

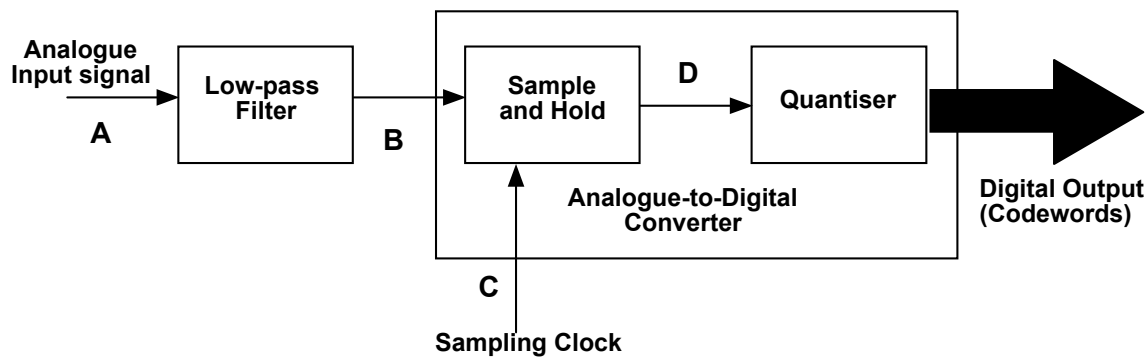
EEE116 – Multimedia Systems 2007/08
Solution to tutorial problem sheet 5 (Week 6)

(Q.9)

An audio signal containing 3 large amplitude frequency components at frequencies 5kHz, 10kHz and 20kHz is sampled at the Nyquist rate into 6 distinct voltage levels as follows: {-2, -1, 0, 1, 2, 3}. The corresponding probabilities of occurrence for these voltage values are {0.4, 0.06, 0.08, 0.12, 0.25, 0.09}, respectively.

- (a) Draw a block diagram of a system for digitising this audio signal and specify the parameter values used for the main components of the system. (Assume that the capturing device adds +/- 1 mV noise to the signal at the time of capturing).

The block diagram:



Parameters:

Signal has 3 frequency components. The highest is 20 kHz. Therefore the sampling system has to preserve all frequencies up to 20 kHz. This means the cut off frequency of the low pass filter is 20 kHz.

The signal is sampled at the Nyquist rate, which is $2 \times 20 = 40$ kHz. Therefore the sampling clock is 40 kHz.

The quantiser has 6 different quantisation levels (Value C in lectures). Therefore the digital code for PCM should use $N=3$ bits for the fixed length digital code.

- (b) If the signal is encoded using Huffman coding, estimate the time to send a recording of 5 minutes of this signal to a destination at 300 km distance via a link with bandwidth of 10 MHz and an SNR of 20 dB.

Huffman coding:

	P _i									Codewords
A	0.4							0	1.00	0
E	0.25					0	0.6	1		10
D	0.12	1	0.21	1	0.35	1				1111
F	0.09	0								1110
C	0.08	1	0.14	0						1101
B	0.06	0								1100

The average code length using Huffman coding is $\sum_{i=1}^5 n_i P_i$

$$= 1 \times 0.4 + 2 \times 0.25 + 4 \times (0.12 + 0.09 + 0.08 + 0.06)$$

$$= 2.3 \text{ bits/sample}$$

Sampling rate 40 kHz

Therefore the data rate = 2.3 x 40 k = 92 k bits/sec.

File size for a 5 minute recoding = 92 k x 60 x 5 = 27 600 k bits.

Now let's compute channel capacity.

Bandwidth 10 MHz

SNR is 20 dB. We need to express this as a ratio before using in the Shannon Hartley equation.

$$10 \log_{10} (S/N) = 20 \text{ dB}$$

$$\log_{10} (S/N) = 2$$

$$S/N = 10^2 = 100.$$

$$\text{Maximum Data capacity} = W \log_2(1+S/N) = 10 \times 10^6 \log_2(1+ 100)$$

$$= 6.658 \times 10^7 \text{ bits/sec}$$

Total delay = Propagation time + Transmit time

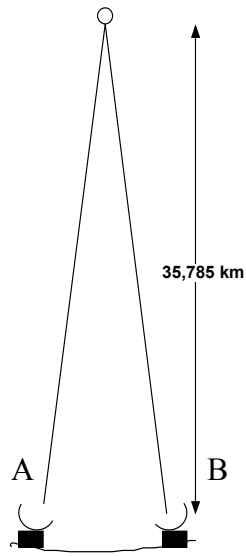
$$= \text{distance/velocity} + \text{data size/channel capacity}$$

(Assuming the link is operating at its maximum capacity)

$$= (300 \times 10^3 / 3 \times 10^8) + (27\,600 \times 10^3 / 6.658 \times 10^7)$$

$$= 0.001 + 0.415 = 0.416 \text{ sec.}$$

(Q.10)



How long would it take to send a 3 minute long CD-quality stereo (2-channels) audio bit stream recorded using 44.1 kHz sampling rate and 16 bits representation from A to B using a geosynchronous satellite link ?

Assume 256 kbps link to and from the satellite, and that the satellite takes 50 ms to receive and then retransmit a signal. Speed of light is $3 \times 10^8 \text{ ms}^{-1}$.

First we have to compute the data size of the music file.

We need to know the data rate for that:

$$\begin{aligned} \text{Data rate for audio: average code length} \times \text{sampling rate} \times \text{number of channels} \\ = 16 \times 44.1 \text{ k} \times 2 \end{aligned}$$

Therefore the data size is data rate \times duration

$$= 16 \times 44.1 \text{ k} \times 2 \times (3 \times 60)$$

Time to send the file is

$$\begin{aligned} &= \text{Propagation time} + \text{transmit time} + \text{other} \\ &= (35785 \text{ k} \times 2) / (3 \times 10^8) + (16 \times 44.1 \text{ k} \times 2 \times 3 \times 60) / (256 \text{ k}) + 0.050 \\ &= 992.5 \text{ s} \\ &= 16.5 \text{ minutes.} \end{aligned}$$

If we use this type of transmission for a live broadcasting, it requires 16.5 minutes to send a 3 minute long audio clip. In order to playback without interruption, at the decoder, we need a buffer to store the full download and then playback.

What would be the effect on the audio quality and the total time taken to reach the destination, if 8 bits per sample quantisation was used instead of 16 bits per sample?

Using a fewer bits per sample quantisation introduces distortion. Since the same sampling rate is used, the frequency response is preserved.

This would halve the data size and halve the transmit time. Since the transmit time is significantly larger than the other two delays (propagation and queuing), this would halve the total time to send the data. (Now the total time is around 8.27 minutes)