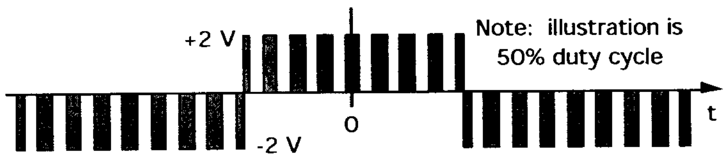
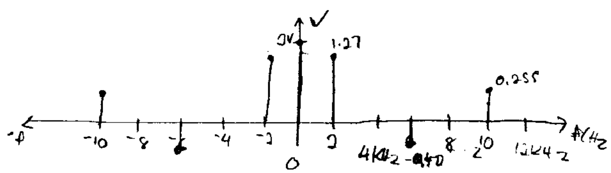
Final - 2001

1. A 4 Vp-p 2 kHz square waveform is sampled at 32 kHz using 1/3 duty cycle natural sampling. The square waveform has zero average value.
   - (a) Sketch the amplitude spectrum of the unsampled square waveform on calibrated axes. Consider only components with frequency 10 kHz or less.
   - (b) Sketch the magnitude of the amplitude spectrum of the sampled waveform on calibrated axes up to 100 kHz.
   - (c) Use Parseval's theorem to verify your results in parts a) and b).



a)



$C_0 = V_{AVG} = 0$

$C_n = A\left(\frac{\tau}{T_0}\right)\left(\mathrm{sinc}\, n\pi\left(\frac{\tau}{T_0}\right)\right)$

$\tau = 0.25\,ms$

$T_0 = 0.5\,ms$

$A = 4\,V$

$C_1 = 4V(\frac{1}{2})(\mathrm{sinc}\, 1\pi(\frac{1}{2})) = 1.27$ @ 2K

$C_2 = 0$

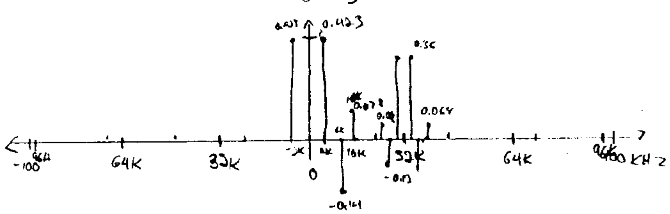
$C_3 = 4V(\frac{1}{2})\mathrm{sinc}(3\pi\frac{1}{2}) = -0.42\ V$

$C_4 = 0$

$C_5 = 0.255\ V$

b) $s(t) = A \cdot D.C. \cdot \mathrm{sinc}\, \pi \tau f$ $\quad \frac{\tau}{T_0} = \frac{1}{3} = D.C$

A



c) a) $P = |C_1|^2 + |C_3|^2 + |C_5|^2 + \approx 3.7\ W$

$P_{Tot} = (V_{rms})^2 = 4\ W$

2. Digitally coded signals are used in most new communication systems. Three techniques used to improve quality are companding, quantizer adaption and oversampling.
   - (a) Compare 16 bit LPCM coding to 12 bit "Mu Law" coding for music signals. Your discussion should include SNR at maximum signal levels, SNR at average signal levels, idle noise and dynamic range.
   - (b) Companding and quantizer adaption acheive a similar benefit for the system user. Explain how the two methods differ in implementation.
   - (c) An audio signal with frequency band 300-3300 Hz sampled at 8 kHz and coded with 10 bit LPCM. Assume an ideal output lowpass filter with bandwidth 4 kHz. Calculate the output SNR for a full load sinusoid in this system. Also calculate the SNR when the sinusoid is oversampled at 160 kHz rate.

2. a) LPCM: $SNR = 6.02(N) + 1.77 = 74.01$

Mulaw has greater SNR at small signals

dynamic range is more constant for mulaw

" " decreases as SNR decreases in LPCM

Mulaw has zero Idle Noise

b) Adaptive quantization uses a linear quantizer whose the step size is increased or decreased based on the recent history of the input signal. The step sizes of LPCM do not change with time. (time-invariant)

c) $SNR = 6.02(10) + 1.77 + 10\log_{10}\left(\frac{f_s/2}{f_{BW}}\right)$

$= 61.97 + 10\log_{10}\left(\frac{4kHz}{4kHz}\right)$

$= 61.97$

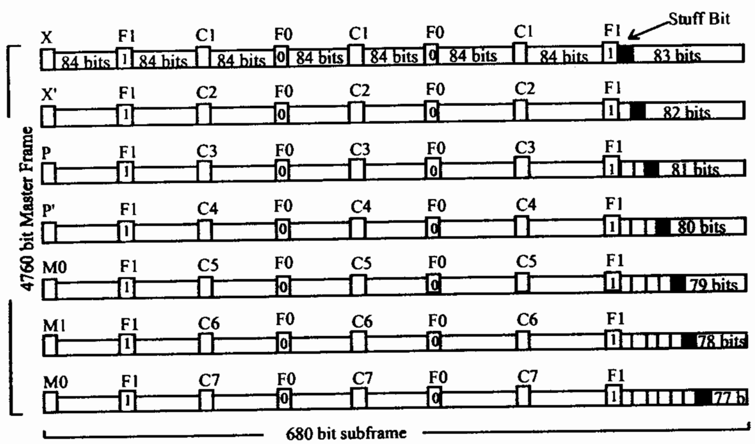
$SNR = 6.02(10) + 1.77 + 10\log_{10}\left(\frac{160k/2}{4k}\right)$

$= 74.9$

3. (a) A M1-C multiplexer is operating with nominal rate DS-1 inputs at 1.544 Mb/s and has an output rate of 3.152 Mb/s. What is the probability that a given stuff location will carry DS-1 information? (stuff control bits will be 000) Select the closest answer.

0.30 0.35 0.40 0.45 0.50 0.55 0.60 0.65 0.70

(b) The multiplexing format for DS-3 digital transmission at 44.732 Mb/s is illustrated below. Seven DS-2 input tributaries at the nominal rate of 6.312 Mb/s can be carried by the DS-3 system. For each input channel, there is one stuff bit in the 4760 bit frame. Assuming that the DS-3 frequency is exact, determine the allowed range of DS-2 input bit rate.



3.a) Max rate = $\frac{(624)}{1272}$ 3.152 Mb/s

$\frac{\text{actual} - \text{min}}{\text{max} - \text{min}}$

Min rate = $\frac{(624-1)}{1272}$ 3.152 =

0.543 $\boxed{\text{Pg } 105}$

b) Max rate = $\frac{12 \times 8 \times 7}{4760}(44.732) = 6.315$ Mb/s

Min = $\frac{12 \times 8 \times 7 - 1}{4760}(44.732) = 6.306$ Mb/s

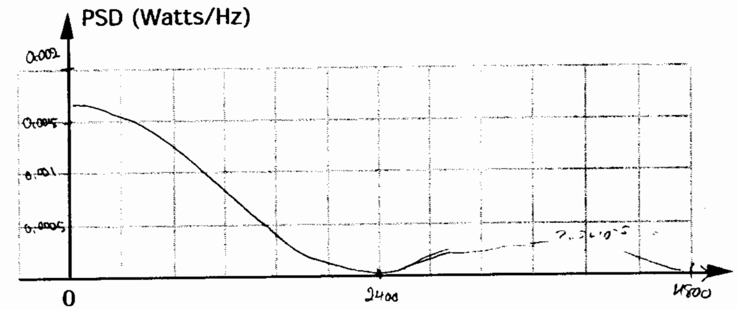
4. A 4 level 2B1Q (two binary, one quaternary) data signal can be developed as the voltage sum of two independent NRZ data streams. If the data in the two NRZ signals are uncorrelated, the power spectral density (PSD) of the 2B1Q signal will simply be the sum of the NRZ power spectral densities.

a) Consider a ± 2 volt NRZ random data signal at rate 2400 b/s. Sketch the two sided normalized PSD on calibrated axes.

b) Add to the signal in part a) a second NRZ random ± 1 volt data signal and sketch the resulting 2 sided PSD on calibrated axes.

c) What are the voltage levels of the resulting 2B1Q signal ?

d) Revise the sketch b) to show the result of passing through a channel which meets the Nyquist conditions for zero intersymbol interference (ISI) and has a raised cosine rolloff factor of 0.3.



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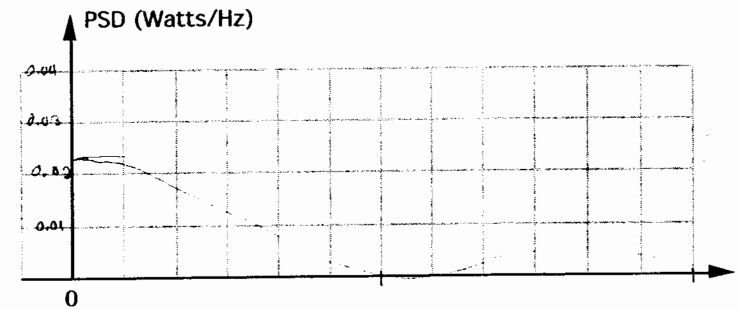
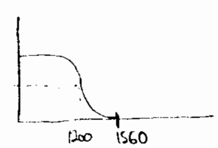


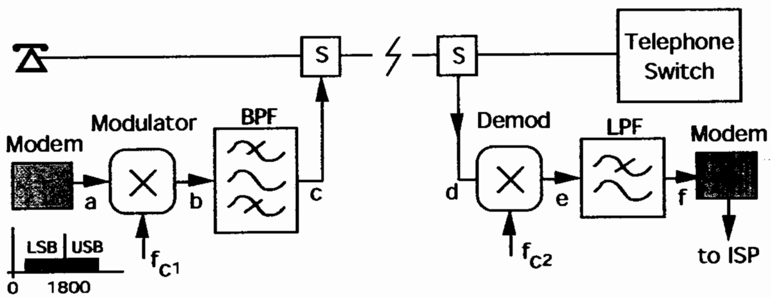
c) +3, +1, −1, −3

d)



5. With "unbundling" of the telephone system, regulators are allowing competition in long distance service and will soon allow competition in local service. Illustrated below is a proposal where an Internet sevice provider (ISP) could use a high frequency/low frequency splitter (S) to provide data service on a customer telephone line in addition to the regular 300-3400 Hz voice service. The voiceband modem signal with a carrier frequency of 1800 Hz is frequency is first translated so that the carrier frequency is 6200 Hz on the telephone line and then translated back to 1800 Hz at the receiving modem. This results in frequency division multiplexing (FDM) for the two services. For the purpose of this question, assume one way transmission.

At the modem, a minimum spectral band of 600-3000 Hz is required to meet Nyquist signalling criteria for zero ISI. Assume that a 0.25 rolloff factor is used which results in a modem bandwidth of 300-3300 Hz. The modem signal amplitude is -2 dBV and the carrier is suppressed.



6200Hz - 1800Hz = 4400 Hz

a) What frequencies are required for f_{c1} and f_{c2} ?

b) The modulator/demodulator could be a sampling gate, a ring modulator or an analog multiplier. Select one and justify the choice as optimum in terms of complexity (cost) and filter requirements.

c) Illustrate the signal spectra at points a, b, c, (assume d=c), e and f based on your choices in a) and b). Assume there is no voice signal.

d) Sketch the required magnitude vs frequency response of the two filters assuming that unwanted signals must be attenuated by 30 dB.

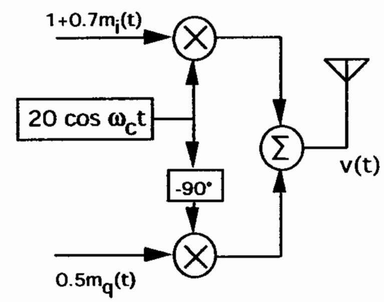
6. A QAM signal is generated by the modulator illustrated below. The carrier frequency is 600 kHz and the sinusoidal modulating signals are:
$m_i(t) = \cos 2\pi 125\, t$ and $m_q(t) = \sin 2\pi 125\, t$

a) Label the correct points on the diagram with DSB-SC and DSB-TC.

b) Recall the relations below (see Couch 5 p228) for complex envelope representation. For the output QAM signal v(t), determine expressions for x(t), y(t) and g(t).

$$v(t) = \mathrm{Re}\{g(t)e^{j\omega_c t}\} = x(t) \cos \omega_c t - y(t) \sin \omega_c t$$
$$g(t) = x(t) + jy(t)$$

c) Sketch the complex envelope "phasor" on real and imaginary axes and a locus to show its variation with time. Identify points at t=0, t=1ms, t=2ms, etc..



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