Acoustic modem project

# Session 1: Audio playback, recording and analysis

## Exercise 1-2: Time-frequency analysis of recorded signals

1. (C) Bartlett’s Method: No overlapping.  
    Welch’s Method: Overlapping segments, these segments are windowed ( in time domain)  
    PSD is not averaged out.
2. There is some leakage because of the nature of the DFT.
3. A large DFT size gives us a more detailed frequency spectrum.
4. There is noise recorded. You can see this contribution on other frequencies

the original signal frequency is clearly visible, when no loud noises are made during the recording, there is mostly random noise present, visible by looking at the dB measurement at the other frequencies. This noise is not a white noise, since the channel is not flat.

1. There is a large DC component and some smearing to the low frequencies, there are also harmonics from the original signal present. The DC component and smearing is expected due to the limited size window used in the DFT.
2. There are other frequencies due to harmonics. The frequencies are a multiplication of the original frequency. These can be harmful to the acoustic modem because these harmonics could interfere with the original signal in a way that’s difficult to filter out.
3. Added rescale(sig, -1,1) in initparams.
4. In the spectrum, we recognize the frequencies: 500Hz, 1500Hz, 2000Hz, 4000Hz and 6000Hz. In the PSD, more frequencies are recognizable, but the higher ones are more dominant. The recorded signals contain less the low frequencies and has also some noise, which you can see in the other frequencies.
5. We can see which frequency bands are well recorded. Some frequency bands are less visible in the spectrogram which means that or the speakers don’t play these properly, or the microphone can’t record this sound very well.
6. In time, it is just a random distribution of frequencies. It’s changing in time but there is not a pattern. It’s completely random. This information is important, because otherwise the transfer function coefficients of the channel would be time dependant.
7. When the microphone moves further away, the received power in each spectrum goes down.

## Exercise 1-3: Your best friend Shannon

1. The useful frequency range of the mic is : 50 Hz - 16 kHz. However we don’t know of this is a flat frequency response. The laptop speakers will have an equivalent range. But also not a flat range. The low frequencies are most of the time poorly represented.
2. The signal and the noise do not correlate, because these are powers. We can add or subtract these. The scaling factor is the frequency resolutions of the PSD’s, this is the factor that takes the bandwith of the signal into account (like the factor B in the integral of the continuous version.)
3. It’s very dependant on the surrounding noise. We got values from 300 to 1500 bits/sec.3
4. Channel capacity: how much information we can send over the channel per second with a specified bandwidth. If this is higher, we have more capacity, so we can send more ‘bits’ per second. This depends on the amount of noise and the power of the channel.

# Session 2: Estimation and analysis of the acoustic channel

## Exercise 2-1 : A first attempt to estimate the channel response

1. –
2. Around 250 samples which means 15ms.
3. Much wider in time and a delayed signal(echo) will also be seen in time domain.
4. The microphone and speakers have an influence.

## Exercise 2-2: A robust channel response estimation

1. Y[k] = h\*u[k] (convolution)