Adaptive equalization

- An equalizer is a filter that compensates for the dispersion effects of a channel. Adaptive equalizer can adjust its coefficients continuously during the transmission of data.
- Pre channel equalization
- Requires feed back channel causes burden on transmission
- Equalization is process of correcting channel induced distortion. To realize the full transmission capability of telephone channel, adaptive equalization is needed. Equalizer is said to be adaptive when it adjusts itself continuously during data transmission by operating on the input signal.

Prechannel equalization is used at the transmitter and post channel equalization is used at the receiver. As prechannel equalization requires a feedback channel, adaptive equalization at the receiving side is considered. This equalization can be achieved before data transmission by training the filter with suitable training sequence transmitted through channel so as to adjust the filter parameters to optimal values.

The adaptive equalizer consists of a tapped delay line filter with 100 taps or more and its coefficients are updated according to LMS algorithm. The adjustments to the filter coefficients are made in a step by step fashion synchronously with the incoming data

Post channel equalization

Achieved prior to data transmission by training the filter with the guidance of a training sequence transmitted through the channel so as to adjust the filter parameters to optimum values.

Adaptive equalization

It consists of tapped delay line filter with set of delay elements, set of adjustable multipliers connected to the delay line taps and a summer for adding multiplier outputs in which the coefficients of filter are to be optimized using algorithms

$$Y(nt)=\sum C_i \, x(nT-iT)$$

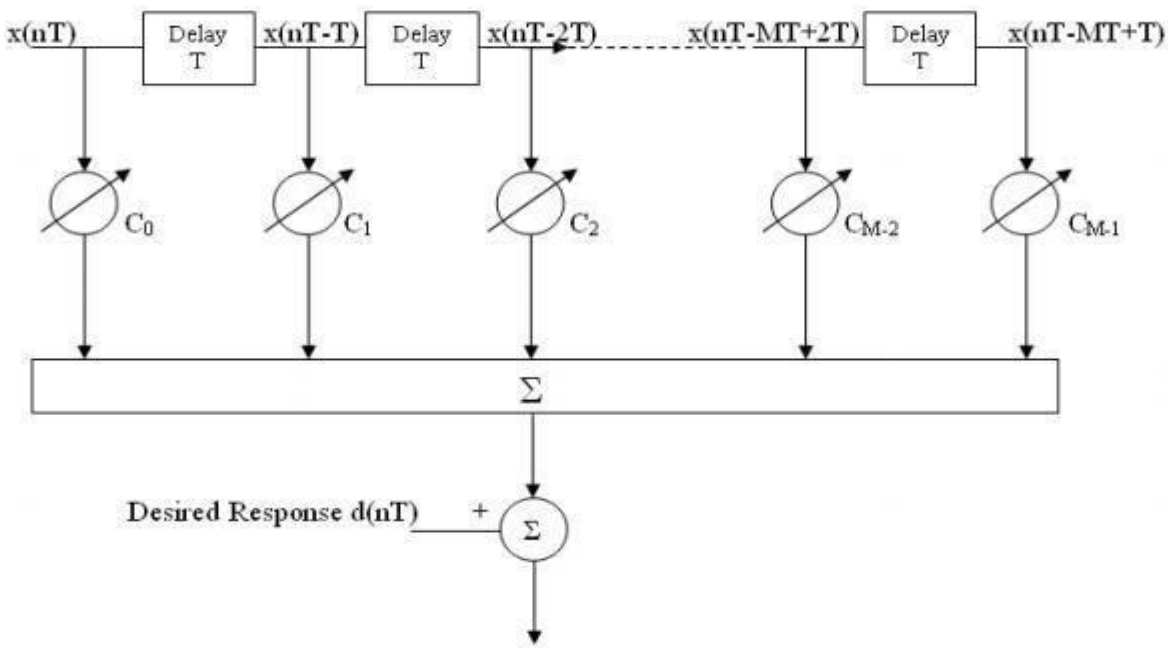


Fig 3.3.1 Adaptive equalization (Source:Brainkart)

Mechanism of adaptation

Fig 3.3.2 Mechanism of adaptation (Source: Brainkart)

Modes of operation:

(i) Training period mode (ii) decision directed mode.

Training mode

A known sequence d(nT) is transmitted and synchronized version of it is generated in the receiver applied to adaptive equalizer. This training sequence has maximal length PN Sequence, because it has large average power and large SNR, resulting response sequence (Impulse) is observed by measuring the filter outputs at the sampling instants. The difference between resulting response y(nT) and desired response d(nT)is error signal which is used to estimate the direction

During the training period, a known sequence is transmitted and a synchronized version of the signal is generated in the receiver .It is applied to the adaptive equalizer as the desired response. The training sequence may be Pseudo Noise sequence and the length of the training sequence may be equal to or greater than the length of adaptive equalizer.

When the training period is completed adaptive equalizer is switched to decision directed mode.

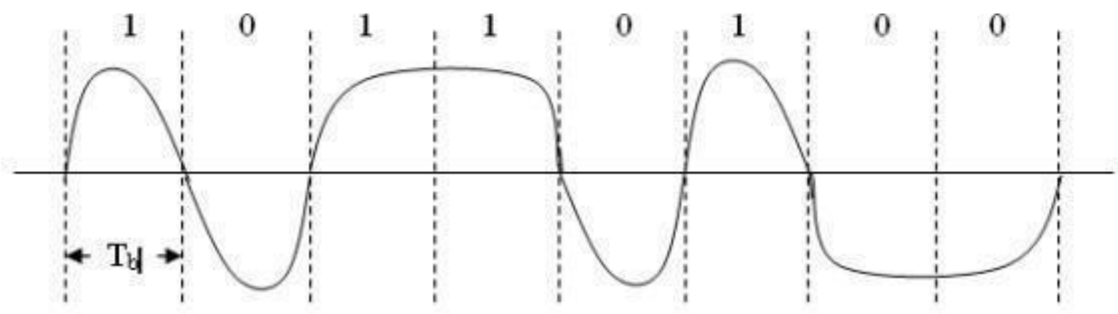
EYE PATTERN

The quality of digital transmission systems are evaluated using the bit error rate. Degradation of quality occurs in each process modulation, transmission, and detection. The eye pattern is experimental method that contains all the information concerning the degradation of quality. Therefore, careful analysis of the eye pattern is important in analyzing the degradation mechanism.

- Eye patterns can be observed using an oscilloscope. The received wave is applied to the vertical deflection plates of an oscilloscope and the saw tooth wave at a rate equal to transmitted symbol rate is applied to the horizontal deflection plates, resulting display is eye pattern as it resembles humaneye.

- The eye pattern, also referred to as the eye diagram, is produced by the synchronized superposition of (as many as possible) successive symbol intervals of the distorted waveform appearing at the output of the receive filter prior to thresholding. As an illustrative example, consider the distorted, but noise-free, waveform shown in part a of Figure 8.12. Part b of the figure displays the corresponding synchronized superposition of the waveform's eight binary symbol intervals. The resulting display is called an "eye pattern" because of its resemblance to a human eye. By the same token, the interior of the eye pattern is called the eye opening

- The interior region of eye pattern is called eye opening



(Source:Brainkart)

We get superposition of successive symbol intervals to produce eye pattern as shown below.

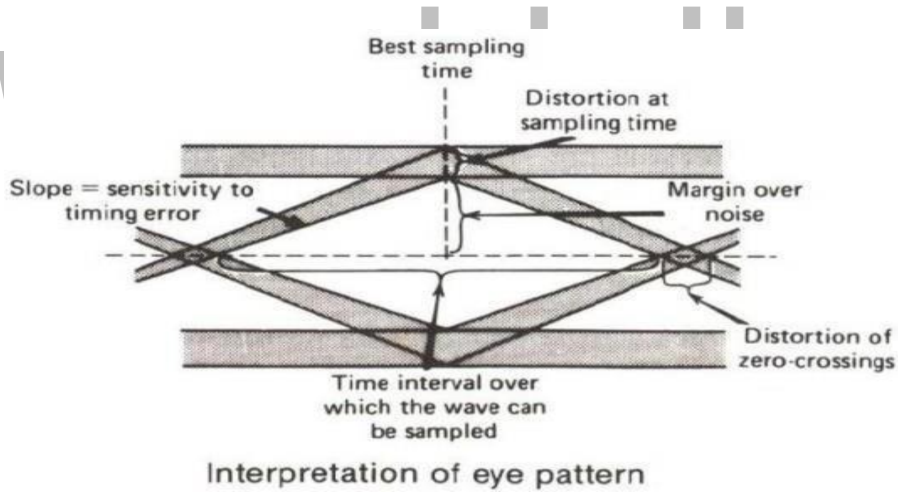


Fig 3.1Eye pattern (Source:Brainkart)

- The width of the eye opening defines the time interval over which the received wave can be sampled without error from ISI
- The optimum sampling time corresponds to the maximum eye opening

- The height of the eye opening at a specified sampling time is a measure of the margin over channel noise.

The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied. Any non linear transmission distortion would reveal itself in an asymmetric or squinted eye. When the effected of ISI is excessive, traces from the upper portion of the eye pattern cross traces from lower portion with the result that the eye is completely closed.

As long as the additive channel noise is not large, then the eye pattern is well defined and may, therefore, be studied experimentally on an oscilloscope. The waveform under study is applied to the deflection plates of the oscilloscope with its time-base circuit operating in a synchronized condition. From an experimental perspective, the eye pattern offers two compelling virtues:

- The simplicity of eye-pattern generation.
- The provision of a great deal of insightful information about the characteristics of the data transmission system. Hence, the wide use of eye patterns as a visual indicator of how well or poorly a data transmission system performs the task of transporting a data sequence across a physical channel.

Timing Features

A generic eye pattern for distorted but noise-free binary data. The horizontal axis, representing time, spans the symbol interval from $-T_b$ 2 to T_b 2, where T_b is the bit duration. From this diagram, we may infer three timing features pertaining to a binary data transmission system, exemplified by a PAM system:

Optimum sampling time. The width of the eye opening defines the time interval over which the distorted binary waveform appearing at the output of the receive filter in the PAM system can be uniformly sampled without decision errors. Clearly, the *optimum sampling time* is the time at which the eye opening is at its widest.

Zero-crossing jitter. In practice, the timing signal (for synchronizing the receiver to the transmitter) is extracted from the *zero-crossings* of the waveform that appears at the receive-filter output. In such a form of synchronization, there will always be irregularities in the zero-crossings, which, in turn, give rise to *jitter* and, therefore, nonoptimum sampling times.

Timing sensitivity. Another timing-related feature is the sensitivity of the PAM system to *timing errors*. This sensitivity is determined by the rate at which the eye pattern is closed as the sampling time is varied.

The Peak Distortion for Intersymbol Interference

Hereafter, we assume that the ideal signal amplitude is scaled to occupy the range from –1 to +1. We then find that, in the absence of channel noise, the eye opening assumes two extreme values:

An eye opening of unity, which corresponds to zero ISI.

An eye opening of zero, which corresponds to a completely closed eye pattern; this second extreme case occurs when the effect of intersymbol interference is severe enough for some upper traces in the eye pattern to cross with its lower traces.

It is indeed possible for the receiver to make decision errors even when the channel is noise free. Typically, *an eye opening of 0.5 or better is considered to yield reliable data transmission*.

In a noisy environment, the extent of eye opening at the optimum sampling time provides a measure of the operating margin over additive channel noise. This measure, as illustrated in Figure 8.13, is referred to as the *noise margin*. From this discussion, it is apparent that the eye opening plays an important role in assessing system performance; hence the need for a formal definition of the eye opening. To this end, we offer the following definition:

$$\text{Eye opening} = 1 - D_{\text{peak}}$$

where D_{peak} denotes a new criterion called the *peak distortion*. The point to note here is that peak distortion is a *worst-case* criterion for assessing the effect of ISI on the performance (i.e., error rate) of a data transmission system. The relationship between the eye opening and peak distortion is illustrated in Figure 8.14. With the eye opening being dimensionless, the peak distortion is dimensionless too. To emphasize this statement, the two extreme values of the eye opening translate as follows:

Zero peak distortion, which occurs when the eye opening is unity.

Unity peak distortion, which occurs when the eye pattern is completely closed.

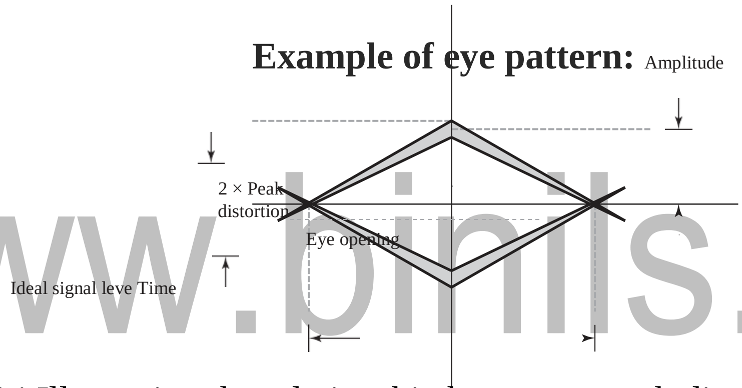


Figure 8.14 Illustrating the relationship between peak distortion and eye opening.

Note: the ideal signal level is scaled to lie inside the range –1 to +1.

(Source: S. Haykin, ―Digital Communications‖, John Wiley, 2005- Page- 465)

Eye Patterns for *M*-ary Transmission

By definition, an *M*-ary data transmission system uses *M* encoded symbols in the transmitter and *M* – 1 thresholds in the receiver. Correspondingly, the eye pattern for an *M*-ary data transmission system contains *M* – 1 eye openings stacked vertically one on top of the other. The thresholds are defined by the amplitude-transition levels as we move up from one eye opening to the adjacent eye opening. When the encoded symbols are all equiprobable, the thresholds will be equidistant from each other. In a strictly linear data transmission system with truly transmitted random data sequences, all the $M - 1$ eye openings would be identical. In practice, however, it is often possible to find asymmetries in the eye pattern of an M-ary data transmission system, which are caused by nonlinearities in the communication channel or other distortion- sensitive parts of the system.

Binary-PAM Perfect channel (no noise and no ISI)

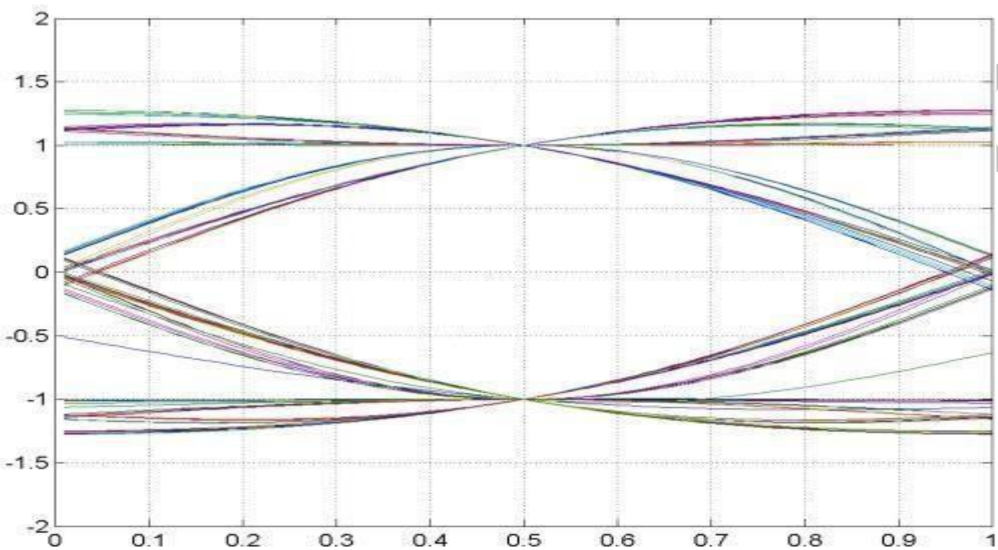
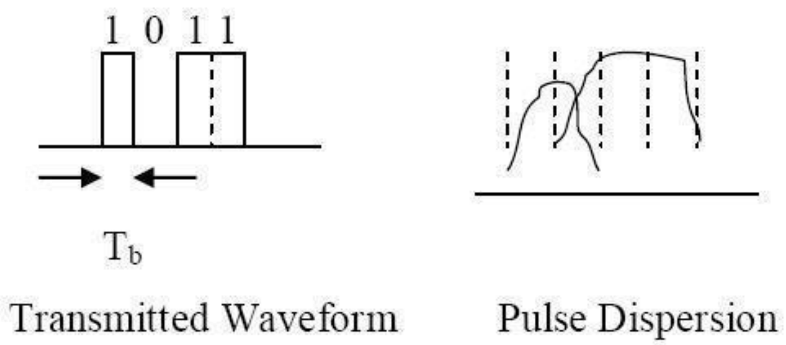


Fig 3.2 Example of eye pattern: Binary-PAM with noise no ISI (Source:Brainkart

Inter symbol Interference

Generally, digital data is represented by electrical pulse, communication channel is always band limited. Such a channel disperses or spreads a pulse carrying digitized samples passing through it. When the channel bandwidth is greater than bandwidth of pulse, spreading of pulse is very less. But when channel bandwidth is close to signal bandwidth, i.e. if we transmit digital data which demands more bandwidth which exceeds channel bandwidth, spreading will occur and cause signal pulses to overlap. This overlapping is called **Inter Symbol Interference**. In short it is called ISI. Similar to interference caused by other sources, ISI causes degradations of signal if left uncontrolled. This problem of ISI exists strongly in Telephone channels like coaxial cables and optical fibers.

The main objective is to study the effect of ISI, when digital data is transmitted through band limited channel and solution to overcome the degradation of waveform by properly shaping pulse



(Source:Brainkart)

The effect of sequence of pulses transmitted through channel is shown in fig. The Spreading of pulse is greater than symbol duration, as a result adjacent pulses interfere. i.e. pulses get completely smeared, tail of smeared pulse enter into adjacent symbol intervals making it difficult to decide actual transmitted pulse. First let us have look at different formats of transmitting digital data.

In base band transmission best way is to map digits or symbols into pulse waveform. This waveform is generally termed as **Line codes**. To proceed with a mathematical study of intersymbol interference, consider a *baseband binary PAM system*, a generic form of which is depicted in Figure. The term "baseband" refers to an information-bearing signal whose spectrum extends from (or near)

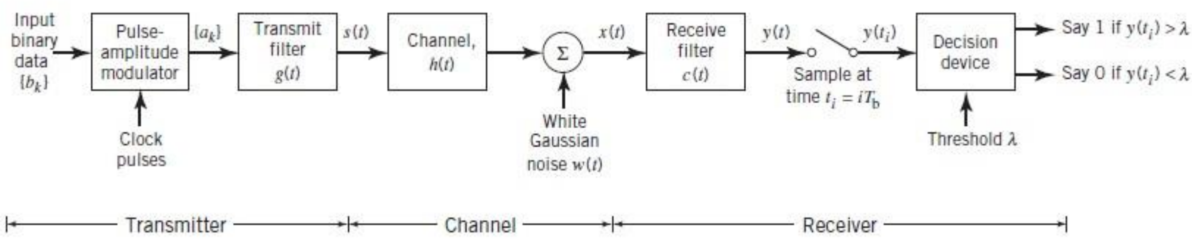


Fig: Baseband Binary data Transmission Systems

(Source: S. Haykin, ―Digital Communications‖, John Wiley, 2005-Page- 448)

The *pulse-amplitude modulator* changes the input binary data stream {*bk*} into a new sequence of short pulses, short enough to approximate impulses. More specifically, the pulse amplitude *ak* is represented in the polar form:

$$a_k = \begin{cases} +1 \text{ if } b_k \text{ is symbol } 1 \\ -1 \text{ if } b_k \text{ is symbol } 0 \end{cases}$$

The sequence of short pulses so produced is applied to a *transmit filter* whose impulse response is denoted by *g*(*t*). The transmitted signal is thus defined by the sequence

$$s(t) = \sum_k a_k g(t - kT_b)$$

A binary data stream represented by the sequence {*ak*}, where *ak* = +1 for symbol 1 and *ak* = –1 for symbol 0, *modulates* the basis pulse *g*(*t*) and superposes *linearly* to form the transmitted signal *s*(*t*). The signal *s*(*t*) is naturally modified as a result of transmission through the *channel* whose impulse response is denoted by *h*(*t*). The noisy received signal *x*(*t*) is passed through a *receive filter* of impulse response *c*(*t*). The resulting filter output *y*(*t*) is sampled *synchronously* with the transmitter, with the sampling instants being determined by a *clock* or *timing signal* that is usually extracted from the receive-filter output. Finally, the sequence of samples thus obtained is used to reconstruct the original data sequence by means of a *decision device*. Specifically, the amplitude of each sample is compared with a *zero threshold*, assuming that the symbols 1 and 0 are equiprobable. If the zero threshold is exceeded, a decision is made in favor of symbol 1; otherwise a decision is made in favor of symbol 0. If the sample amplitude equals the zero threshold exactly, the receiver simply makes a random guess.

Except for a trivial scaling factor, we may now express the receive filter output as

$$y(t) = \sum_k a_k p(t - kT_b)$$

where the pulse *p*(*t*) is to be defined. To be precise, an arbitrary time delay *t*0 should be included in the argument of the pulse *p*(*t* – *kT*b) in (8.6) to represent the effect of transmission delay through the system. To simplify the exposition, we have put this delay equal to zero in (8.6) without loss of generality; moreover, the channel noise is ignored. The scaled pulse *p*(*t*) is obtained by a double convolution involving the impulse response *g*(*t*) of the transmit filter, the impulse response *h*(*t*) of the channel, and the impulse response *c*(*t*) of the receive filter, as shown by

$$p(t) = g(t)\star h(t)\star c(t)$$

where, as usual, the star denotes convolution. We assume that the pulse *p*(*t*) is *normalized* by setting which justifies the use of a scaling factor to account for amplitude changes incurred in the course of signal transmission through the system. Since convolution in the time domain is transformed into multiplication in the frequency domain, we may use the Fourier transform to change (8.7) into the equivalent form

$$P(f) = G(f)H(f)C(f)$$

where *P*(*f*), *G*(*f*), *H*(*f*), and *C*(*f*) are the Fourier transforms of *p*(*t*), *g*(*t*), *h*(*t*), and *c*(*t*), respectively.

The receive filter output *y*(*t*) is sampled at time *ti* = *iT*b, where *i* takes on integer values,

$$y(t_i) = \sum_{k=-\infty}^{\infty} a_k p[(i-k)T_b]$$

$$= a_i + \sum_{\substack{k=-\infty \\ k \neq i}}^{\infty} a_k p[(i-k)T_b]$$

The first term *ai* represents the contribution of the *i*th transmitted bit. The second term represents the residual effect of all other transmitted bits on the decoding of the *i*th bit. This residual effect due to the occurrence of pulses before and after the sampling instant *ti* is called *intersymbol interference* (ISI).

In the absence of ISI—and, of course, channel noise—we observe from (8.10) that the summation term is zero, thereby reducing the equation to which shows that, under these ideal conditions, the *i*th transmitted bit is decoded correctly.

$$y(t_i) = a_i$$

Nyquist criterion for distortion less transmission

Consider then the sequence of samples {*p*(*nT*b)}, where *n* = 0,

In particular, we may write

$$P_\delta(f) = R_b \sum_{n=-\infty}^{\infty} P(f - nR_b)$$

where *R*b = 1□*T*b is the *bit rate* in bits per second; *P*□□(*f*) on the left-hand side of (8.12) is the Fourier transform of an infinite periodic sequence of delta functions of period *T*b whose individual areas are weighted by the respective sample values of *p*(*t*). That is, *p* (*f*) is given by

$$P_\delta(f) = \int_{-\infty}^{\infty} \sum_{m=-\infty}^{\infty} [p(mT_b)\delta(t - mT_b)]\exp(-j2\pi ft)\, dt$$

Let the integer *m* = *i* – *k*. Then, *i* = *k* corresponds to *m* = 0 and, likewise, *i* □ *k* corresponds to *m* □ 0 . Accordingly, imposing the conditions of (8.11) on the sample values of *p*(*t*) in the integral in (8.13), we get

$$P_\delta(f) = p(0)\int_{-\infty}^{\infty} \delta(t)\exp(-j2\pi ft)\, dt$$

$$= p(0)$$

where we have made use of the sifting property of the delta function. Since from (8.8) we have $p(0) = 1$, it follows from (8.12) and (8.14) that the frequency-domain condition for zero ISI is satisfied, provided that

$$\sum_{n=-\infty}^{\infty} P(f - nR_b) = T_b$$

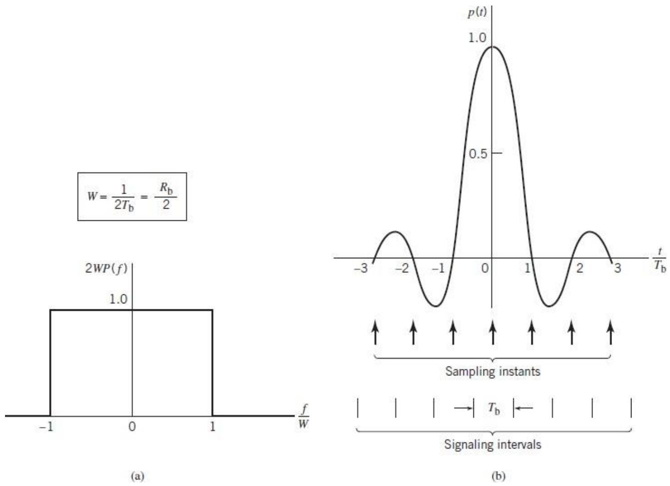
where $T_b = 1/R_b$. We may now make the following statement on the *Nyquist criterion*[1] *for distortionless baseband transmission* in the frequency domain:

The frequency function $P(f)$ eliminates intersymbol interference for samples taken at intervals T_b Correspondingly, the baseband pulse $p(t)$ for distortionless transmission described in (8.18) is called the *ideal Nyquist pulse*, ideal in the sense that the bandwidth requirement is one half the bit rate.

There are two practical difficulties that make it an undesirable objective for signal design:

1. It requires that the magnitude characteristic of $P(f)$ be flat from $-W$ to $+W$, and zero elsewhere. This is physically unrealizable because of the abrupt transitions at the band edges $\pm W$, in that the Paley–Wiener criterion discussed in Chapter 2 is violated.

2. The pulse function $p(t)$ decreases as $1/|t|$ for large $|t|$, resulting in a slow rate of decay. This is also caused by the discontinuity of $P(f)$ at $\pm W$. Accordingly, there is practically no margin of error in sampling times in the receiver.



r.

Fig: a) Ideal Magnitude Response b)Ideal basic pulse shape

(Source: S. Haykin, ―Digital Communications‖, John Wiley, 2005-Page- 452)

Pulse Shaping

It is the process of changing the waveform of transmitted pulses. Its purpose is to make the transmitted signal better suited to its purpose or the communication channel, typically by limiting the effective bandwidth of the transmission. By filtering the transmitted pulses this way, the inter symbol interference caused by the channel can be kept in control. In RF communication, pulse shaping is essential for making the signal fit in its frequency band.

Typically pulse shaping occurs after line coding and modulation.

Need for pulse shaping

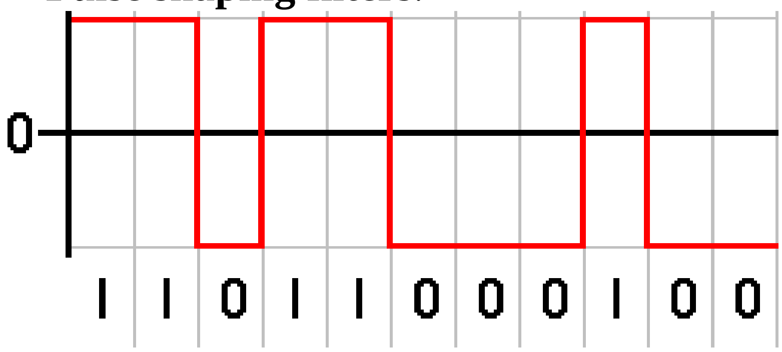
Transmitting a signal at high modulation rate through a band-limited channel can create inter symbol interference. As the modulation rate increases, the signal's bandwidth increases. When the signal's bandwidth becomes larger than the channel bandwidth, the channel starts to introduce distortion to the signal. This distortion usually manifests itself as inter symbol interference.

The signal's spectrum is determined by the modulation scheme and data rate used by the transmitter, but can be modified with a pulse shaping filter. Usually the transmitted symbols are represented as a time sequence of dirac delta pulses. This theoretical signal is then filtered with the pulse shaping filter, producing the transmitted signal.

In many base band communication systems the pulse shaping filter is implicitly a boxcar filter. Its Fourier transform is of the form *sin(x)/x*, and has significant signal power at frequencies higher than symbol rate. This is not a big problem when optical fibre or even twisted pair cable is used as the communication channel. However, in RF communications this would waste bandwidth, and only tightly specified frequency bands are used for single transmissions. In other words, the channel for the signal is band-limited. Therefore better filters have been developed, which attempt to minimize the bandwidth needed for a certain symbol rate.

An example in other areas of electronics is the generation of pulses where the rise time need to be short; one way to do this is to start with a slower-rising pulse, and decrease the rise time, for example with a step recovery diode circuit

Pulse shaping filters:



(Source:Brainkart)

A typical NRZ coded signal is implicitly filtered with a sinc filter.

Not every filter can be used as a pulse shaping filter. The filter itself must not introduce inter symbol interference — it needs to satisfy certain criteria. The Nyquist ISI criterion is a commonly used criterion for evaluation, because it relates the frequency spectrum of the transmitter signal to intersymbol interference.

Examples of pulse shaping filters that are commonly found in communication systems are:

- Sinc shaped filter
- Raised-cosine filter
- Gaussian filter

Sender side pulse shaping is often combined with a receiver side matched filter to achieve optimum tolerance for noise in the system. In this case the pulse shaping is equally distributed between the sender and receiver filters. The filters' amplitude responses are thus point wise square roots of the system filters.

Other approaches that eliminate complex pulse shaping filters have been invented. In OFDM, the carriers are modulated so slowly that each carrier is virtually unaffected by the bandwidth limitation of the channel.

Sinc filter

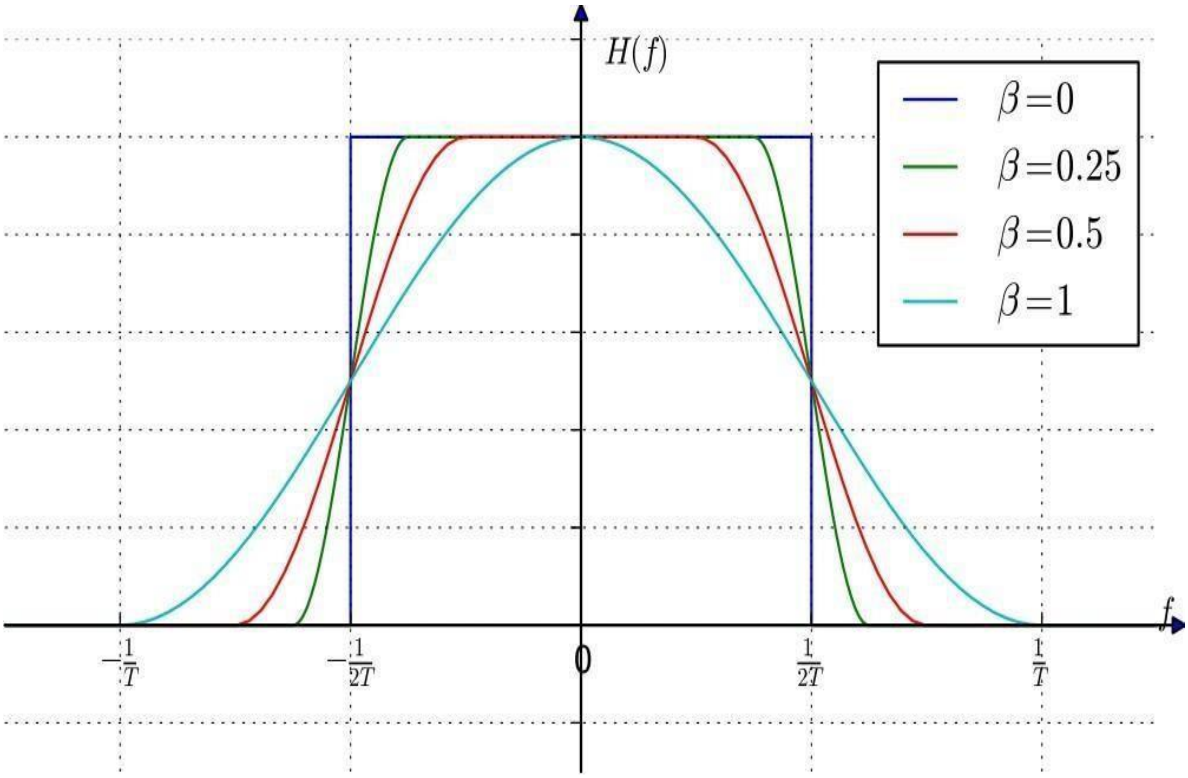


Fig 3.5 Amplitude response of raised-cosine filter with various roll-off factors
(Source:Brainkart)

It is also called as Boxcar filter as its frequency domain equivalent is a rectangular shape. Theoretically the best pulse shaping filter would be the sinc filter, but it cannot be implemented precisely. It is a non-causal filter with relatively slowly decaying tails. It is also problematic from a synchronization point of view as any phase error results in steeply increasing inter symbol interference.

Raised-cosine filter

Raised-cosine is similar to sinc, with the tradeoff of smaller side lobes for a slightly larger spectral width. Raised-cosine filters are practical to implement and they are in wide use. They have a configurable excess bandwidth, so communication systems can choose a trade off between a simpler filter and spectral efficiency.

Gaussian filter

This gives an output pulse shaped like a Gaussian function.

Nyquist criterion

When the baseband filters in the communication system satisfy the Nyquist criterion, symbols can be transmitted over a channel with flat response within a limited frequency band, without ISI. Examples of such baseband filters are the raised-cosine filter, or the sinc filter as the idealcase.

Correlative Coding

So far, we've discussed that ISI is an unwanted phenomenon and degrades the signal. But the same ISI if used in a controlled manner, is possible to achieve a bit rate of **2W** bits per second in a channel of bandwidth **W** Hertz. Such a scheme is called as **Correlative Coding** or **Partial response signaling schemes**.

Since the amount of ISI is known, it is easy to design the receiver according to the requirement so as to avoid the effect of ISI on the signal. The basic idea of correlative coding is achieved by considering an example of **Duo-binary Signaling**.

Duo-binary Signaling:

The name duo-binary means doubling the binary system's transmission capability. To understand this, let us consider a binary input sequence $\{a_k\}$ consisting of uncorrelated binary digits each having a duration T_a seconds. In this, the signal **1** is represented by a **+1** volt and the symbol **0** by a **-1** volt.

Therefore, the duo-binary coder output c_k is given as the sum of present binary digit a_k and the previous value a_{k-1} as shown in the following equation.

$$ck=ak+ak-1ck=ak+ak-1$$

The above equation states that the input sequence of uncorrelated binary sequence $\{a_k\}$ is changed into a sequence of correlated three level pulses $\{c_k\}$. This correlation between the pulses may be understood as introducing ISI in the transmitted signal in an artificial manner.