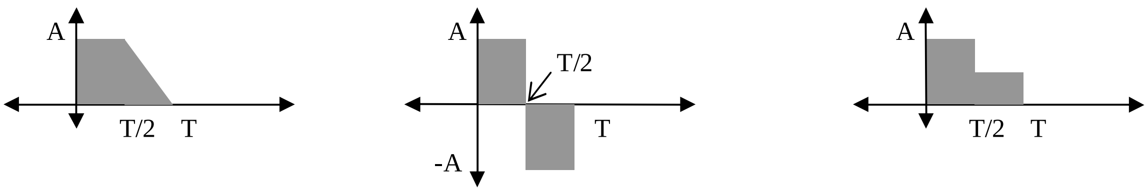
EC-3201 Communication Systems

Feb. 2012 **Tutorial #1** KG / IITM

1. Draw the output of the matched filter over 3 symbol intervals for the bit-sequence 1,-1,1 for each of the below choices of the pulse-shape $g(t)$:



2. For a bit-stream with 20 consecutive bits given by 100000110000000001 make a rough plot of the following line-codes: (a) NRZ (b) AMI (c) Split-phase Manchester (d) CMI (e) Differential Modulation (f) Differential Split-phase Manchester (g) B3ZS (or HDB3)

3. If the split-phase Manchester code with "rectangular" pulse shape is used for signaling, describe the matched filter receiver structure using a block diagram. If the transmitted stream is "1011" over 4 bit intervals, what will be signal shape at the output of the matched filter(s) ? *Hint:* There is more than one way to define the matched filter receiver, and the decoding logic in this case.

4. A state-dependent (1,3) Miller code of rate 1/2 is defined by the following rule:
(a) input "0" → output di-bit is: x 0 where x=0 if preceding input bit is "1" (and "x=1" otherwise)
(b) input "1" → output di-bit is: 0 1
Use this code on the bit-sequence in Pbm.2. What properties of the Miller code do you observe? What does the (1,3) mean?

5. A state independent (2,7) variable block size code, used by IBM storage devices, is described as follows:

Input Bits	Output Bits
10	1000
11	0100
011	000100
010	001000
000	100100
0011	00100100
0010	00001000

Use this on the bit-stream in Pbm.2. What do you observe? What does (2,7) mean? This code is said to be uniquely, and instantaneously decodable. What does this mean?

6. From the EC-305 Tutorial #1 of year 2008, solve problems 1, 2, 4, & 5 on <u>uniform</u> quantization.

7. Llyod-Max Non-uniform Quantization: Assume that a random signal x(t) with $x=x(t=t_0)$, follows a probability density function $f_X(x)$. and the objective is to find the N-1 signal levels $\{a_1, a_2, \cdots, a_{N-1}\}$ and the corresponding N quantized values $\{\hat{x}_1, \hat{x}_2, \cdots, \hat{x}_N\}$ such that the quantization noise variance given by

$$E_q = E[(x-x_q)^2] = \int_{-\infty}^{a_1} (x-\hat{x}_1)^2 f_X(x)\,dx + \sum_{i=2}^{N-1} \int_{a_{i-1}}^{a_i} (x-\hat{x}_i)^2 f_X(x)\,dx + \int_{a_{N-1}}^{\infty} (x-\hat{x}_N)^2 f_X(x)\,dx$$

is minimized. In other words, differentiate E_q w.r.t $\{a_1,\cdots,a_{N-1}\}$ and $\{\hat{x}_1,\cdots,\hat{x}_N\}$, equate them to zero, and show the following:

$$(i)\, a_i = \frac{\hat{x}_i + \hat{x}_{i+1}}{2} \quad \text{and} \quad (ii)\, \hat{x}_i = \frac{\int_{a_{i-1}}^{a_i} x\, f_X(x)\,dx}{\int_{a_{i-1}}^{a_i} f_X(x)\,dx}$$

8. The Llyod-Max quantizer indicates that the signal level should be the mid-point of the quantization interval (see (*i*)), and that the quantized value is the "centroid" of the corresponding interval (see (*ii*)). However, a practical method to construct this non-uniform quantizer is given by the iterative method below for a *k*-bit quantizer where $N=2^k$:

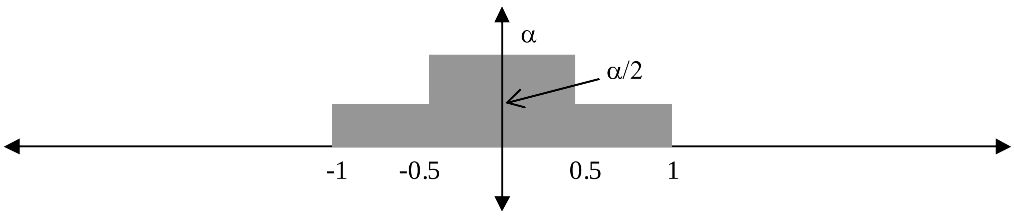
Step 1: choose *N*-1 uniform intervals $\{a_1, a_2, \cdots, a_{N-1}\}$

Step 2: find the corresponding centroids $\{\hat{x}_1, \hat{x}_2, \cdots, \hat{x}_N\}$ using (*ii*)

Step 3: re-compute $\{a_i\}$ using (*ii*) in (*i*)

Iterate between steps 2 and 3 until "convergence"

Use the above procedure for 3 iterations to define the 2-bit LM quantizer for the below source pdf. Compare the E_q obtained between the uniform quantizer (mid-tread) and the LM quantizer.



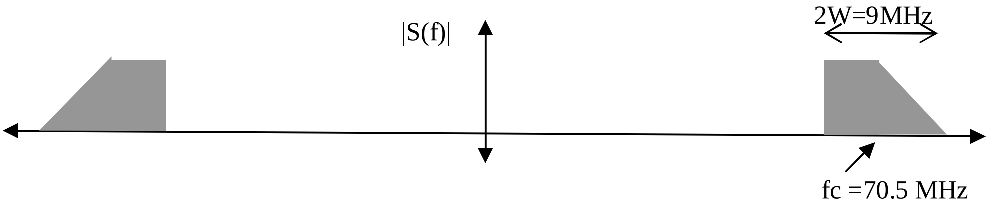
9. From the EC-305 Tutorial #1 of year 2008, solve problem 6 on non-uniform quantization.

10. From "Digital Telephony" J.C.Bellamy, 3rd Ed., pp. 158-160: Problems **3.1, 3.2, 3.3, 3.4, 3.5, 3.8*, 3.14*, 3.16*,3.17*, 3.18,** and **3.19.**

11. A low-pass signal of one-sided bandwidth of W=2MHz is sent as a DSB-SC signal. If the receiver uses an IF sampling scheme, with center frequency f_{IF} = 71MHz, determine the least sampling rate required. (Assume that the incoming signal frequency and the receiver's notion of frequency are identical.)

12. In the above problem, suppose that the incoming signal has f_{IF} = 71MHz, while the receiver synthesizes f_{IF} = 71.5MHz, then, what will happen to the received samples? In other words, provide the time-domain description of the samples at the output of the ADC in this case where there is a frequency offset.

13. For the QCM signal with magnitude response as below, find the least possible band-pass sampling rate. Make a rough plot of the frequency response of the sampled sequence around 0Hz.



14. In the above problem, assume that the received signal has a phase offset of θ radians; in other words, $s(t) = m_1(t)Cos(2\pi f_c t + \theta) + m_2(t)Sin(2\pi f_c t + \theta)$. Now, what will be the time-domain representation of the sampled sequence? For the special case when $\theta = \pi/2$,what will be the samples of the received signal?

15. A dozen DSB-SC signals of one-sided (low-pass) bandwidth W = 4MHz are present between 800MHz and 896MHz, as shown below. Describe the operations (sampling, rate-conversion, filtering) that you need to do to recover Nyquist rate samples of the 7th DSB-SC signal (i.e., the signal present between 848Mz and 856MHz).