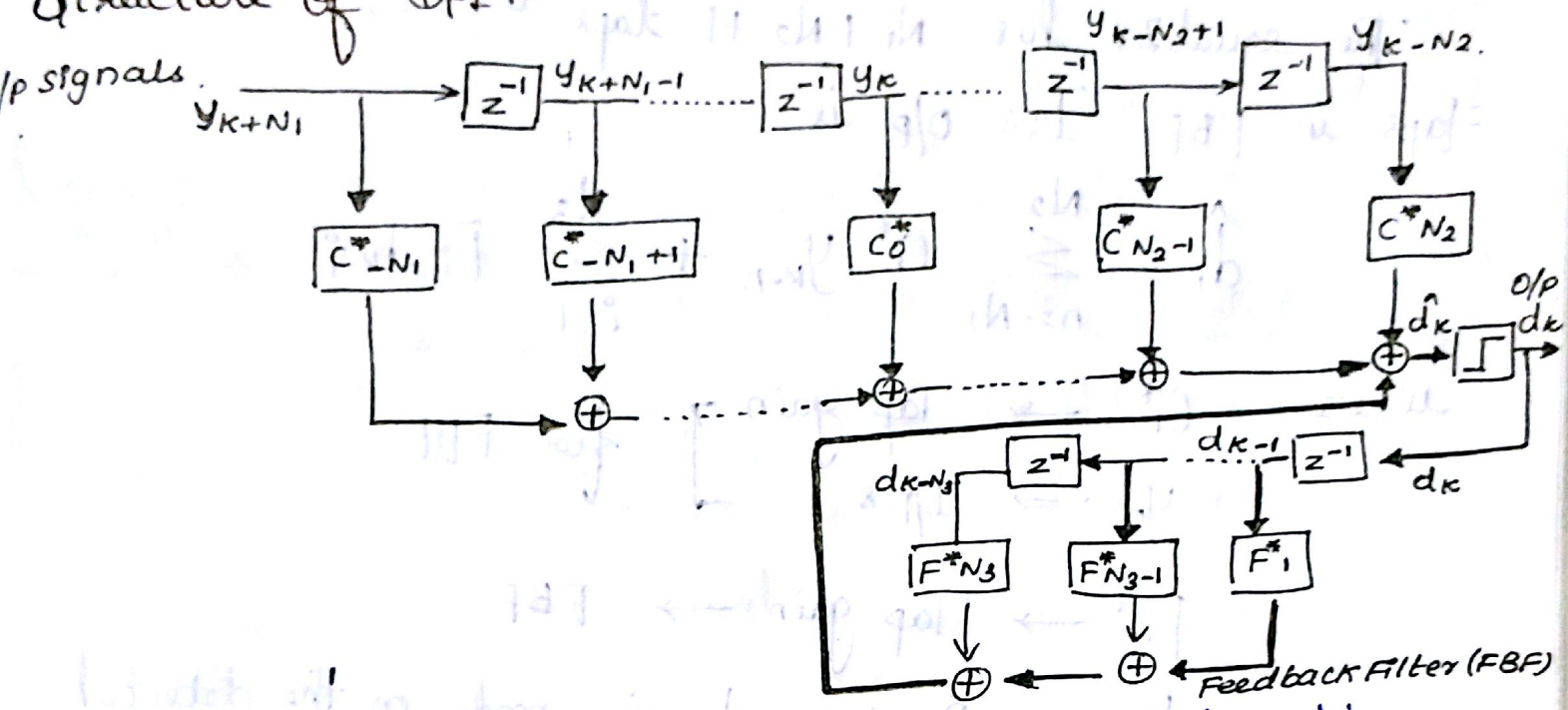
Decision Feedback equalization [DFE] :-

* It is a non-linear equalizer.
* Widely used in most 2G and 3G Systems.

Structure of DFE :-



The basic idea behind DFE 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.

This can be realized in either the direct transversal form or as a lattice filter.

It consists of a feed forward filter (FFF) and a feedback filter (FBF).

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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 FFF and N_3 taps in FBF. Its o/p is

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

where $C_n^* \rightarrow$ Tap gain] for FFF

$y_n \rightarrow$ I/ps

$F_i^* \rightarrow$ Tap gain $\rightarrow$ FBF

$d_i \rightarrow$ prevision decision made on the detected signal.

* The lattice implementation of DFE is equivalent to DFE having FFF of length N_1 & FBF of length N_2, where $N_1 > N_2$.

* Another form of DFE is predictive DFE.
* It also consists of a FFF. However FBF is driven by an input sequence formed by the difference of the o/p of the detector and the o/p of the FFF.

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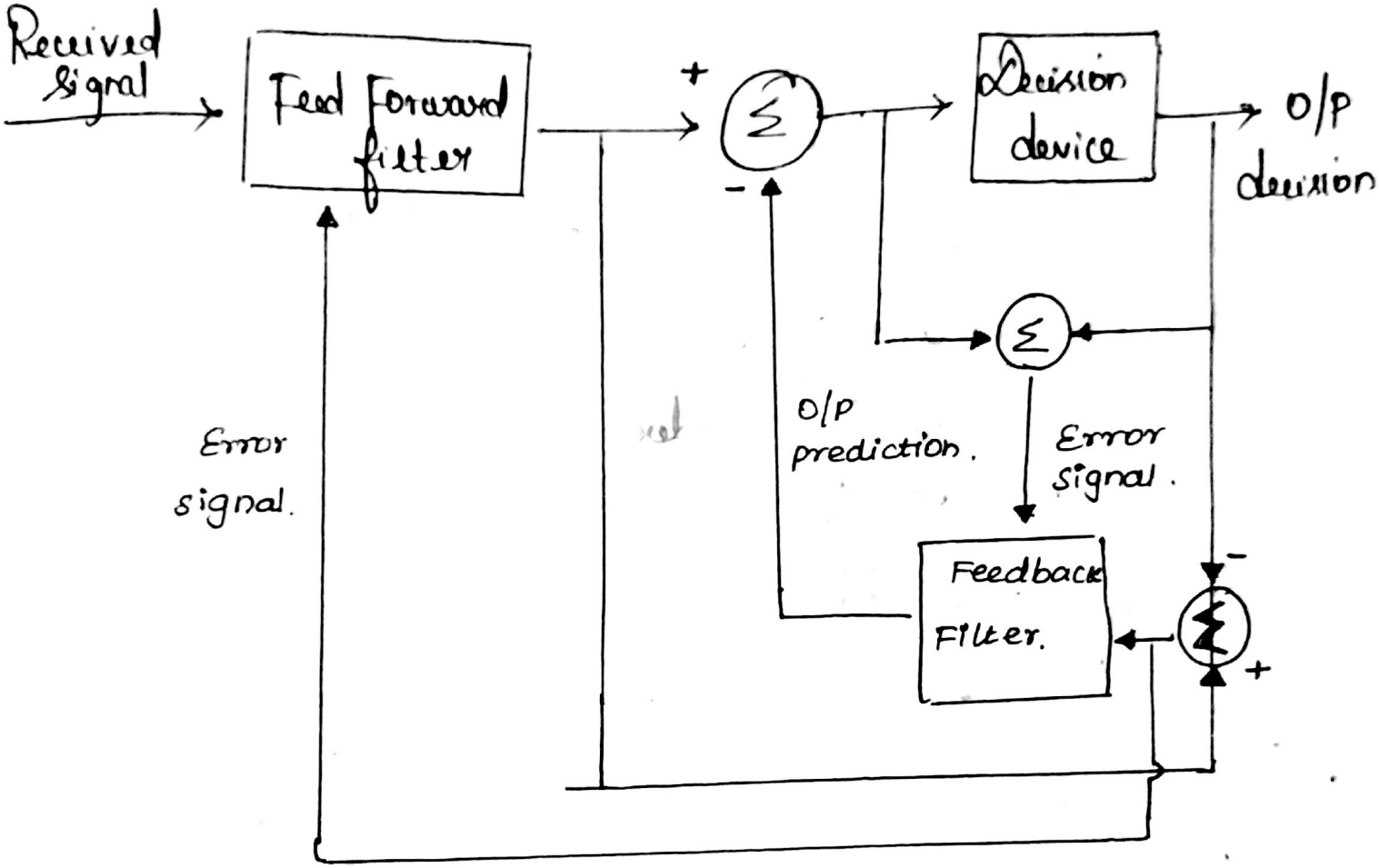
* Hence FBF here is called a noise predictor because it predicts the noise and the residual ISI at the o/p. Also it subtracts the noise from o/p after some feedback delay.

Predictive DFE :-



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Coding Techniques :-

Speech Coding Techniques :-

The goal of speech coding systems is to transmit speech with the highest possible quality using the least possible channel capacity.

When the bit rate is low, more signals are compressed within a given bandwidth.

There are different types of speech coding techniques [based on the method of compression].

* Waveform Coders
* Vocoders

Types of Speech Coders :-

- Waveform Coders
  - Time domain Methods :-
    1) Non differential PCM
    2) Differential [delta, ADPCM]
  - Frequency domain Methods :-
    1) SBC [Sub band Coding]
    2) ATC [Adaptive transform Coding]
- Source Coders
  - LPC
  - Vocoders

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Waveform Coders :-

* It is a source independent scheme.
* It reproduces the time waveform of the speech signal as closely as possible.
* Robust for a wide range of speech characteristics and for noisy environments.

Characteristics of Speech Signals :-

* Amplitude follows non-uniform probability density function (Pdf)
* power spectral density is non flat.
* Correlation between adjacent samples of a segment of speech is high
* Voice speech signals are quasiperiodicity in nature.

Classification of Speech Signals :-

Two types :-

* Voiced Sound [Produced by quasiperiodic vibrations of the vocal chord e.g:- "m", "n", "v"]
* Unvoiced Sound [Produced by turbulent air flow through by a constriction. e.g:- "f", "s", "sh"]

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Vocoders :-

Vocoders are a class of Speech Coding Systems that analyze the Voice Signal at the transmitter, transmit parameters derived from the analysis, and then synthesize the voice at the receiver using those parameters.

Characteristics of vocoders :-

* More Complex,
* Gives good bit rate,
* Less Robust.

Different vocoding schemes :-

* Channel vocoders,
* Formant vocoders,
* Cepstrum Vocoders,
* Voice - Excited vocoders,

Channel Vocoders :-

* It was the first analysis - synthesis System for Speech. These vocoders are frequency domain vocoders that determine the envelope of the Speech Signal for a number of frequency bands and then Sample, encode and multiplex these Samples with the encoded outputs of

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the other filters.

* The Sampling is done Synchronously every 10ms to 30ms.
* Along with the energy information about each band, the voiced / unvoiced decision, and the pitch frequency for voiced speech are also transmitted.

Formant vocoders :-

Formants :-

* Peak Amplitude of Speech Signal.

Concepts :-

* The Concept of formant vocoder is similar to channel vocoder.
* This can operate at lower bit rates than the channel vocoder because it uses fewer Control Signals.
* Instead of Sending Samples of the power Spectrum envelope, the formant vocoder transmits the position of the peaks of the spectral envelope.
* This can produce speech at bit rates lower than 1200 bits/sec.

Cepstrum Vocoders :-

* It Separates the excitation and vocal tract Spectrum by inverse Fourier transforming of the log magnitude spectrum to produce the cepstrum of the Signal.
* The low frequency coefficients in the cepstrum Correspond to

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the vocal tract spectral envelope, with the high frequency excitation coefficients forming a periodic pulse train at multiples of the sampling period.

* Linear filtering is performed to separate the vocal tract cepstral coefficients from the excitation coefficients.

* In the receiver, the vocal tract cepstral coefficients are Fourier transformed to produce the vocal tract impulse response.

* By convolving this impulse response with a synthetic excitation signal, the original speech is reconstructed.

Voice - Excited Vocoders :-

* This eliminates the need for pitch extraction and voice detection operations.

* This uses a hybrid combination of PCM transmissions for the low frequency band of speech, combined with channel vocoding of higher frequency bands.

* A pitch signal is generated at the synthesizer by rectifying, bandpass filtering and clipping the baseband signal, thus creating a spectrally flat signal with energy at pitch harmonics.

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Linear predictive vocoders [LPC] :-

* This belong to the time domain class of vocoders.
* This type helps to extract the significant features of speech from the time waveform.
* It is possible to transmit good quality voice at 4.8Kbps and poorer quality voice at even lower rates.

The linear predictive Coding System models the vocal tract as an all pole linear filter with a transfer function described by

$$H(z) = \frac{G}{1 - \sum_{K=1}^{M} b_K z^{-K}}$$

where G is a gain of the filter

z^{-1} represents a unit delay operation.

The excitation to this filter is either a pulse at the pitch frequency or random white noise depending on whether the speech segment is voiced or unvoiced.

The coefficients of all pole filter are obtained in the time domain using linear prediction techniques.

This system transmits only selected characteristics of the speech error signal like gain factor, pitch information and the voiced/unvoiced decision information.

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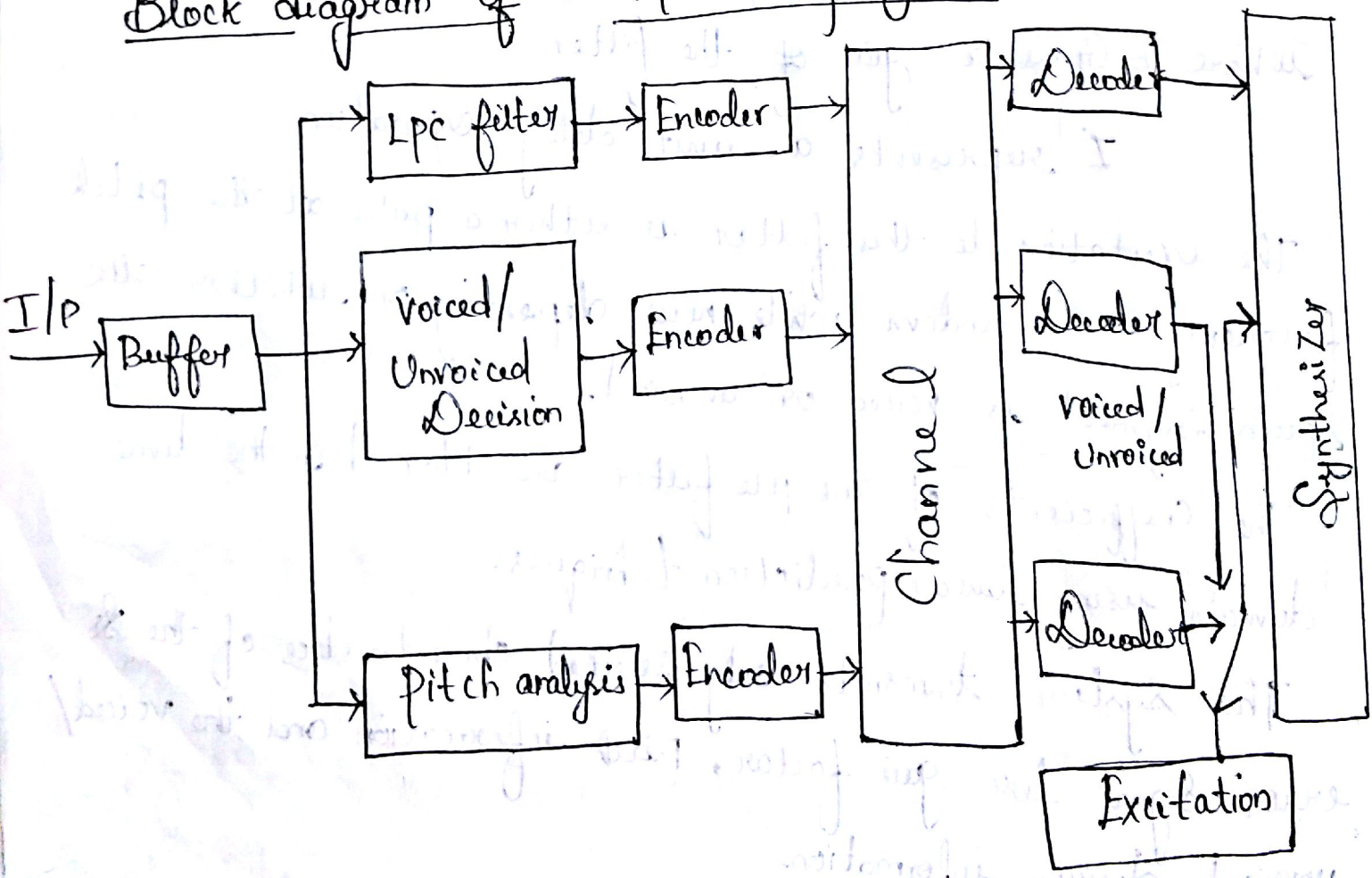
* At the receiver, the received information about the error signal is used to determine the appropriate excitation for the synthesis filter.

* That is, the error signal is the excitation to the decoder.

* The synthesis filter is designed at the receiver using the received predictor coefficients.

* Practically, many LPC coders transmit the filter coefficients which already represent the error signal and can be directly synthesized by the receiver.

Block diagram of a LPC Coding System :-



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This technique requires that the transmitter extract pitch frequency information which is often very difficult.

The phase coherence between the harmonic components of the excitation pulse tends to produce a buzzy twang in the synthesised speech.

These problems are mitigated by the following two methods :-

* Multipulse excited LPC
* Code - Excited LPC

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