Afbeelding met schermopname, diagram, tekst, schets

Automatisch gegenereerde beschrijvingPracticum 1 PSD

# Vraag 1

function [ simin,nbsecs,fs ] = initparams( toplay, fs, varargin )

* Simin:
  + 2-column matrix (total aantal samples x 2) = [toplay zeros(size(toplay))]
  + the first columns contains the vector toplay, with 2 seconds of silence in the beginning, and one second of silence at the end. The second column contains all zeros
* toplay :
  + with 2 seconds of silence in the beginning
  + one second of silence at the end
  + sinewave = a 2-seconds long sine wave of 1500 Hz, saved in the variable sinewave
* nbsecs: The number of seconds the playback/recording must run (should be at least as long as the signal in simin).
* fs: This is the same fs that is given as an input.
  + 16000HZ

# Vraag 2

### Deel 1

In a separate figure, do the same for the PSD of the transmitted and recorded signal (again a dB scale). Calculate the PSD by averaging multiple spectrum estimates, all with a DFT size equal to N.

This way of estimating the PSD is referred to as Bartlett’s method or Welch’s method.

* What is the difference between these two methods, and how does their result differ from the actual PSD?

Ant:

* + The Bartlett's method uses sequential, non-overlapped, windowless data segments. The Welch method uses overlapped windowed data segments.
  + https://www.youtube.com/watch?v=YK1F0-3VvQIAfbeelding met tekst, schermopname, Lettertype

    Automatisch gegenereerde beschrijving
  + uitleg punt 3: want je gebruikt minder punten in welch (window) terwijl de full fft alle punten van het spectrum neemt.
  + Uitleg punt 4: temporal dynamics heb je wel in een spectogram met short fourier transform
* Also calculate the PSD by squaring the frequency magnitude spectrum without averaging. Why are Welch’s method and Bartlett’s method preferred over squaring the magnitude frequency spectrum?

ChatGPT

Welch's method and Bartlett's method are preferred over simply squaring the magnitude of the frequency spectrum when estimating the power spectral density (PSD) of a signal because they provide more reliable and statistically stable estimates of the PSD. Here’s why:

**1. Reducing Variability in the Estimate**

* **Squaring the Magnitude Spectrum**: This approach, also called the "periodogram" method, involves taking the Fourier transform of the signal, then squaring the magnitude to get the power spectrum. However, this technique results in a highly variable PSD estimate, as it is sensitive to noise and fluctuations in the signal.
* **Welch's and Bartlett's Methods**: Both methods reduce variability by averaging multiple periodograms calculated over different sections of the signal. This averaging process produces a smoother, more consistent estimate of the PSD, with reduced variance compared to a single periodogram.

**2. Segmenting and Averaging to Improve Statistical Confidence**

* **Bartlett’s Method**: Divides the signal into non-overlapping segments, computes the periodogram for each segment, and then averages them. This method improves the statistical reliability of the PSD estimate by trading off frequency resolution for variance reduction. Each segment contributes independently to the final PSD estimate, so the averaging reduces the influence of random noise in each segment.
* **Welch’s Method**: Extends Bartlett's method by using overlapping segments (typically with 50% overlap), which provides even better variance reduction. The overlapping segments mean more periodograms can be averaged without needing to increase the segment length, which preserves frequency resolution while reducing the variance further.

**3. Trade-off Between Frequency Resolution and Variance**

* Welch's and Bartlett’s methods allow for a trade-off between frequency resolution and variance. By controlling the segment length and overlap (in Welch’s case), users can balance the desired frequency resolution against the statistical smoothness (lower variance) of the PSD estimate.
* **Periodogram Method** (squaring the spectrum) does not offer this flexibility, as it uses the entire length of the signal in a single window. The result may have higher frequency resolution but with high variance, making the spectral estimate less reliable.

**4. Reduced Spectral Leakage**

* By applying window functions (e.g., Hamming or Hanning windows) to each segment before computing the Fourier transform, Welch’s and Bartlett's methods reduce spectral leakage, which can otherwise distort the PSD estimate by introducing artificial peaks or smearing the power across frequencies.
* **Direct Squaring** (without windowing or averaging) can amplify spectral leakage, especially for non-periodic or finite-length signals, leading to misleading PSD estimates.

In summary, Welch's and Bartlett's methods offer advantages in terms of reducing variance, controlling spectral leakage, and providing statistically stable PSD estimates, making them preferred for power spectral density estimation over the simple periodogram approach.

The Power Spectral Density function computed for one signal cannot be negative. The only one case for such kind of output is the cross PSD for which the values for particular frequency are complex number. If you make a mistake in the interpretation of that output then you can assume, that you got negative values

### Deel 2

Set sig in analyze rec.m to a sine signal of 400 Hz, and execute the file. Take a look at the PSD and the spectrogram of the transmitted signal (i.e., what is sent to the loudspeakers, not what is recorded).

Is this what you expected? Are there other frequencies besides the 400 Hz present in the PSD/spectrogram? How can you get rid of the harmonics in the PSD?

Ant: Filter

### Deel 3

What is the influence of the DFT size? Try to answer first and then verify experimentally

Ant: Hoe meer samples, hoe meer precies. Spectrale precisie. Hoe ruwer de PSD eruit ziet.

The “total” integrated spectral density equals the variance of the series.

### Deel 6

Audio clipping is what occurs when an audio signal exceeds what a system can handle, leading to parts of the waveform being abruptly cut off. This typically happens if you record with levels too high or push your equipment's limits, like the master fader in a digital audio workstation (DAW) going into the red zone.

peak\_amplitude = max(abs(signal));

if peak\_amplitude > 1 scaled\_signal = signal / peak\_amplitude; else scaled\_signal = signal; % No scaling needed if already within range end

# Vraag 3

### Deel 1

One of those was **frequency response**, which is a measurement of how a particular microphone responds to a range of frequencies.

100-10kHz