Elimination of narrowband interference (sound of clarinet) in the human voice record

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1 Abstract

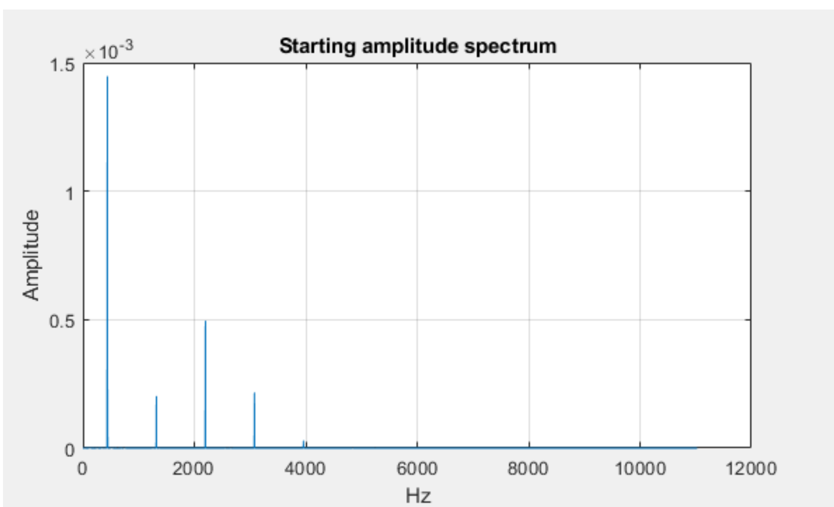
First I will introduce the task, and why I picked it, afterwards I describe my methods, and any problems I ran into. I worked with MatLab as it is the best tool to do sound filtering that I know of. At the end I describe the results and my findings regarding the task.

2 Introduction

The goal of removing narrowband interference is a very old problem people have been trying to solve for a long time. And accomplishing such a task wasn't very difficult, though it was somewhat time consuming, when using the bandstop filtering method.

3 Methods

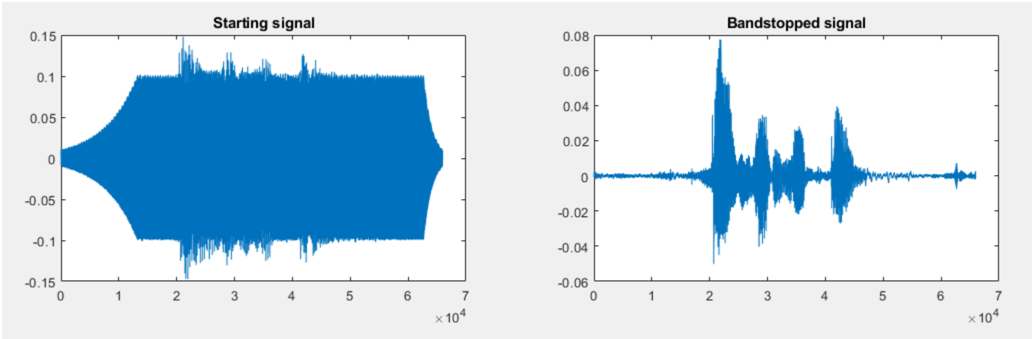
The first step in the process of eliminating narrowband interference from the recording was identifying which frequencies were interfering with the original the most. This was done by creating the amplitude spectrum of the given recording.



Slika 1: Amplitude of the received recording

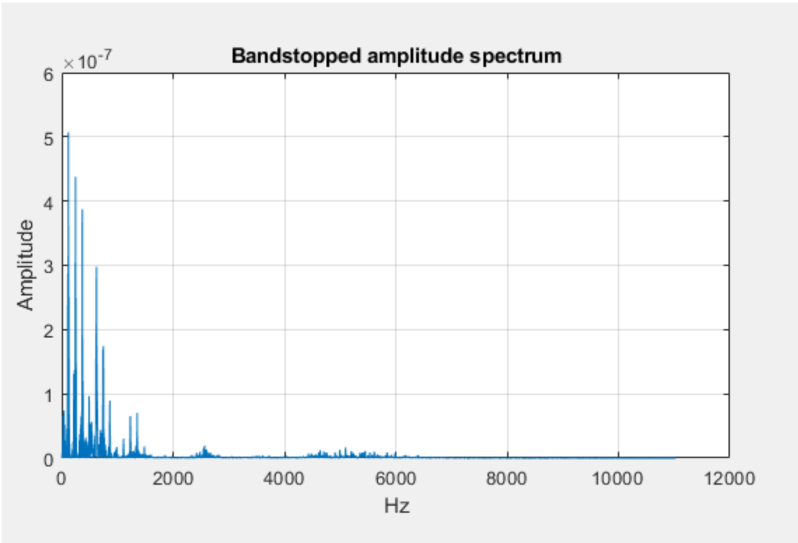
Subsequently I was able to remove all the identified frequencies using MatLab's built in function "bandstop". This function requires a signal, the sampling frequency, and most importantly it requires the frequencies from and to which the bandstop filter is supposed to work on. By using multiple, each with a range around the faulty frequencies. I was able to eliminate most of the clarinet sound.

4 Results



Slika 2: Signals before and after filtering

In these graphs displaying the signals before and after we can see just how much was removed using the bandstop filters. Ultimately some of the clarinet can still be heard, but I have removed as much as possible without distorting the human speech too much. It is clear from the graph that the human voice is much more distinct and audible as it was before.



Slika 3: Amplitude after using the bandstop functions

If we look at the amplitude of the signal after I used the bandstop functions, it is clearly visible just how much the spikes have reduced (of course when you look at the graph you need to consider that the scale is much smaller). When removing the frequencies, some were easy to identify and I was able to eliminate them without much trouble, removing some others however, was much more of a problem, as they were not very common, and subsequently didn't show as big of a spike. As I was applying the bandstop filters, I also got an unwanted result, which is the contortion of the human speech, that I was trying to preserve.

5 Discussion

Now I will explain how an amplitude spectrum graph is created. First we need to perform DFT (Discrete Fourier Transformation) on the original signal.

$$X(k) = \sum_{n=0}^{N-1} x(n)\, e^{-j(\frac{2\pi k}{N}).n}$$

After preforming this this transformation we are left with a complex number represented with a real value and an imaginary value. To create the amplitude spectrum we need to find the root of the sum of each of the two values squared.

$$|X(e^{j\omega})| = \sqrt{X_R^2(e^{j\omega}) + X_I^2(e^{j\omega})} \qquad |X(k)| = \sqrt{X_R^2(k) + X_I^2(k)}$$

After performing this, we are left with a function that maps all the points on the amplitude spectrum. Of course this manual method would only be used when working with very short signals, of with very short lengths (N).

The second part of the seminar required me to use the bandstop function. Bandstop is a function that greatly reduces the level of specified frequencies. Using these formula we can calculate H_{BS}(z), which helps us determine the filtering.

$$H_{BS}(z) = \frac{1+\alpha}{2} \frac{1 - 2\beta z^{-1} + z^{-2}}{1 - \beta(1+\alpha)z^{-1} + \alpha z^{-2}}$$

ω_C is center frequency (not cutoff): $\beta = \cos \omega_C$

B is 3dB bandwidth: $\frac{2\alpha}{1+\alpha^2} = \cos B$

To use the provided formula for H_{BS}(z), we must first determine α and β, which is easily achievable using the above formula, as ω_C and cos B are given when starting the calculation. z is a complex number that tells us in the form of polar coordinates where the two points lie inside the unit circle of complex numbers.

At the lecture we tested different IIR filters, and discovered that a bandstop filter is present (and effective), when the magnitude of zero is set to 1, and the angle of the pole is the same as the angle of the zero. When the pole is closer to the zero the filtering is more strict, and when it is further away, the filtering lets more of the noise through.