concerned, we may consider the data transmission ... band.

However, in a *switched telephone network*,* we find that two factors contribute to pulse distortion on different link connections: (1) differences in the transmission characteristics of the individual links that may be switched together, and (2) differences in the number of links in a connection. The result is that the telephone channel is random in the sense of being one of an ensemble of possible channels. Consequently, the use of a fixed pair of transmitting and receiving filters designed on the basis of average channel characteristics may not adequately reduce intersymbol interference. To realize the full transmission capability of a telephone channel, there is need for *adaptive equalization*.† By equalization we mean the process of correcting channel-induced distortion. This process is said to be adaptive when it adjusts itself continuously during data transmission by operating on the input signal.

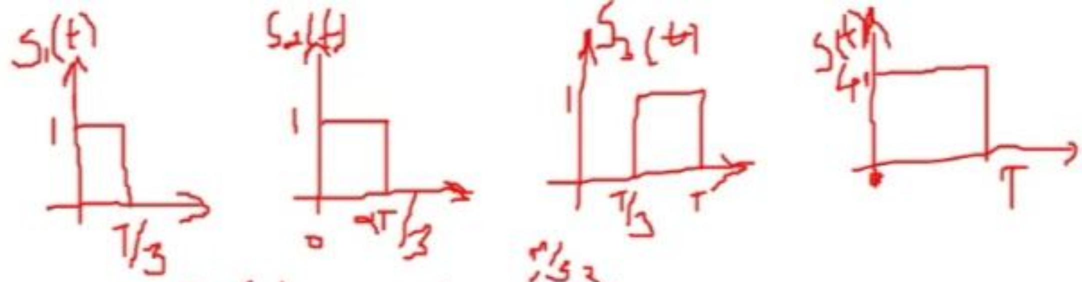
Among the philosophies for adaptive equalization of data transmission systems, we have *prechannel equalization* at the transmitter, and *postchannel equalization* at the receiver. Because the first approach requires a feedback channel, we consider only adaptive equalization at the receiving end of the system. This equalization can be achieved, prior to data transmission, by training the filter with the guidance of a suitable *training sequence* transmitted through the channel so as to adjust the filter parameters to optimum values. The typical telephone channel changes little during an average data call, so that precall equalization with a training sequence is sufficient in most cases encountered in practice. The equalizer is positioned after the receiving filter in the receiver.

The adaptive equalizer consists of a tapped-delay-line filter (with as many as 100 taps or more), whose coefficients are updated (starting from zero initial values) in accordance with the *least-mean square* (LMS) algorithm; the LMS algorithm ... o the filter coefficients ... data.‡ ... synchronously with the incoming where $y(nT)$ is the equalizer output and $b(nT)$ is the final (not necessarily) correct estimate of the transmitted symbol $b(nT)$. Now, in normal operation the decisions made by the receiver are correct with high probability. This means that the error estimates are correct most of the time, thereby permitting the adaptive equalizer to operate satisfactorily. Furthermore, an adaptive equalizer operating in a decision-directed mode is able to *track* relatively slow variations in channel characteristics.

The methods of implementing adaptive equalizers may be divided into three broad categories: *analog*, *hardwired digital*, and *programmable digital*, as described here:

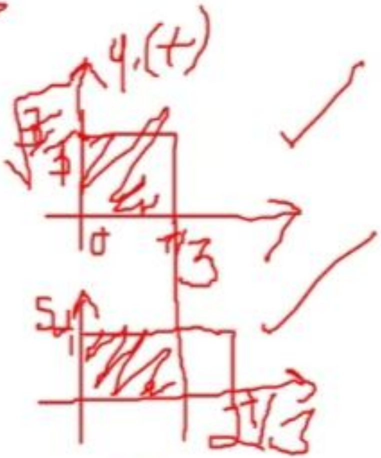
1. The analog approach is primarily based on the use of *charge-coupled device* (CCD) technology. The basic circuit realization of the CCD is a row of *field-effect transistors* (FET) with drains and sources connected in series, and the drains capacitively coupled to the gates. The set of adjustable tap weights are stored in digital memory locations, and the multiplications of the analog sample values by the digitized tap weights take place in analog fashion. This approach has significant potential in applications where the symbol rate is too high for digital implementation.
2. In hardwired digital implementation of an adaptive equalizer, the equalizer input is first sampled and then quantized into a form suitable for storage in shift registers. The set of adjustable tap weights are also stored in shift registers. Logic circuits are used to perform the required digital arithmetic

* Pseudo-noise (PN) sequences are discussed in Chapters 8 and 9.




$$\varphi_1(t) = \frac{s_1(t)}{\sqrt{E_1}} \qquad E_1 = \int_0^{T/3} 1^2\,dt = T/3$$

$$\varphi_1(t) = \begin{cases} \sqrt{3/T}, & 0 < t < T/3 \\ 0, & \end{cases}$$




$$g_2(t) = s_2(t) - \sum_{i=1}^{1} s_{2i}\,\varphi_i(t)$$

$$s_{21} = \int_0^T s_2(t)\,\varphi_1(t) = \int_0^{T/3} 1\cdot\sqrt{3/T} = \sqrt{T/3}$$

$N=3$ $\phi_1(t)$ $\phi_2(t)$ $\phi_3(t)$

$M=4$

$$s_1(t) = s_{11}\phi_1(t) = \sqrt{T/3}\,\phi_1(t)$$

$$s_2(t) = s_{21}\phi_1(t) + s_{22}\phi_2(t) = \sqrt{T/3}\,\phi_1(t) + \sqrt{T/3}\,\phi_2(t)$$

$$s_3(t) = s_{31}\phi_1(t) + s_{32}\phi_2(t) + s_{33}\phi_3(t) = 0 + \sqrt{T/3}\,\phi_2(t) + \sqrt{T/3}\,\phi_3$$

$$s_4(t) = s_1(t) + s_3(t)$$

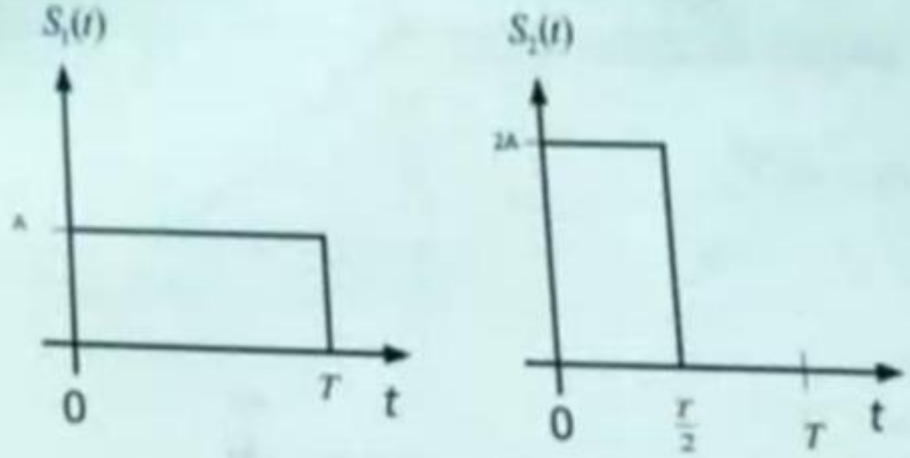
$$s_1(t) = (\sqrt{T/3}, 0, 0)$$

$$s_2(t) = (\sqrt{T/3}, \sqrt{T/3}, 0)$$

$$s_3(t) = (0, \sqrt{T/3}, \sqrt{T/3})$$

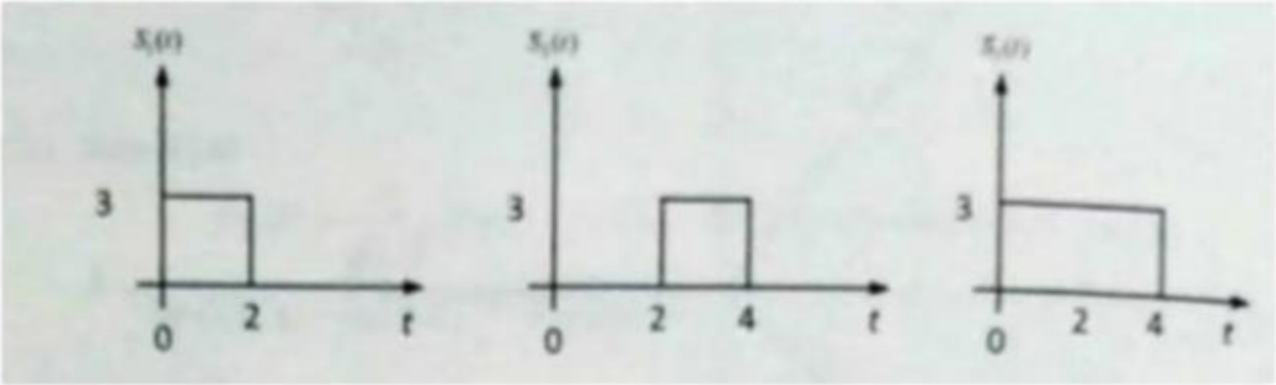
$$s_4(t) = (\sqrt{T/3}, \quad )$$

Two functions $S_1(t)\&S_2(t)$ are given in the Figure.The interval is $0 \le t \le T$ seconds. Using Gram-Schmidt Procedure,express these functions in terms of orthonormal functins.Also Sketch $\phi_1(t)$ and $\phi_2(t)$




Solution:

3 signals $S_1(t)$, $S_2(t)$, $S_3(t)$ are as shown in the figure. Apply GSO to obtain orthonormal basis functions for signals. Express the signals in terms of set of basis functions




Holla