

### Chapter I Problems Solutions

1. • Determine the range of permissible cutoff frequencies for the ideal lowpass filter used to reconstruct the signal

$$x(t) = 10 \cos(600\pi t) \cos^2(1600\pi t)$$

which is sampled at 4000 samples per second. Sketch  $X(\omega)$ ,  $X_s(\omega)$

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$$x(t) = 10 \cos(600\pi t) \cos^2(1600\pi t)$$

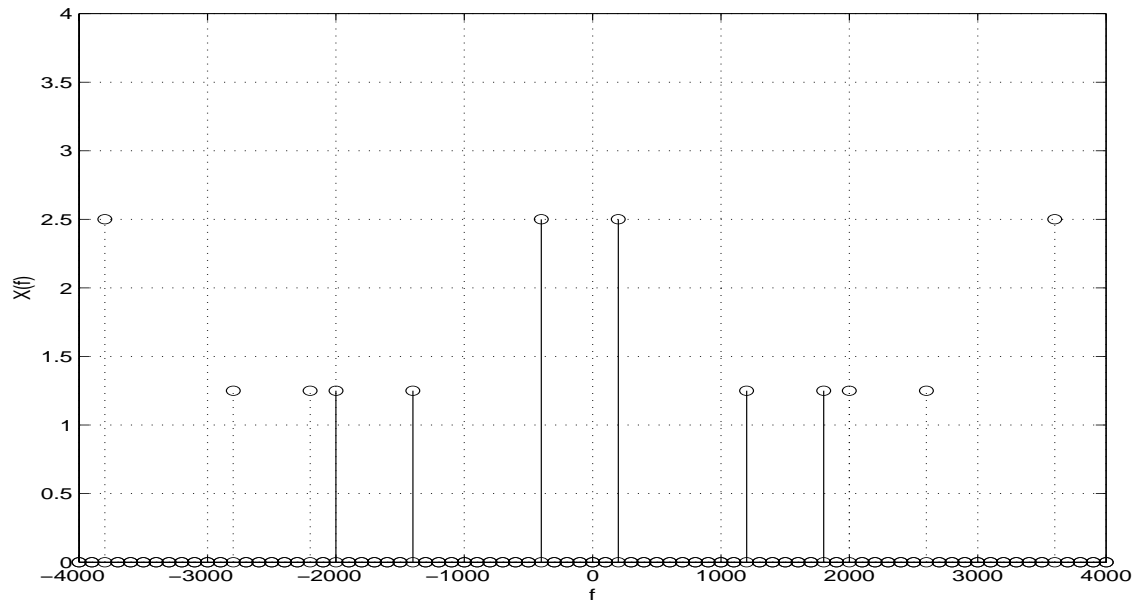
$$= 10 \cos(600\pi t) \left[ \frac{1}{2} + \frac{1}{2} \cos(3200\pi t) \right]$$

$$= 5 \cos(600\pi t) + 2.5 \cos(3800\pi t) + 2.5 \cos(2600\pi t)$$

The sampling frequency = 4000Hz, i.e spectrum repeats every 4000Hz.

The replica of the spectrum are shown dotted.

The range of the cutoff frequency of the ideal LPF is  $1900\text{Hz} < f_c < 2100\text{Hz}$



2. • Consider a signal  $f(t)$  having a probability density function

$$p(f) = \begin{cases} Ke^{-|f|} & -4 < f < 4 \\ 0 & \text{otherwise} \end{cases}$$

- a) Find  $K$ .  
 b) Determine the step size  $\Delta$  if there are four quantization levels.  
 c) Calculate the variance of the quantization error when there are four quantization levels. Do not assume that  $p(f)$  is constant over each level.
- a)  $\int_{-\infty}^{\infty} p(f)df = \int_{-4}^4 Ke^{-|f|}df = 2K \int_0^4 e^{-f}df = 2K(1 - e^{-4}) = 1$   
 $\implies K = \frac{1}{2(1-e^{-4})} \simeq 0.51$
- b) The step size  $\Delta = 8/4 = 2$ , (Quantization levels  $f_1 = -3, f_2 = -1, f_3 = 1, f_4 = 3$ )
- c) Total variance of the quantization error is equal to the sum of the variance of the quantization error at each step of quantization, i.e

$$\begin{aligned} E\{e^2\} &= \sum_{i=1}^4 \int_{f_i - \frac{\Delta}{2}}^{f_i + \frac{\Delta}{2}} (f - f_i)^2 p(f) df \\ &= 2K \int_0^2 (f - 1)^2 e^{-f} df + 2K \int_2^4 (f - 3)^2 e^{-f} df \\ &= 2K(1 - 5e^{-2}) + 2K(e^{-2} - 5e^{-4}) \simeq 0.374 \end{aligned}$$

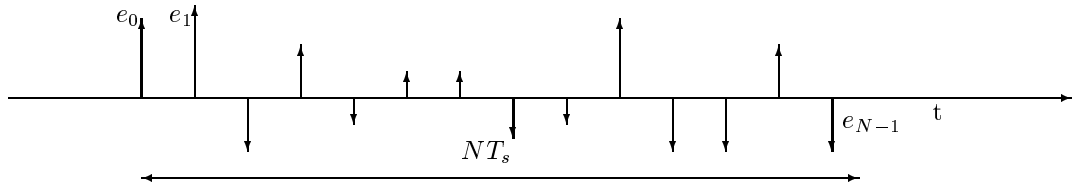


Figure 1:

3. • A signal  $f(t)$  is bandlimited and is ideally sampled at a sampling period  $T_s$  such that there is no aliasing error. Each of the ideal samples is then quantized to a step-size  $\Delta$ . The resulting signal can be written as

$$x(t) = \sum_{n=-\infty}^{n=\infty} f(nT_s)\delta(t - nT_s) + \sum_{n=-\infty}^{n=\infty} e_n\delta(t - nT_s)$$

where  $f(nT_s)$  are the original unquantized sample values and  $e_n$  is the quantization noise associated with the sample at  $nT_s$ . Assuming that  $\{e_n\}$  is a set of independent random variables, show that the power spectral density of the quantization noise is a constant (white noise) within the frequency range of  $-\frac{\pi}{T_s} \leq \omega \leq \frac{\pi}{T_s}$ . Also assume that there are many levels of quantization.

- The noise process is given by

$$e(t) = \sum_{n=-\infty}^{\infty} e_n\delta(t - nT_s)$$

where  $e_n$  is a random variable. We will approach the problem by considering a finite sequence instead of an infinite sequence, i.e. let

$$e_T(t) = \sum_{n=0}^{N-1} e_n\delta(t - nT_s)$$

where  $N$  is a finite number.

The finite length sequence is shown in figure (1). Now imagine a periodic sequence of impulses with period  $T$  such that each impulse is of strength  $e_0$ . This sequence is shown in figure (2). The periodic sequence is designated  $p_0(t)$  and can be expressed as a Fourier series

$$p_0(t) = e_0(t) \sum_{k=-\infty}^{\infty} \delta(t - kT) = e_0 \sum_{k=-\infty}^{\infty} \alpha_{0k} \exp(jk2\pi t/T)$$

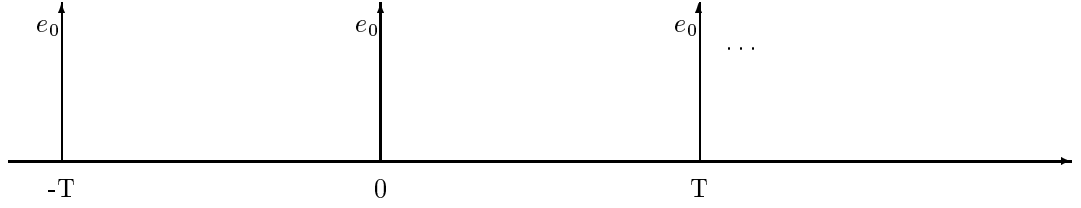


Figure 2:

$$= e_0 \sum_{k=-\infty}^{\infty} \alpha_{0k} \exp(jk\Delta\omega t)$$

where  $\Delta\omega = 2\pi/T$ . The power of the periodic sequence is given by

$$\begin{aligned} P_0 &= \frac{1}{T} \int_{-T/2}^{T/2} p_0^2(t) dt = \frac{1}{T} \int_{-T/2}^{T/2} [e_0 \sum_{k=-\infty}^{\infty} \alpha_{0k} \exp(jk\Delta\omega t)]^2 dt \\ &= e_0^2 \sum_{k=-\infty}^{\infty} |\alpha_{0k}|^2 \end{aligned}$$

But

$$\alpha_{0k} = \frac{1}{T} \int_{-T/2}^{T/2} \delta(t) \exp(-jk\Delta\omega t) dt = \frac{1}{T}, \forall k,$$

Thus  $P_0$  is a staircase function increasing by a step of  $\frac{e_0^2}{T^2}$  for every  $\Delta\omega$ . (Figure (3))

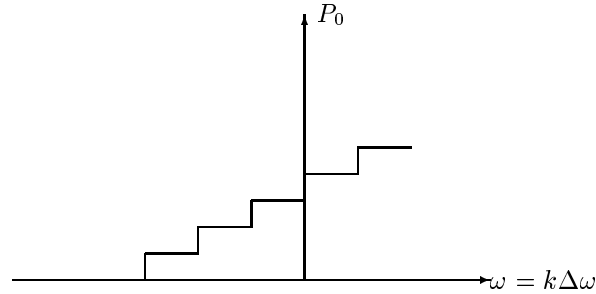


Figure 3:

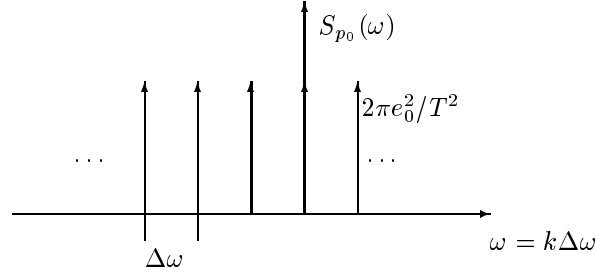


Figure 4:



Figure 5:

The power spectral density of the impulse sequence is given by (see Figure (4))

$$S_{p_0}(\omega) = 2\pi \frac{dP_0}{d\omega} = 2\pi \frac{e_0^2}{T^2} \sum_{k=-\infty}^{\infty} \delta(\omega - k\Delta\omega)$$

Now consider a similar sequence of impulses each of a strength  $e_1$  but occurring at  $t = kT + T_s$  (Figure (5)).

It can be similarly expressed as a Fourier series such that

$$p_1(t) = e_1 \sum_{k=-\infty}^{\infty} \delta(t - kT - T_s) = e_1 \sum_{k=-\infty}^{\infty} \alpha_{1k} \exp(jk\Delta\omega t - j\Delta\omega T_s)$$

Again  $\alpha_{1k} = 1/T$ . Now imagine a periodic sequence  $\tilde{e}_T(t)$  of period  $T$ , such that the impulses inside each period have strengths  $e_0, e_1, \dots, e_{N-1}$ . In other words, imagine the sequence in Figure (1) to repeat itself every  $T = NT_s$ , then this periodic sequence can be

written as

$$\begin{aligned}\tilde{e}_T(t) &= \sum_{n=0}^{N-1} p_n(t) = \sum_{n=0}^{N-1} e_n \sum_{k=-\infty}^{\infty} \alpha_{nk} \exp(jk\Delta\omega t - j\Delta\omega nT_s) \\ &= \sum_{n=0}^{N-1} \frac{e_n}{T} \exp(-j\Delta\omega nT_s) \sum_{k=-\infty}^{\infty} \exp(jk\Delta\omega t) \quad (1)\end{aligned}$$

But  $\tilde{e}_T(t)$  is a periodic function, therefore it can be expressed as a Fourier series, i.e.

$$\tilde{e}_T(t) = \sum_{k=-\infty}^{\infty} a_k \exp(jk\Delta\omega t) \quad (2)$$

Comparing (1) and (2), we see that

$$a_k = \sum_{n=0}^{N-1} \frac{e_n}{T} \exp(-j\Delta\omega nT_s) \quad (3)$$

The power spectral density of  $\tilde{e}_T(t)$  is

$$S_{\tilde{e}_T}(\omega) = 2\pi \sum_{k=-\infty}^{\infty} |a_k|^2 \delta(\omega - k\Delta\omega)$$

The power spectral density of  $\tilde{e}_T(t)$  associated with the interval is  $S_{\tilde{e}_T}(k\Delta\omega)$  such that the power falling within the frequency range  $(k\Delta\omega - \frac{\Delta\omega}{2})$  to  $(k\Delta\omega + \frac{\Delta\omega}{2})$  is  $|a_k|^2 = a_k a_k^*$ .

But  $a_k$  is a random variable since  $e_n$  is a random variable. Their relationship is given by (3). Hence we have to consider the average power falling within the frequency range  $(k\Delta\omega - \frac{\Delta\omega}{2})$  to  $(k\Delta\omega + \frac{\Delta\omega}{2})$  such that

$$\begin{aligned}\frac{1}{2\pi} \mathbb{E}\{S_{\tilde{e}_T}(k\Delta\omega)\Delta\omega\} &= \mathbb{E}\{a_k a_k^*\} \\ &= \mathbb{E}\left\{\left[\sum_{n=0}^{N-1} \frac{e_n}{T} \exp(-j\Delta\omega nT_s)\right] \left[\sum_{m=0}^{N-1} \frac{e_m}{T} \exp(j\Delta\omega mT_s)\right]\right\}\end{aligned}$$

But  $\mathbb{E}\{e_n e_m\} = 0$  for  $m \neq n$  since  $e_m, e_n$  are independent and zero mean,

$$\begin{aligned}\implies \frac{1}{2\pi} \mathbb{E}\{S_{\tilde{e}_T}(k\Delta\omega)\}\Delta\omega &= \mathbb{E}\left\{\frac{1}{T^2} \sum_{n=0}^{N-1} e_n^2\right\} = \frac{1}{T^2} \sum_{n=0}^{N-1} \mathbb{E}\{e_n^2\} \\ &= \frac{1}{T^2} \sum_{n=0}^{N-1} \left(\frac{\Delta^2}{12}\right) = \frac{1}{T^2} N \frac{\Delta^2}{12}\end{aligned}$$

Now,  $T = NT_s$ , and  $\Delta\omega = 2\pi/T$

$$\implies \mathbb{E}\{S_{\tilde{e}_T}(k\Delta\omega)\} = \frac{1}{T_s} \frac{\Delta^2}{12}$$

Finally, let  $T \rightarrow \infty$ , such that  $k\Delta\omega \rightarrow \omega$  and we have

$$S_e(\omega) = \lim_{T \rightarrow \infty} S_{\tilde{e}_T}(\omega) = \frac{1}{T_s} \frac{\Delta^2}{12}$$

i.e, the noise is white.

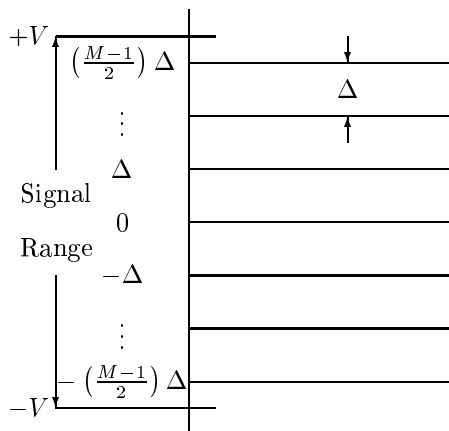


Figure 6:

4. • The signal in Problem I.3 has probability density uniformly distributed between  $\pm V$ . It is quantized into  $M$  discrete values, i.e.

$$-\left(\frac{M-1}{2}\right)\Delta, -\left(\frac{M-1}{2}-1\right)\Delta, \dots, 0, \Delta, 2\Delta, \dots, \left(\frac{M-1}{2}\right)\Delta$$

Find the signal to quantization noise ratio.

- The ideal sampled signal (without quantization) is given by

$$f_s(t) = \sum_{n=-\infty}^{\infty} f(nT_s)\delta(t - nT_s)$$

We are given that  $f(nT_s)$  is a random variable evenly distributed between  $+V$  and  $-V$  (Figure 6). This sampled sequence is very similar to the noise sequence considered in Problem I.3, except that  $f(nT_s)$  has a much larger range than  $e_n$ . More precisely, the range for  $f(nT_s)$  is from  $-M\Delta/2$  to  $+M\Delta/2$  while the range for  $e_n$  is from  $-\Delta/2$  to  $+\Delta/2$ .

Hence we can draw the conclusion that the power spectral density for this sampled signal is given by

$$S_{f_s}(\omega) = \frac{1}{T_s} \frac{(M\Delta)^2}{12}$$

To recover the signal we pass this sequence through a lowpass filter of bandwidth  $\omega_c = \omega_s/2 = \pi/T_s$  (Nyquist rate is assumed) so that



the signal power at the output of the filter is

$$\frac{1}{2\pi} \int_{-\pi/T_s}^{\pi/T_s} S_{f_s}(\omega) d\omega = \frac{1}{T_s^2} \frac{M^2 \Delta^2}{12}$$

The noise power at the output of the filter is

$$\frac{1}{2\pi} \int_{-\pi/T_s}^{\pi/T_s} S_e(\omega) d\omega = \frac{1}{T_s^2} \frac{\Delta^2}{12}$$

$$\implies \frac{S}{N_q}|_{output} = M^2 = 2^{2N}$$

where N is the number of bits to represent the M levels of quantization.

5. • A signal  $f(t)$  is not strictly bandlimited. We bandlimit  $f(t)$  and then sample it. Due to bandlimiting, distortion occurs even without quantization.

a) Show that the noise power due to the bandlimiting distortion is given by

$$N_D = \frac{1}{\pi} \int_{\omega_m}^{\infty} S_f(\omega) d\omega$$

where  $S_f(\omega)$  is the power spectral density of  $f(t)$  and  $\omega_m$  is the cutoff frequency of the bandlimiting filter.

b) If  $S_f(\omega) = A_0 e^{-|\omega/\omega_0|}$ , find  $N_D$

c) If the bandlimited signal  $f(t)$  is sampled at the Nyquist rate and quantized to a step-size  $\Delta$ , find the total output signal-to-noise power, i.e. find  $S_0/(N_D + N_q)$  assuming that the power spectral density of the quantization noise is constant within  $-\pi/T_s \leq \omega \leq \pi/T_s$ . (See problem I.3)

- a) Let the noise power due to the distortion of filtering be  $N_D$ , then

$$N_D = \frac{1}{2\pi} \int_{-\infty}^{\infty} S_f(\omega) d\omega - \frac{1}{2\pi} \int_{-\omega_m}^{\omega_m} S_f(\omega) d\omega$$

$$\frac{1}{2\pi} \left[ \int_{-\infty}^{-\omega_m} S_f(\omega) d\omega + \int_{\omega_m}^{\infty} S_f(\omega) d\omega \right] = \frac{1}{2\pi} \int_{\omega_m}^{\infty} S_f(\omega) d\omega$$

b) Given that  $S_f(\omega) = A_0 e^{-|\omega/\omega_0|}$ ,

$$\Rightarrow N_D = \frac{1}{\pi} \int_{\omega_m}^{\infty} A_0 e^{-\omega/\omega_0} d\omega = \frac{1}{\pi} A_0 \omega_0 e^{-\omega_m/\omega_0}$$

c) Let the signal which has been bandlimited to  $\omega_m$  be  $f_B(t)$ . This bandlimited signal already consists of a distortion noise  $n_D(t)$  the power of which is  $N_D$ . Thus we can write

$$f_B(t) = f(t) + n_D(t)$$

The power spectral density of this bandlimited signal is

$$S_{f_B}(\omega) = \begin{cases} A_0 e^{-|\omega/\omega_0|} & \text{for } -\omega_m \leq \omega \leq \omega_m \\ 0 & \text{elsewhere} \end{cases}$$

The bandlimited signal  $f_B(t)$  is then sampled at the Nyquist rate, i.e.  $\omega_s = 2\pi/T_s = 2\omega_m$ . Because of sampling, the power spectral density repeats at every  $\omega_s$ , and scaled by  $\frac{1}{T_s^2}$ . Figure (7) illustrates this fact. Thus,

$$S_{f_{B_s}}(\omega) = \sum_{n=-\infty}^{\infty} \frac{1}{T_s^2} S_{f_B}(\omega - n\omega_s)$$

where  $f_{B_s}(t)$  is the bandlimited signal sampled at  $\omega_s$ .

Suppose we do not quantize this sampled bandlimited signal. To

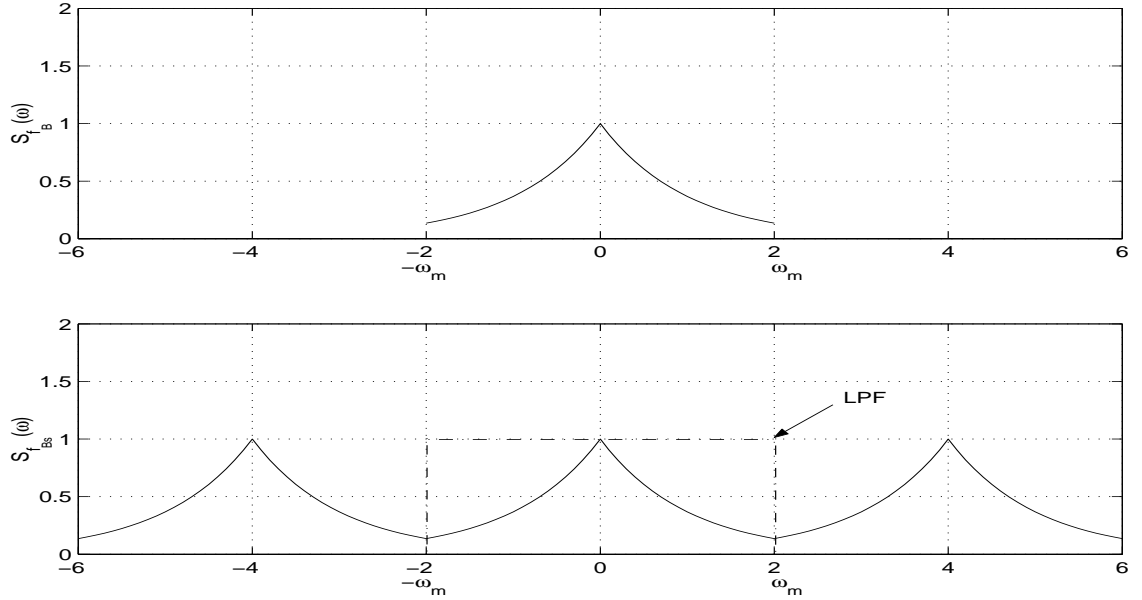


Figure 7:

recover this signal we use a LPF having cutoff frequency at  $\omega_m$ . Hence the output signal power is

$$S_0 = \frac{1}{2\pi} \int_{-\omega_m}^{\omega_m} \frac{A_0}{T_s^2} e^{-|\omega/\omega_0|} d\omega = \frac{A_0}{\pi T_s^2} \omega_0 [1 - e^{-\omega_m/\omega_0}]$$

Since the signal has been bandlimited before, hence this output signal power is the power of the distorted signal. We can never recover the true signal at the output even if there were no quantization because part of the signal has been filtered off. Hence, we have to regard  $S_0$  as the signal power at the output. From Problem I.3, quantization

noise at the output of the filter is

$$N_q = \frac{1}{2\pi} \int_{-\pi/T_s}^{\pi/T_s} \frac{1}{T_s} \frac{\Delta^2}{12} d\omega = \frac{1}{T_s^2} \frac{\Delta^2}{12}$$

Hence output signal-to-quantization noise ration is

$$\frac{S_0}{N_q} = \frac{12A_0}{\pi\Delta^2} \omega_0 [1 - e^{-\omega_m/\omega_0}]$$

Note:  $N_D$  has already been taken into account since we chose  $S_0$  to be the output signal power.

6. • A compressor has the characteristic  $f_0 = C(f)$  where  $f$  is the input signal and  $f_0$  is the compressed signal. Thus, if no compression is employed,  $f_0 = f$ .
- a) Show that as a result of compression, a uniform step size of  $\Delta$  volts in the output  $f_0$  results in nonuniform quantization, i.e., varying step size, of the input  $f$ . Do this by dividing  $f_0$  into 8 equal quantization steps.
- b) Show that the variance of the quantization error is now,

$$\begin{aligned} E\{e_c^2\} &= \int_{f_1 - \frac{\Delta_1}{2}}^{f_1 + \frac{\Delta_1}{2}} (f - f_1)^2 p(f) df + \int_{f_2 - \frac{\Delta_2}{2}}^{f_2 + \frac{\Delta_2}{2}} (f - f_2)^2 p(f) df \\ &\quad + \cdots + \int_{f_8 - \frac{\Delta_8}{2}}^{f_8 + \frac{\Delta_8}{2}} (f - f_8)^2 p(f) df \end{aligned}$$

where  $f_1 - \frac{\Delta_1}{2} = f_{min}$  and  $f_8 + \frac{\Delta_8}{2} = f_{max}$ ,  $f_{i+1} - \frac{\Delta_{i+1}}{2} = f_i + \frac{\Delta_i}{2}$ .

- c) If there is a large number of quantization levels, show that

$$\Delta_i = \frac{\Delta}{C'(f_i)}$$

where  $C'(f_i) = \left. \frac{dC(f)}{df} \right|_{f=f_i}$

Hint: Note  $\frac{\Delta f_0}{\Delta f_i} \simeq C'(f)$

- d) If  $p(f)$  is approximately constant throughout each step, show that  $E\{e_c^2\}$  becomes

$$E\{e_c^2\} \simeq \frac{1}{12} [\Delta_1^3 p(f_1) + \Delta_2^3 p(f_2) + \cdots + \Delta_8^3 p(f_8)]$$

- e) Using the result of c), show that if there are many quantization levels, that is ,

$$E\{e_c^2\} = \frac{\Delta^2}{12} \sum_i \frac{\Delta_i p(f_i)}{[C'(f_i)]^2} \simeq \frac{\Delta^2}{12} \int_{f_{min}}^{f_{max}} \frac{p(f)}{[C'(f)]^2} df$$

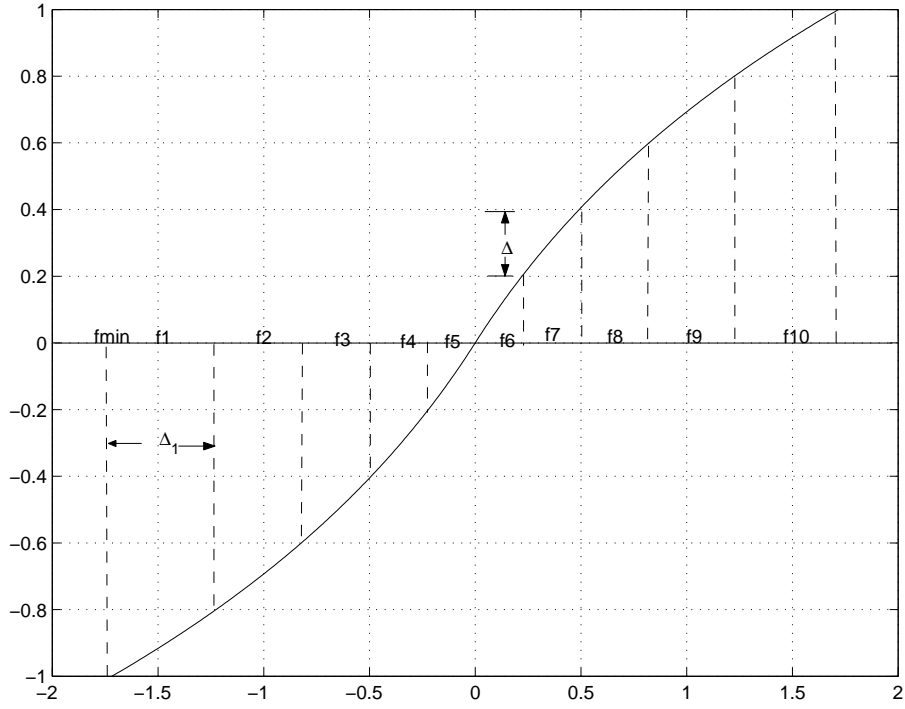


Figure 8:

- b) From figure(8),

$$E\{e_c^2\} = \sum_{k=1}^8 \int_{f_k - \frac{\Delta_k}{2}}^{f_k + \frac{\Delta_k}{2}} (f - f_k)^2 p(f) df$$

- c) If there is a large number of quantization levels, then  $\Delta$  is small,

$$\Rightarrow \frac{\Delta}{\Delta_k} = \frac{\Delta f_o}{\Delta f_{in}} \simeq \frac{df_o}{df_{in}} \Big|_{f_{in}=f_k} = C'(f_{in}) \Big|_{f_{in}=f_k}$$

$$\Rightarrow \Delta_k = \frac{\Delta}{C'(f_k)}$$

- d)

$$E\{e_c^2\} = \sum_{k=1}^8 \int_{f_k - \frac{\Delta_k}{2}}^{f_k + \frac{\Delta_k}{2}} (f - f_k)^2 p(f) df \quad \forall k = 1, \dots, 8$$

But,  $p(f) \simeq p(f_k)$  within the interval  $f_k - \frac{\Delta_k}{2} \leq f \leq f_k + \frac{\Delta_k}{2}$

$$\begin{aligned} \implies \mathbb{E}[e_c^2] &\simeq \sum_{k=1}^8 \int_{f_k - \frac{\Delta_k}{2}}^{f_k + \frac{\Delta_k}{2}} (f - f_k)^2 p(f_k) df \\ &= \sum_{k=1}^8 p(f_k) \int_{f_k - \frac{\Delta_k}{2}}^{f_k + \frac{\Delta_k}{2}} (f - f_k)^2 df = \sum_{k=1}^8 \frac{\Delta_k^3}{12} p(f_k) \end{aligned}$$

e)

$$\begin{aligned} \mathbb{E}[e_c^2] &\simeq \sum_{k=1}^8 \frac{\Delta_k^3}{12} p(f_k) \simeq \sum_{k=1}^8 \frac{\Delta_k}{12} \left[ \frac{\Delta}{C'(f_k)} \right]^2 p(f_k) \\ &= \frac{\Delta^2}{12} \sum_k \frac{\Delta_k p(f_k)}{\{C'(f_k)\}^2} \simeq \frac{\Delta^2}{12} \int_{f_{min}}^{f_{max}} \frac{p(f)}{\{C'(f)\}^2} df \end{aligned}$$

7. This problem is too tedious. You can just skip it.



8. • Assume logarithmic companding with  $C(f)$  given by

$$C(f) = f_{max} \operatorname{sgn}(f) \frac{\log(1 + \mu|f|/f_{max})}{\log(1 + \mu)}$$

where  $\mu$  is a constant known as the compression parameter. Find the mean square quantization error. Assume signal to be uniformly probable between  $\pm f_{max}$ .

•

$$C(f) = f_{max} \operatorname{sgn}(f) \frac{\log(1 + \mu|f|/f_{max})}{\log(1 + \mu)}$$

Assuming the signal has a pdf uniformly distributed between  $\pm f_{max}$  then

$$p(f) = \frac{1}{2f_{max}}$$

Using part e) of I.6,

$$\mathbb{E}[e_c^2] \simeq \frac{\Delta^2}{12} \int_{-f_{max}}^{f_{max}} \frac{1}{2f_{max}} \frac{1}{\{C'(f)\}^2} df = \frac{\Delta^2}{12} \int_0^{f_{max}} \frac{1}{\{C'(f)\}^2} df$$

$$f_{omax} = f_{max} \left[ \frac{\log(1 + \mu)}{\log(1 + \mu)} \right] = f_{max}$$

$$f_{omin} = -f_{omax}$$

$$\Delta = (f_{omax} - f_{omin})/M = \frac{2f_{max}}{M}$$

where M is the number of quantization levels.

$$C'(f) = \frac{\mu}{\log(1 + \mu)(1 + \mu f/f_{max})}$$

$$\begin{aligned} \Rightarrow \mathbb{E}[e_c^2] &\simeq \frac{\Delta^2}{12} \int_0^{f_{max}} \frac{1}{f_{max}} \frac{\log^2(1 + \mu)(1 + \mu f/f_{max})^2}{\mu^2} df \\ &= \frac{f_{max}^2}{3\mu^2} \log^2(1 + \mu) \left[ 1 + \mu + \frac{\mu^2}{3} \right] \end{aligned}$$

9. • Given an audio waveform given by

$$f(t) = 3 \sin(500t) + 4 \sin(1000t) + 4 \sin(1500t)$$

Find the signal to quantization noise ratio if this is coded using delta modulation.

- Signal is

$$f(t) = 3 \sin(500t) + 4 \sin(1000t) + 4 \sin(1500t)$$

Therefore signal power is

$$S = \frac{3^2 + 4^2 + 4^2}{2} = 20.5$$

In delta modulation, the sampling period is flexible, the smaller the sampling period, the smaller the quantization error. The maximum sampling period is at the Nyquist rate, i.e.  $T_{smax} = \pi/1500$ .

Now the slope of the signal is

$$\begin{aligned} \frac{df}{dt} &= -3 \times 500 \cos(500t) - 4 \times 1000 \cos(1000t) \\ &\quad - 4 \times 1500 \cos(1500t) \end{aligned}$$

The maximum slope is

$$\frac{df}{dt} = 3 \times 500 + 4 \times 1000 + 4 \times 1500 = 11500$$

To ensure no overloading, according to Eq(I.4.1), we have

$$\frac{\Delta}{T_s} \geq \left. \frac{df}{dt} \right|_{max}, \text{ i.e. } \Delta \geq 11500T_s$$

where  $\Delta$  is the quantization step size. Using Eq(I.4.5), the quantization noise power is

$$N_q = \frac{\Delta^2 \omega_m T_s}{6\pi} = \frac{(11500T_s)^2 (1500T_s)}{6\pi}$$

At Nyquist rate,

$$N_q = 11500 \times 11500 \times 1500 \times \frac{\pi^3}{1500^3} / 6\pi = 96.69$$

Therefore, at the Nyquist rate,

$$\frac{S}{N_q} = \frac{20.5}{96.69} = -6.74 \text{ dB}$$

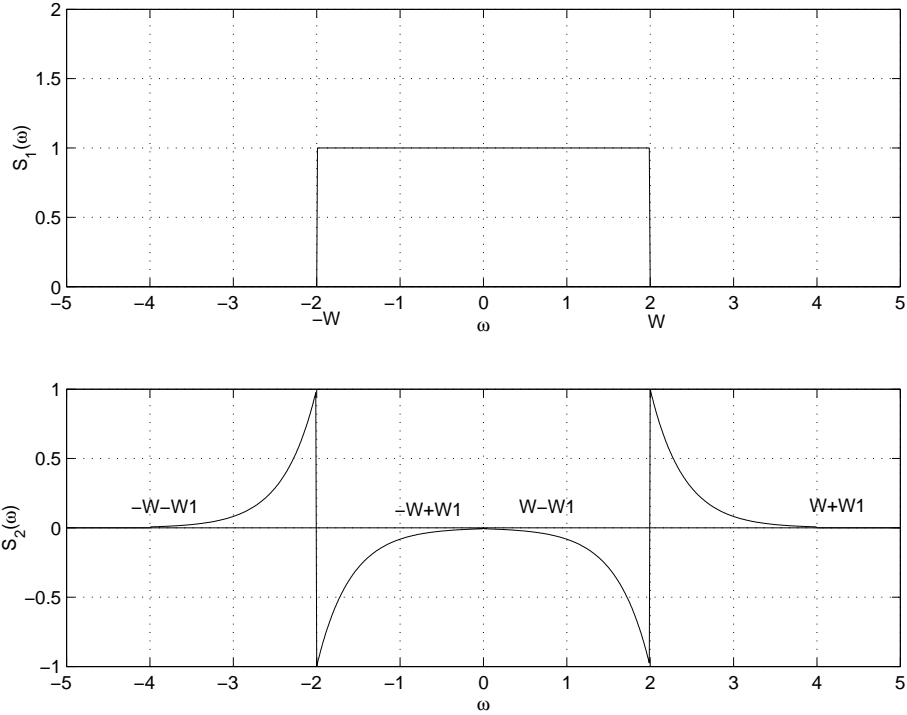
This value is unacceptable. Suppose, we increase the sampling rate 32 times, then  $T_s = \pi/(32 \times 1500)$ , therefore

$$N_q = 11500 \times 11500 \times 1500 \times \frac{\pi^3}{(1500 \times 32)^3} / 6\pi = 2.95 \times 10^{-3}$$

Therefore at this sampling rate

$$\frac{S}{N_q} = \frac{20.5}{2.95 \times 10^{-3}} = 38.42 \text{ dB}$$

which is acceptable.



10. • Prove Eq.(I.7.5)

$$s_1(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} S_1(\omega) e^{j\omega t} d\omega = \frac{1}{2\pi} \int_{-W}^W e^{j\omega t} d\omega = \frac{W}{\pi} \left[ \frac{\sin(Wt)}{Wt} \right]$$

$$s_2(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} S_2(\omega) e^{j\omega t} d\omega$$

But  $S_2(\omega)$  is an even function. Therefore,

$$s_2(t) = 2 \left[ \frac{1}{2\pi} \int_0^{\infty} S_2(\omega) \cos(\omega t) d\omega \right]$$

$$= \frac{1}{\pi} \int_0^W S_2(\omega) \cos(\omega t) d\omega + \frac{1}{\pi} \int_W^{2W} S_2(\omega) \cos(\omega t) d\omega$$

Let  $\omega = W - \lambda$  in the first integral  
and  $\omega = W + \lambda$  in the second integral, then

$$s_2(t) = \frac{1}{\pi} \int_0^W S_2(W - \lambda) \cos(W - \lambda)t d\lambda + \frac{1}{\pi} \int_0^W S_2(W + \lambda) \cos(W + \lambda)t d\lambda$$

$$= \frac{1}{\pi} \int_0^W [S_2(W - \lambda) \cos(W - \lambda)t + S_2(W + \lambda) \cos(W + \lambda)t] d\lambda$$

But  $S_2(\omega)$  has odd symmetry about  $W$ , i.e.  $S_2(W - \lambda) = -S_2(W + \lambda)$ .  
Therefore,

$$s_2(t) = \frac{1}{\pi} \int_0^W S_2(W + \lambda) [\cos(W + \lambda)t - \cos(W - \lambda)t] d\lambda$$

Now,

$$\cos(W + \lambda)t - \cos(W - \lambda)t = -2 \sin(Wt) \sin(\lambda t)$$

Therefore,

$$s_2(t) = \frac{1}{\pi} [-2 \sin(Wt) \int_0^W S_2(W + \lambda) \sin(\lambda t) d\lambda]$$

or,

$$s_2(t) = \frac{-2Wt}{\pi} \left[ \frac{\sin(Wt)}{Wt} \int_0^W S_2(W + \omega) \sin(\omega t) d\omega \right]$$

11. • A computer output is a train of binary symbols at 56Kbit/sec. Raised-cosine spectral shaping with  $W_1/W = 0.3$  is used prior to baseband transmission.
- Determine the minimum bandwidth required.
  - Repeat if two successive digits are combined into one pulse with four possible amplitudes.

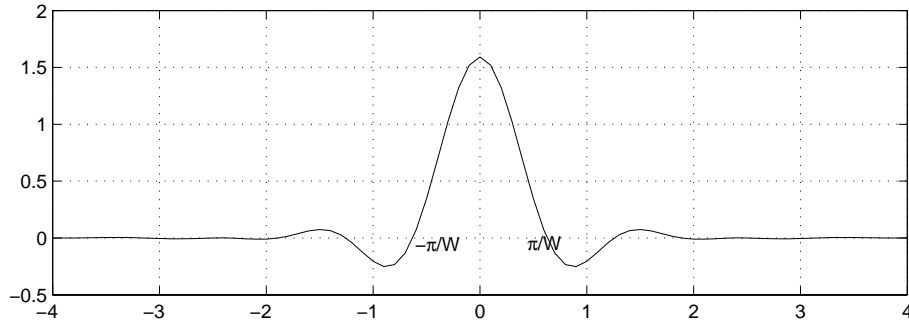
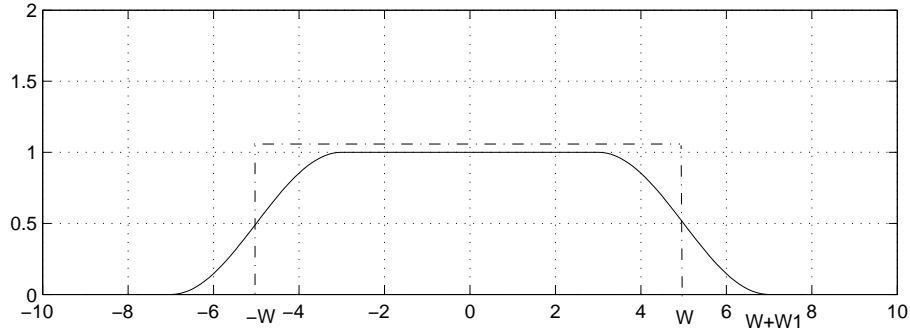


Figure 9:

- a) Bit Period  $T_b = \frac{1}{56 \times 10^3}$   
Therefore,

$$\frac{\pi}{W} = \frac{1}{56 \times 10^3}$$

Total bandwidth =  $W + 0.3W = 1.3W = 1.3 \times \pi \times 56 \times 10^3 = 228.7 \text{Krad/sec}$

- b) If two pulses are combined into one, then symbol rate is halved.  
Therefore,

$$T_s = 2 \times \frac{1}{56 \times 10^3}$$

and

$$W = \frac{\pi \times 56 \times 10^3}{2}$$

The total bandwidth =  $(1.3)W = 114.35 \text{Krad/sec}$

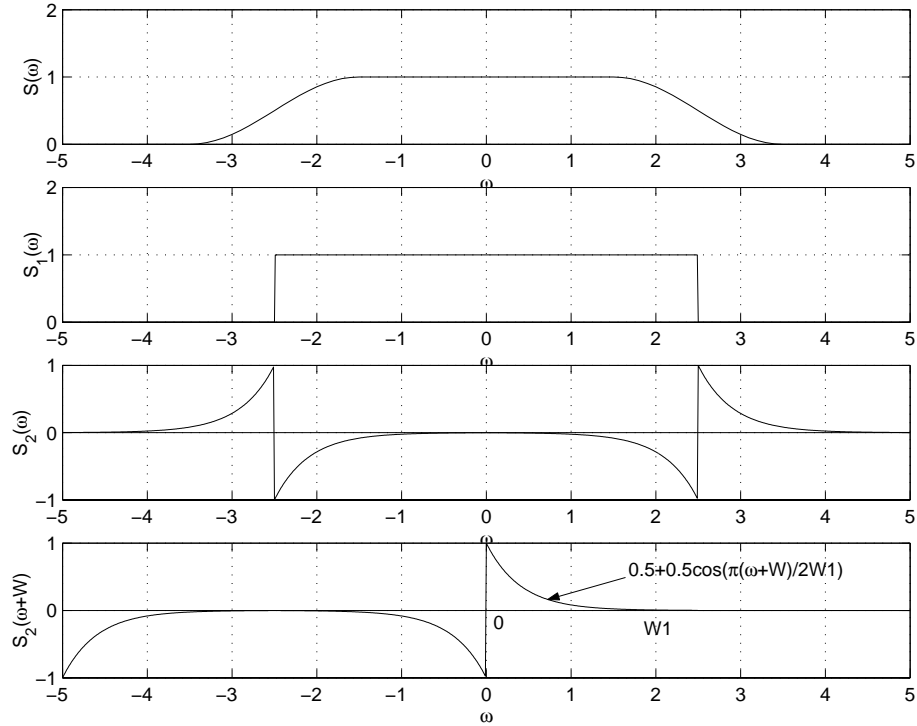


Figure 10:

12. • Consider the raised-cosine spectrum. Assume linear phase shift, i.e.

$$\theta(\omega) = -\omega t_0$$

Show that the impulse response signal is given by

$$s(t) = \frac{W}{\pi} \frac{\sin Wt}{Wt} \left[ \frac{\cos W_1 t}{1 - (2W_1 t/\pi)^2} \right]$$

- From Eq.(I.7.6)

$$s(t) = \frac{W}{\pi} \frac{\sin Wt}{Wt} \left[ 1 - 2t \int_0^{W_1} S_2(\omega + W) \sin \omega t d\omega \right]$$

But over the integration interval,

$$S_2(\omega + W) = \frac{1}{2} + \frac{1}{2} \cos\left[\frac{\pi}{2W_1}(\omega + W_1)\right]$$

Therefore,

$$s(t) = \frac{W}{\pi} \frac{\sin Wt}{Wt} \left[ 1 - 2t \int_0^{W_1} \frac{1}{2} \sin \omega t + \frac{1}{2} \sin \omega t \cos\left[\frac{\pi}{2W_1}(\omega + W_1)\right] d\omega \right]$$



$$\begin{aligned}
&= \frac{W \sin Wt}{\pi Wt} \left[ 1 + t \frac{\cos \omega t}{t} \Big|_0^{W_1} - \frac{t}{2} \int_0^{W_1} \left[ \sin\left(\omega\left(t + \frac{\pi}{2W_1}\right) + \frac{\pi}{2}\right) + \sin\left(\omega\left(t - \frac{\pi}{2W_1}\right) - \frac{\pi}{2}\right) \right] d\omega \right] \\
&= \frac{W \sin Wt}{\pi Wt} \left[ \cos \omega t + \frac{t \cos\left(\omega\left(t + \frac{\pi}{2W_1}\right) + \frac{\pi}{2}\right)}{2\left(t + \frac{\pi}{2W_1}\right)} \Big|_0^{W_1} + \frac{t \cos\left(\omega\left(t - \frac{\pi}{2W_1}\right) - \frac{\pi}{2}\right)}{2\left(t - \frac{\pi}{2W_1}\right)} \Big|_0^{W_1} \right] \\
&= \frac{W \sin Wt}{\pi Wt} \left[ \cos W_1 t - \frac{1}{2} \frac{\cos W_1 t}{\left(1 + \frac{\pi}{2W_1 t}\right)} - \frac{1}{2} \frac{\cos W_1 t}{\left(1 - \frac{\pi}{2W_1 t}\right)} \right] \\
&= \frac{W \sin Wt}{\pi Wt} \cos W_1 t \left[ \frac{1 - \left(\frac{\pi}{2W_1 t}\right)^2 - \frac{1}{2}\left(1 - \frac{\pi}{2W_1 t}\right) - \frac{1}{2}\left(1 + \frac{\pi}{2W_1 t}\right)}{1 - \left(\frac{\pi}{2W_1 t}\right)^2} \right] \\
&= \frac{W \sin Wt}{\pi Wt} \frac{\cos W_1 t}{1 - \left(\frac{2W_1 t}{\pi}\right)^2}
\end{aligned}$$

13. • Consider a sequence of pulse samples  $x(kT_s)$  which is assumed to have finite energy. The correlation matrix of this sequence is defined as

$$\Phi_{xx} = E[\mathbf{x}_k \mathbf{x}_k^t]$$

where  $\mathbf{x}_k^t$  is the transpose of  $\mathbf{x}_k$ , and

$$\mathbf{x}_k = \begin{bmatrix} x(kT_s) \\ x(kT_s - T_s) \\ \vdots \\ x(kT_s - NT_s + T_s) \end{bmatrix}$$

Show that the matrix  $\Phi_{xx}$  is positive semi-definite, i.e.  $\mathbf{w}^t \Phi_{xx} \mathbf{w} \geq 0$  for any non-zero vector  $\mathbf{w}$ .

•

$$\Phi_{xx} = E[\mathbf{x}_k \mathbf{x}_k^t]$$

Therefore

$$\mathbf{w}^t \Phi_{xx} \mathbf{w} = E[\mathbf{w}^t \mathbf{x}_k \mathbf{x}_k^t \mathbf{w}] = E[(\mathbf{w}^t \mathbf{x}_k)(\mathbf{x}_k^t \mathbf{w})]$$

Now,

$$\mathbf{w}^t \mathbf{x}_k = \sum_{n=0}^{N-1} w_n x(kT_s - nT_s) = y(kT_s), \quad (\text{which is the } k\text{th sample of the output})$$

Also,

$$\mathbf{x}_k^t \mathbf{w} = \sum_{n=0}^{N-1} x(kT_s - nT_s) w_n = y(kT_s)$$

Therefore,

$$\mathbf{w}^t \Phi_{xx} \mathbf{w} = E[y^2(kT_s)] \geq 0$$

That is  $\Phi_{xx}$  is positive semi-definite.

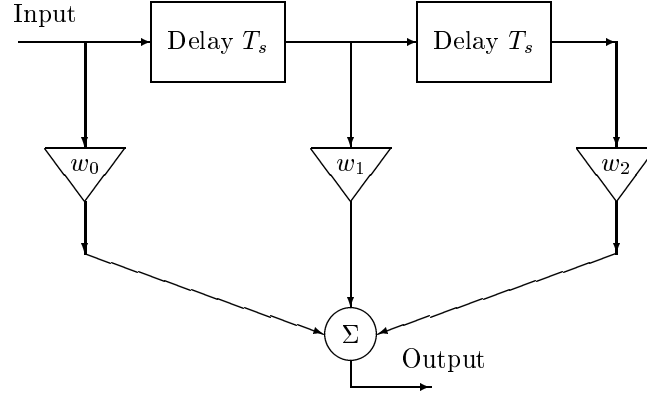


Figure 11:

14. • Some radio systems suffer from multipath distortion which is caused by the existence of more than one propagation path between the transmitter and the receiver. Consider a channel the output of which, in response to a signal  $s(t)$ , is defined by

$$x(t) = K_1s(t - t_1) + K_2s(t - t_2)$$

where  $K_1$  and  $K_2$  are constants,  $t_1$  and  $t_2$  represent transmission delays. It is proposed to use the 3-tap delay line filter (Fig. 11) to equalize the multipath distortion produced by this channel.

a) Evaluate the transfer function of the channel.

b) Evaluate  $w_0, w_1$  and  $w_2$  in terms of  $K_1, K_2, t_1$  and  $t_2$ , assuming that  $K_2 \ll K_1$  and  $t_2 > t_1$ .

- a) The channel output is

$$x(t) = K_1s(t - t_1) + K_2s(t - t_2)$$

Taking the Fourier transform, we have

$$X(\omega) = K_1S(\omega)e^{-j\omega t_1} + K_2S(\omega)e^{-j\omega t_2}$$

Hence the transfer function of the channel is

$$H_c(\omega) = \frac{X(\omega)}{S(\omega)} = K_1e^{-j\omega t_1} + K_2e^{-j\omega t_2}$$

b) Ideally, the equalizer should be designed so that

$$H_c(\omega)H_e(\omega) = K_0e^{-j\omega t_0}$$

where  $K_0$  and  $t_0$  are constants. Now for the tap-delay line equalizer, the transfer function is

$$H_e(\omega) = w_0 + w_1e^{-j\omega T_s} + w_2e^{-j2\omega T_s}$$

$$= w_0 \left[ 1 + \frac{w_1}{w_0} e^{-j\omega T_s} + \frac{w_2}{w_0} e^{-j2\omega T_s} \right]$$

For the ideal equalizer,

$$\begin{aligned} H_{e_i}(\omega) &= \frac{K_0 e^{-j\omega t_0}}{H_c(\omega)} \\ &= \frac{K_0 e^{-j\omega t_0}}{K_1 e^{-j\omega t_1} + K_2 e^{-j\omega t_2}} \\ &= \frac{(K_0/K_1) e^{-j\omega(t_0-t_1)}}{1 + (K_2/K_1) e^{-j\omega(t_2-t_1)}} \end{aligned}$$

Using the binomial expansion with  $\frac{K_2}{K_1} < 1$ ,

$$H_{e_i}(\omega) \simeq (K_0/K_1) e^{-j\omega(t_0-t_1)} \left[ 1 - \frac{K_2}{K_1} e^{-j\omega(t_2-t_1)} + \left( \frac{K_2}{K_1} \right)^2 e^{-j2\omega(t_2-t_1)} + \dots \right]$$

With a 3-tap transversal delay line equalizer, we equate  $H_e(\omega)$  to  $H_{e_i}(\omega)$ . Therefore,

$$\frac{K_0}{K_1} = w_0$$

$$t_0 - t_1 = 0$$

$$-\frac{K_2}{K_1} = \frac{w_1}{w_0}$$

$$\left( \frac{K_2}{K_1} \right)^2 = \frac{w_2}{w_0}$$

$$T_s = t_2 - t_1$$

$K_0$  is an arbitrary constant. Choosing  $K_0 = K_1$ , we find that the tap weights are,

$$w_0 = 1, \quad w_1 = -\frac{K_2}{K_1}, \quad w_2 = \left( \frac{K_2}{K_1} \right)^2$$

15. • In order to prevent detection error from propagating in the duobinary signalling scheme, we employ the precoding method shown in figure (12).

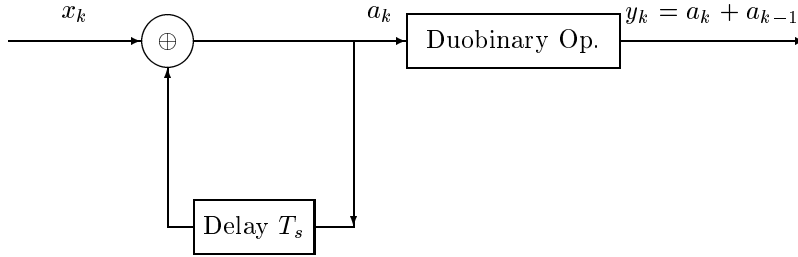


Figure 12:

We first form the sequence

$$a_k = x_k \oplus a_{k-1}$$

where  $\oplus$  represents the modulo-2 sum. Then we obtain the duobinary sequence  $y_k$  such that

$$y_k = a_k + a_{k-1} = (x_k \oplus a_{k-1}) + a_{k-1}$$

Since  $a_k = 0$  or  $1$ ,  $y_k = 0, 1$  or  $2$ .

a) Find the values of  $x_k$  when  $y_k = 0, 1$  or  $2$ . Hence obtain a decoding rule at the receiver for  $\hat{x}_k$ .

b) If the sequence  $x_k$  is  $0\ 0\ 1\ 1\ 0\ 1\ 0$ , find the corresponding sequences  $a_k$ ,  $y_k$ , and  $\hat{x}_k$ .

c) Now, assuming an error is made in one of the values of the received sequence  $y_k$ , verify that the sequence  $\hat{x}_k$  has only one error in the corresponding position and that the error does not propagate.

- Let  $x_k$  be either  $0$  or  $1$ . Then,

$$a_k = x_k \oplus a_{k-1}$$

means that  $a_k = 0$  or  $1$ . Hence

$$y_k = \begin{cases} 0 & \text{if } a_k = 0, a_{k-1} = 0 \\ 1 & \text{if } a_k = 1, a_{k-1} = 0 \text{ or } a_k = 0, a_{k-1} = 1 \\ 2 & \text{if } a_k = 1, a_{k-1} = 1 \end{cases}$$

a)

**Case I ( $y_k = 2$ )** Since

$$y_k = a_k + a_{k-1} = (a_{k-1} \oplus x_k) + a_{k-1}$$

$$y_k = 2 \implies a_{k-1} = 1 \implies a_{k-1} \oplus x_k = 1 \implies x_k = 0$$

**Case II ( $y_k = 1$ )** In this case,  $a_{k-1} = 0$  and  $a_k = 0$

$$\implies x_k = 0$$

**Case III ( $y_k = 0$ )** In this case either,

$$(a) a_{k-1} = 1, \implies a_k = 0 \implies x_k = 1$$

$$(b) a_{k-1} = 0, \implies a_k = 1 \implies x_k = 1$$

Hence for

$$y_k = \begin{cases} 0 \\ 2 \end{cases} \implies x_k = 0$$

and for

$$y_k = 1 \implies x_k = 1$$

Thus to find  $\hat{x}_k$ , we put

$$\hat{x}_k = y_k \text{ mod } -2 \quad \text{i.e., } \hat{x}_k = y_k \text{ in binary without carry}$$

b-c)

$x_k$		0	0	1	1	0	1	0
	<i>assumed</i>							
	$\underbrace{\hspace{1em}}_1$	1	1	0	1	1	0	0
	$y_k$	2	2	1	1	2	1	0
$\hat{x}_k = y_k \text{ mod } -2$		0	0	1	1	0	1	0
				<i>error</i>				
	$y'_k$	2	$\underbrace{1}_{\text{error}}$	1	1	2	1	0
			<i>error</i>					
$\hat{x}'_k = y'_k \text{ mod } -2$		0	$\underbrace{1}_{\text{error}}$	1	1	0	1	0

Notice that this precoding scheme resulted in no error propagation. That is error was just limited to one single bit.

16. • Show, by using the Shifting Theorem, that Eq.(I.9.6) in the text book represents the transfer function of the pulse addition network in the duobinary scheme.

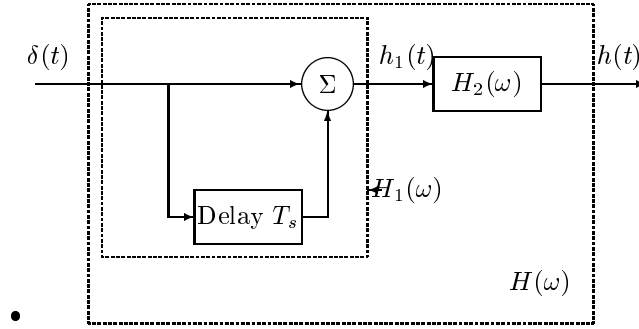


Figure 13:

$$h_1(t) = \delta(t) + \delta(t - T_s)$$

Therefore,

$$H_1(\omega) = 1 + e^{-j\omega T_s}$$

And,

$$H_2(\omega) = \begin{cases} T_s & -\frac{\pi}{T_s} \leq \omega \leq \frac{\pi}{T_s} \\ 0 & |\omega| > \frac{\pi}{T_s} \end{cases}$$

Therefore,

$$H(\omega) = H_1(\omega)H_2(\omega) = \begin{cases} (1 + e^{-j\omega T_s})T_s = 2T_s \cos \frac{\omega T_s}{2} e^{-j\omega T_s/2} & -\frac{\pi}{T_s} \leq \omega \leq \frac{\pi}{T_s} \\ 0 & |\omega| > \frac{\pi}{T_s} \end{cases}$$

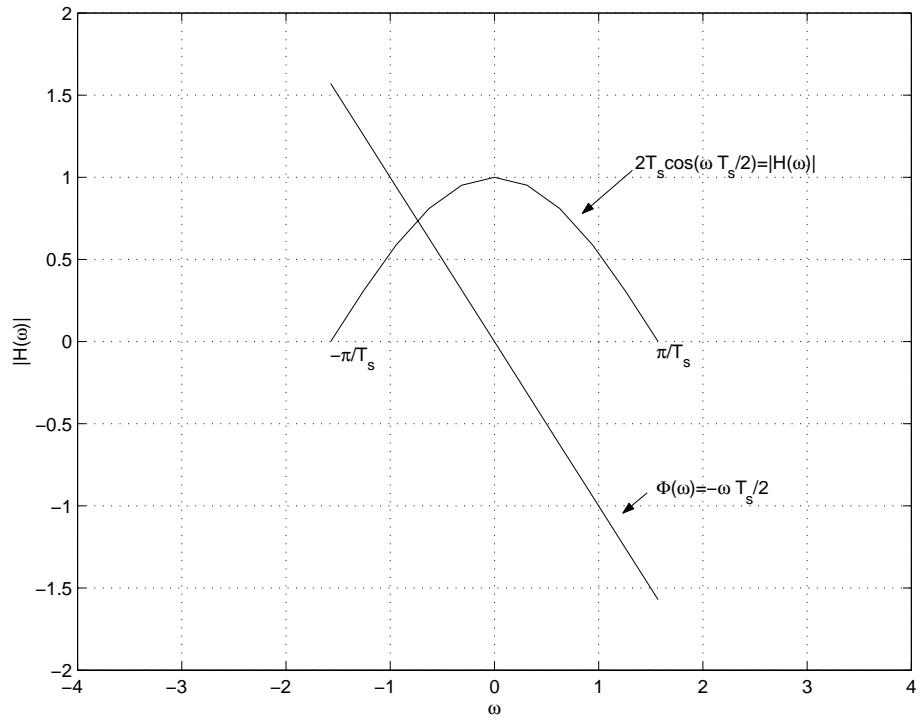


Figure 14:



17. • Show that in the word synchronization scheme as shown in Fig. I.10.2, if each word has  $n$  bits excluding the sync. bit, and if  $M$  of the sync. bits are summed together, the probability of having a synchronization error is given by  $P_e = 1 - [1 - (1/2^M)]^n$ .
- Let  $P_c = P(\text{correct word sync.}) = 1 - P_e$   
 $P_c = P(\text{The 1st bit in each of the } M \text{ data frames is not equal to 1}) \times$   
 $P(\text{The 2nd bit in each of the } M \text{ data frames is not equal to 1}) \times \dots$   
 $P(\text{The } n\text{th bit in each of the } M \text{ data frames is not equal to 1})$   
 Assuming the received bits to be independent and identically distributed, therefore  
 $P_c = P^n(\text{The 1st bit in each of the } M \text{ data frames is not equal to 1})$   
 But the probability of occurrence of a specific state in a binary  $M$ -bit register =  $\frac{1}{2^M}$ .  
 Thus the probability that this specific state (of all 1's) does not occur =  $1 - \frac{1}{2^M}$ . Thus,

$$P_c = \left(1 - \frac{1}{2^M}\right)^n$$

Equivalently,

$$P_e = 1 - P_c = 1 - \left(1 - \frac{1}{2^M}\right)^n$$