This property follows from Property (1). As the auto-correlation function and the energy spectral density form F.T. pair, by taking IFT of above equation, we may write,

$$\varphi_0(t) = R_\varphi(t-T)$$

Where $R_\phi(\tau)$ is ACF of $\Phi_0(t)$. At $t=T$, we get $R(0)$=energy of the signal.

Property (3): The output SNR of a matched filter depends only on the ratio of the signal energy to the psd of the white noise at the filter input.

Let us consider a filter matched to the input signal $\Phi(t)$

From property (2), we see that the maximum value of $\Phi_0(t)$at $t=T$ is$\Phi_0(T-t)=E$

Now, it may be shown that the average noise power at the output of the matched filter is given by,

$$E[n^2(t)] = \frac{N_0}{2}\int_{-\infty}^{\infty}|\varphi(f)|^2\,df = \frac{N_0}{2}.E$$

The maximum signal power

$$= |\varphi_0(T)|^2 = E^2$$

Hence

$$(SNR)_{\max} = \frac{E^2}{\frac{N_0}{2}E} = \frac{2E}{N_0}$$

Note that SNR i[...]

by,

$$E[n^2(t)] = \frac{N_0}{2}\int_{-\infty}^{\infty}|\varphi(f)|^2\,df = \frac{N_0}{2}E$$

The maximum signal power

$$= |\varphi_0(T)|^2 = E^2$$

Hence

$$(SNR)_{\max} = \frac{E^2}{\frac{N_0}{2}E} = \frac{2E}{N_0}$$

Note that SNR in the above expression is a dimensionless quantity.

This is a very significant result as we see that the SNR_{max} depends on E and N_0 but not on the shape of $\phi(t)$ This means a freedom to the designer to select specific pulse shape to optimize other design requirement (the most usual requirement being the spectrum or, equivalently, the transmission bandwidth) while ensuring same SNR.

Property (4): The matched-filtering operation may be separated into two matching condition: namely, spectral phase matching that produces the desired output peak at t = T and spectral amplitude matching that gives the peak value its optimum SNR.

$$\Phi(f) = |\Phi(f)|\exp[j\theta(f)]$$

The filter is said to be matched to the signal $\phi(t)$ in spectral phase if the transfer function of the filter follows:

$$E[n^2(t)] = \frac{N_0}{2}\int_{-\infty}^{\infty}|\varphi(f)|^2\,df = \frac{N_0}{2}E$$

The maximum signal power

$$= |\varphi_0(T)|^2 = E^2$$

Hence

$$(SNR)_{\max} = \frac{E^2}{\frac{N_0}{2}E} = \frac{2E}{N_0}$$

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$$\Phi(f) = |\Phi(f)|\exp[j\theta(f)]$$

The filter is said to be matched to the signal $\phi(t)$ in spectral phase if the transfer function of the filter follows:

$$H(f) = |H(f)|\exp[-j\theta(f) - j2\pi fT]$$

Here H (f) is real

INTERSYMBOL INTERFERENCE

avya Holla

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INTERSYMBOL INTERFERENCE (ISI)

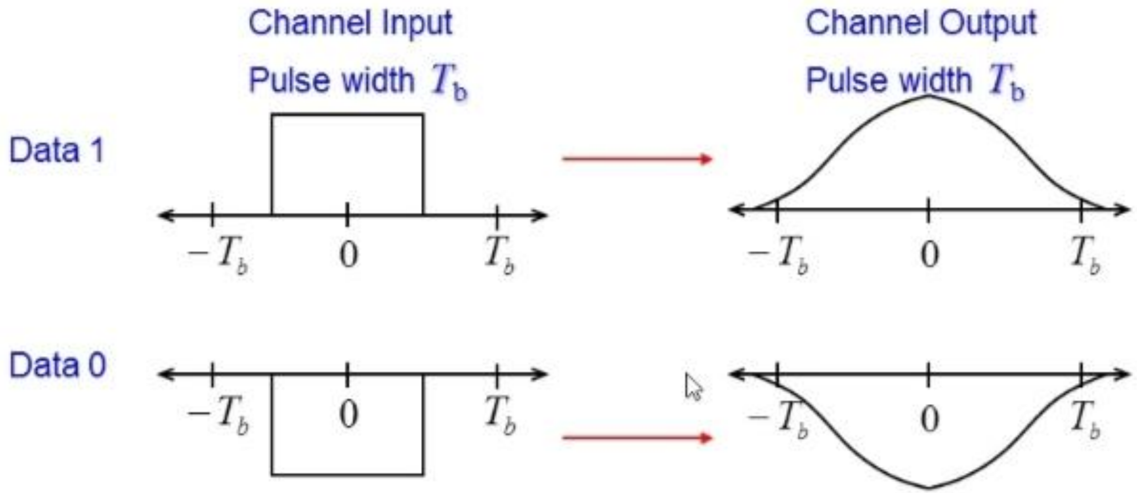
- Intersymbol Interference
- ISI on Eye Patterns

Navya Holla

Intersymbol Interference

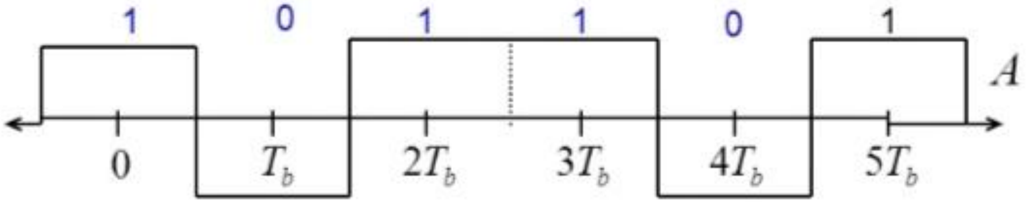
- **Intersymbol interference (ISI)** occurs when a pulse spreads out in such a way that it interferes with adjacent pulses *at the sample instant*.

- Example: assume polar NRZ line code. The channel outputs are shown as spreaded (width T_b becomes $2T_b$) pulses shown (Spreading due to bandlimited channel characteristics).

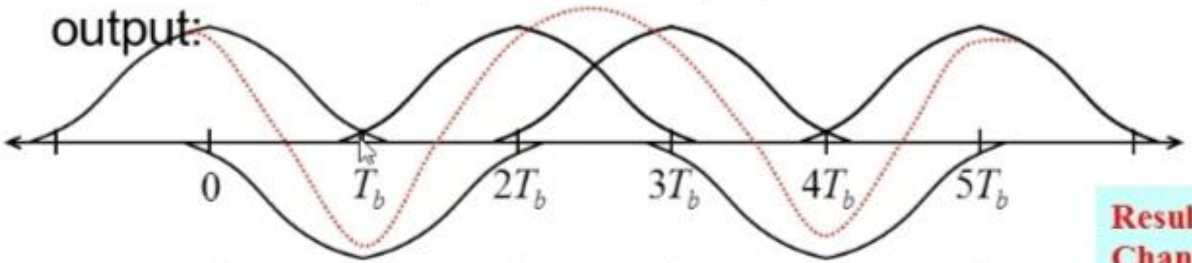


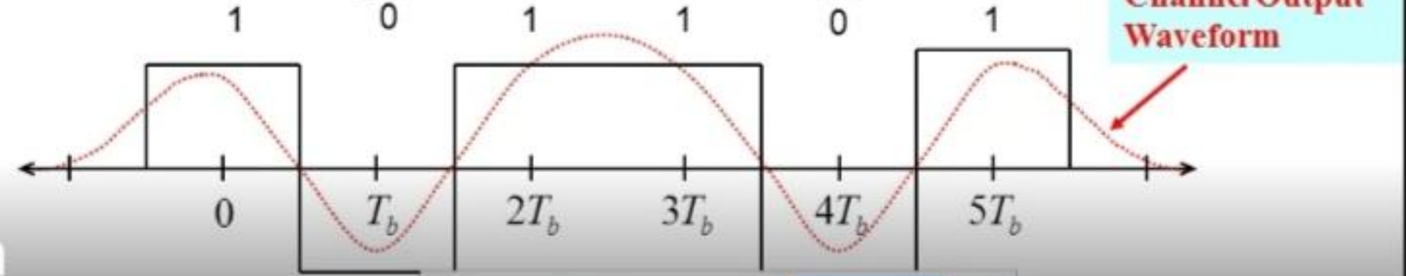
Intersymbol Interference

- For the input data stream:



- The channel output is the superposition of each bit's output:





6.3 INTERSYMBOL INTERFERENCE

Consider Fig. 6.5, which depicts the basic elements of a *baseband binary PAM system*. The input signal consists of a binary data sequence $\{b_k\}$ with a bit duration of T_b seconds. This sequence is applied to a pulse generator, producing the discrete PAM signal

$A_k \to$ random variable

$a_k \to$ sample value

$$x(t) = \sum_{k=-\infty}^{\infty} a_k v(t - kT_b) \tag{6.18}$$

where $v(t)$ denotes the basic pulse, normalized such that $v(0) = 1$, as in Eq. 6.2. The coefficient a_k depends on the input data and the type of format used. The waveform $x(t)$ represents one realization of the random process $X(t)$ considered in Section 6.2. Likewise, a_k is a sample value of the random variable A_k.

The PAM signal $x(t)$ passes through a transmitting filter of transfer function $H_T(f)$. The resulting filter output defines the transmitted signal, which is modified as a result of transmission through the channel of transfer function $H_C(f)$. The channel may represent a coaxial cable or optical fiber, where the major source of system degradation is *dispersion* in the channel. In any event, for the present we assume that the channel is *noiseless* but dispersive. The channel output is passed through a receiving filter of transfer function $H_R(f)$. This filter output is sampled synchronously with the transmitter, with the sampling instants being determined by a clock or timing signal that is usually extracted from the receiving filter output. Finally, the sequence of samples thus obtained is used to reconstruct the original data sequence by means of a decision device. Each sample is compared to a threshold. We assume that symbols 1 and 0 are equally likely, and the threshold is set half way between their representation levels. If the threshold is exceeded, a decision is made in favor of symbol 1. If, on the other hand, the threshold is not exceeded, a decision is made in favor of symbol 0. If the [illegible] fair coin will determine [illegible]

The receiving [illegible]

output is sampled synchronously with the transmitter, with the sampling instants being determined by a clock or timing signal that is usually extracted from the receiving filter output. Finally, the sequence of samples thus obtained is used to reconstruct the original data sequence by means of a decision device. Each sample is compared to a threshold. We assume that symbols 1 and 0 are equally likely, and the threshold is set half way between their representation levels. If the threshold is exceeded, a decision is made in favor of symbol 1. If, on the other hand, the threshold is not exceeded, a decision is made in favor of symbol 0. If the sample value equals the threshold exactly, the flip of a fair coin will determine which symbol was transmitted.

The receiving filter output may be written as*

$$y(t) = \mu \sum_{k=-\infty}^{\infty} a_k p(t - kT_b) \tag{6.19}$$

where μ is a scaling factor and the pulse $p(t)$ is *normalized* such that

$$p(0) = 1 \tag{6.20}$$

* To be precise, an arbitrary time delay t_0 should be included in the argument of the pulse $p(t - kT_b)$ in Eq. 6.19 to represent the effect of transmission delay through the system. For convenience, we have put this delay equal to zero in Eq. 6.19.

The output $y(t)$ is produced in response to the binary data waveform applied to the input of the transmitting filter. Specifically, the pulse $\mu p(t)$ is the response of the cascade connection of the transmitting filter, the channel, and the receiving filter, which is produced by the pulse $v(t)$ applied to the input of this cascade connection. Therefore, we may relate $p(t)$ to $v(t)$ in the frequency domain by writing

$$\mu P(f) = V(f)H_T(f)H_C(f)H_R(f) \tag{6.21}$$

where $P(f)$ and $V(f)$ are the Fourier transforms of $p(t)$ and $v(t)$, respectively. Note that the normalization of $p(t)$ as in Eq. 6.20 means that the total area under the curve of $P(f)$ equals unity.

The receiving filter output $y(t)$ is sampled at time $t_i = iT_b$ (with i taking on integer values), yielding

$$\begin{aligned} y(t_i) &= \mu \sum_{k=-\infty}^{\infty} a_k p(iT_b - kT_b) \\ &= \mu a_i + \mu \sum_{\substack{k=-\infty \\ k \neq i}}^{\infty} a_k p(iT_b - kT_b) \end{aligned} \tag{6.22}$$

In Eq. 6.22, the first term μa_i is produced by the ith transmitted bit. The second term represents the residual effect of all other transmitted bits on the decoding of the ith bit; this residual effect is called *intersymbol interference* (ISI).

In physical terms, ISI arises because of imperfections in the overall frequency response of the system. When a short pulse of duration T_b seconds is transmitted through a band-limited system, the frequency components constituting the input pulse are differentially attenuated and, more significantly, differentially delayed by the system. Consequently, the pulse appearing at the output of the system is *dispersed* over an interval longer than T_b seconds. Thus, when a sequence of short pulses (representing binary 1s and 0s) are transmitted through the system, one pulse every T_b seconds, the dispersed responses originating from different symbol intervals will interfere with each other, thereby resultir

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ISI on Eye Patterns

- The amount of ISI can be seen on an oscilloscope using an ***Eye Diagram*** or ***Eye pattern***.

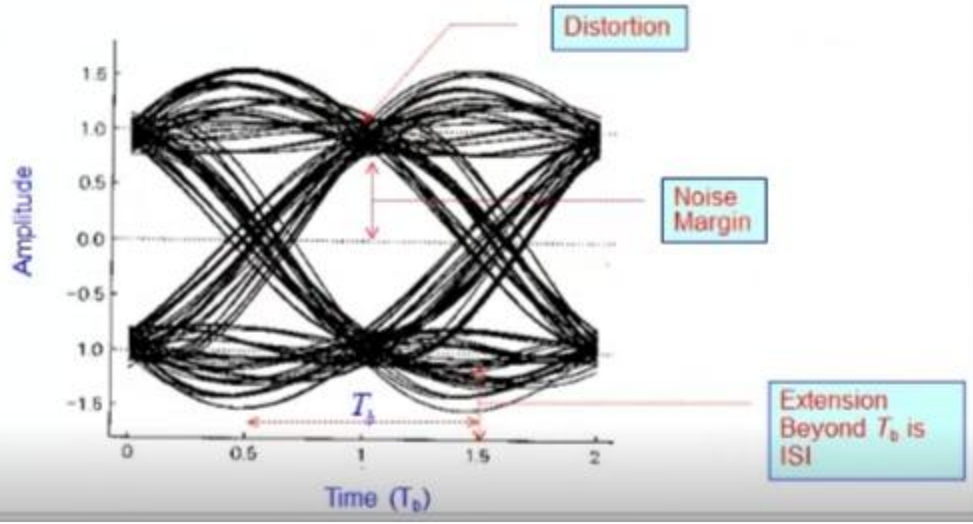

Figure 6.18 (*a*) Distorted binary wave. (*b*) Eye pattern.

ence. It is apparent that the preferred time for sampling is the instant of time at which the eye is open widest.

2. The sensitivity of the system to timing error is determined by the rate of closure of the eye as the sampling time is varied.
3. The height of the eye opening, at a specified sampling time, defines the margin over noise.

When the effect of intersymbol interference is severe, traces from the upper portion of the eye pattern cross traces from the lower portion, with the result that the eye is completely closed. In such a situation, it is impossible to avoid errors due to the combined presence of intersymbol interference and noise in the system, and a solution has to be found to correct for them.

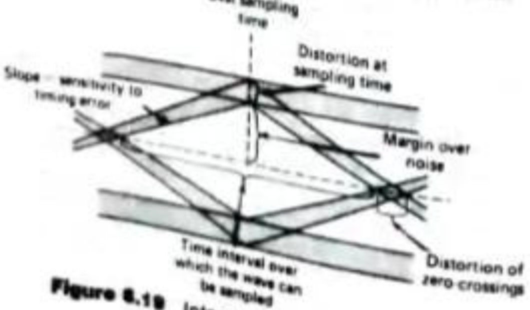

Figure 6.19 Interpretation of eye