Audio Coding - Exercise

Multimedia Technology,

Tutorial 2, section 4

Audio Coding

Suppose we want to digitize an analog audio signal (single channel) with a frequency range from 2 to 12 kHz. Calculate:

a) the required sampling rate,

b) the signal-to-noise ratio (SNR) of the sampling process when we have 12 bits per sample, and

c) the resulting bit rate based on the above parameters.

A) Sampling rate

- Nyquist Theorem: an analog signal can be digitized without aliasing error if and only if the sampling rate is greater than or equal to twice the highest frequency component in a given signal.
- I find the bandwidth to be 12-2 = 10 KHz
- The sampling rate is twice that, so 20 KHz
- What matters is the bandwidth, not the maximum frequency…
- …because the frequencies can be shifted (i.e. from 4 to 14 KHz).

B) the SNR when we have 12 bits per sample

- The SNR is at least 6*n when we have n bits per sample.
- So it will be 6 x 12 = 72 dB.
- We assume linear quantization.
- Variation: if a required SNR is given and the number of bits is asked for,
- Same steps, but backwards — dividing the SNR by 6.
- Example: if the SNR needs to be 48 dB,
- Thus I need 8 bits per sample.

Linear Quantization:
$20 \log_{10} 2^n = 6*n$,
I use the number 6 to multiply/divide due to the logarithmic relationship between the number of bits (n) and the signal-to-noise ratio (SNR) in decibels.
(Refer to the lecture slides for more information)

c) Resulting bit rate based on the parameters.

- I multiply the sampling rate (20 kHz = 20,000 samples/sec)
- by the bits per sample (12 bits/sample)
- So it will be 240 kbps.