ECE2312 – Discrete-Time Signal and System Analysis
Project 3
Due: 5.00 pm on Thursday March 2, 2023

1. Project Goals

The objective of this course project is to explore the concept called filterbanks, where a signal can be represented as a collection of parallel subsignals, each of which contains a specific subband of frequency information from the original signal. Filterbanks can analyze a signal into N parallel subsignals, as well as synthesize these subsignals back into the original signal. For this project, you will construct your own analysis and synthesis filterbanks and apply it to one of the recorded speech signals from Project 1.

2. "The Big Picture"

For this project, each team is responsible for constructing the analysis filterbank shown in Figure 1, as well as its corresponding synthesis filterbank.

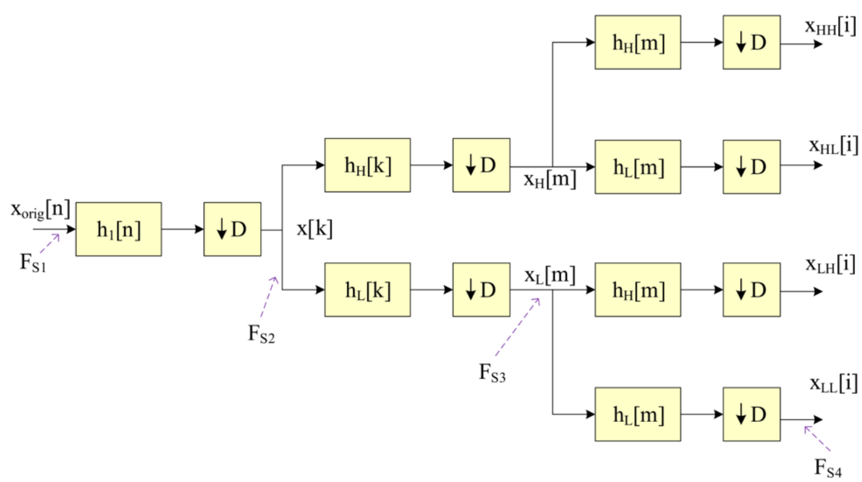

Figure 1: Illustration of an analysis filterbank, which takes a signal and ultimately divides it into four subsignals that consists of a specific subband of frequency for the original signal.

The first step in this process is to extract out only the speech signal and remove any content that is higher than audible speech. To achieve this, we perform the first decimation stage and apply it to the original signal $x_{orig}[n]$ of sampling frequency F_{S1}, yielding the speech-only signal $x[k]$ with sampling frequency F_{S2}. Note that used D to downsample the signal in this stage, although the **downsampling rate in the subsequent parts of the analysis filterbank might be different** based on what F_{S1} was chosen originally and being able to isolate the speech signal data from that recording.

Once the speech-only signal has been extracted, apply the analysis filterbank to it such that at every subsequent stage the lower and higher frequency components are filtered out, producing two separate subsignals. For instance, the lowpass filter $h_L[k]$ and highpass filter $h_H[k]$ are applied to $x[k]$ in two parallel processes and then subsequently downsampled by a factor of $D = 2$ in order to produce low frequency signal $x_L[m]$ and high frequency signal $x_H[m]$. This process is then repeated to each of these signals in order to produce signals $x_{LL}[i]$, $x_{LH}[i]$, $x_{HL}[i]$, *and* $x_{HH}[i]$. Note that the same lowpass and highpass filters are used at each stage of the analysis filterbank.

As for the synthesis filterbank, the exact opposite process is applied, where each signal is interpolated (upsampled and filtered) before being combined with its counterpart signal.

3. Focusing on the Speech

The first step of this project is to isolate the speech signal from your audio recordings from Project 1. For this project, please select one of your recordings for processing by the analysis and synthesis filterbanks you are about to implement. It should look similar to the spectrogram shown in Figure 2.

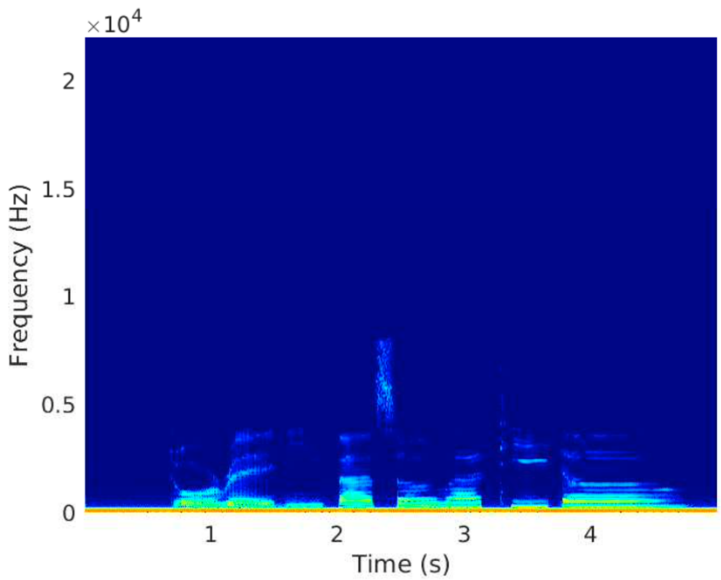

Figure 2: Spectrogram of Professor Wyglinski saying "do-re-mi-fa-so-la-ti-do" during a 5 second time interval. This speech recording was performed using a sampling rate of $F_s = 44000$ Hz and 8-bit resolution

For this section, you will need to filter out the speech content from your original recording and downsample it such that your signal only consists of that speech signal (see Figure 3).

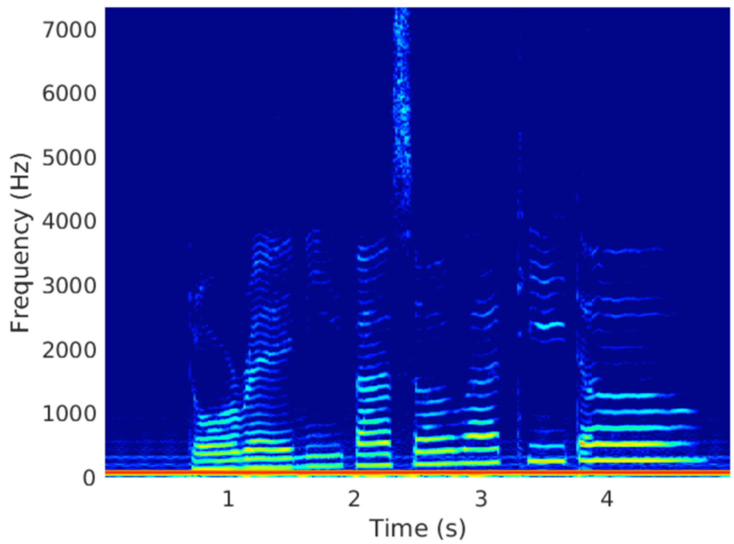

Figure 3: A decimated version of the audio recording shown in Figure 2 that only focuses on the speech content of that recording.

> Design a lowpass filter and downsampling implementation that only exacts out the speech content of one of your original audio recordings from Project 1.
>
> - Plot the spectrograms of the original audio recording and the speech-only version to include in your report.

4. First Frequency Decomposition

Using the speech-only version of your signal, the next step is to make two copies of it and lowpass filter one copy while highpass filter the other copy. The result should look like the spectrogram shown in Figure 4.

Once the filtered copies have been obtained, downsample both signals by a factor of two such that you produce outputs with spectrograms that appear like those shown in Figure 5.

> Design and implement the lowpass and highpass filters, as well as the down-sampling process, and apply them to the speech-only signal from the previous section.
>
> - Plot the spectrograms of the two downsampled versions of the lowpass and highpass filtered speech-only signals to include in your report.

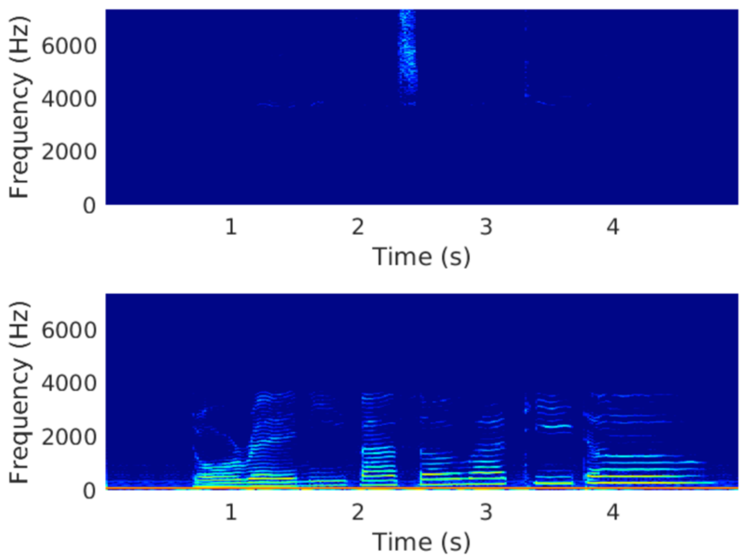

Figure 4: Lowpass filtered (bottom) and highpass filtered (top) versions of the speech-only signal from the previous section.

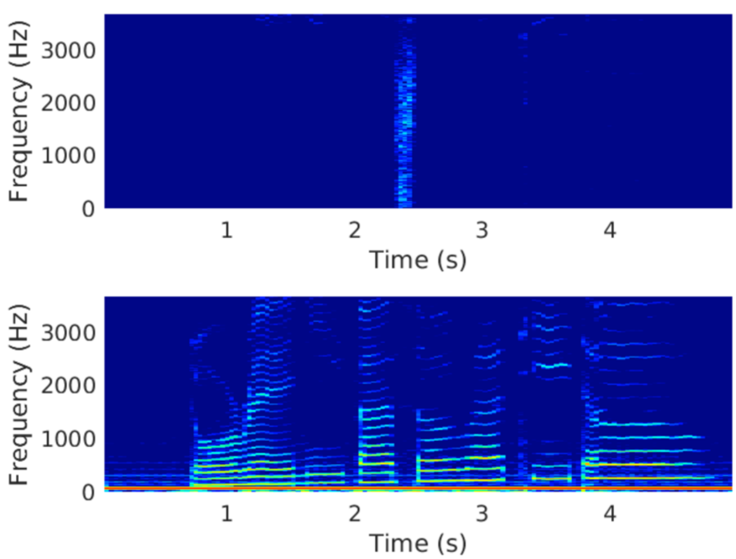

Figure 5: Downsampled versions of the lowpass filtered (bottom) and highpass filtered (top) versions of the speech-only signal.

5. Second Frequency Decomposition

Taking the two downsampled versions of the lowpass and highpass filtered speech-only signals from the previous section, let us make two copies of each signal and then lowpass and highpass filter each signal. Then, after being filtered, take all four filtered signals and downsample them by a factor of two. The filtered signals before downsampling should ressemble the signals shown in Figure 6 while the signals after downsampling should appear to be similar to those shown in Figure 7.

Apply the lowpass and highpass filters, as well as the downsampling process, from the previous section to generate the downsampled versions of the low-pass and highpass filtered versions of the downsampled lowpass filtered and highpass filtered versions of the speech-only signals from the previous section.

- Plot the spectrograms of the four resulting signals to include in your report.

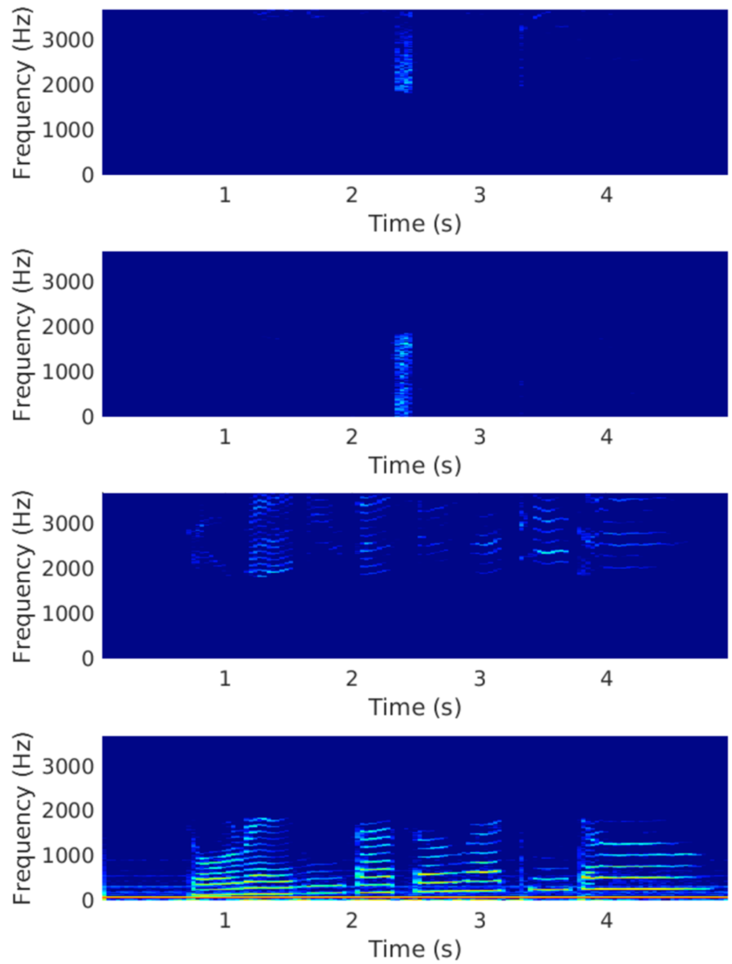

Figure 6: Lowpass and highpass filtered versions of the downsampled lowpass filtered and highpass filtered versions of the speech-only signals from the previous section. The top two subplots show the high frequencies and low frequencies of the downsampled highpass filtered speech-only signal, while the bottom two subplots show the high frequencies and low frequencies of the downsampled lowpass filtered speech-only signal.

6. Reassembling the Signal

Based on your implementation of the analysis filterbank, it is now time to reverse the process and recreate the original speech-only signal using a synthesis filterbank. Using the same technique but in reverse, one would need to use a collection of lowpass and highpass filters along with upsampling processes to generate a synthesized signal.

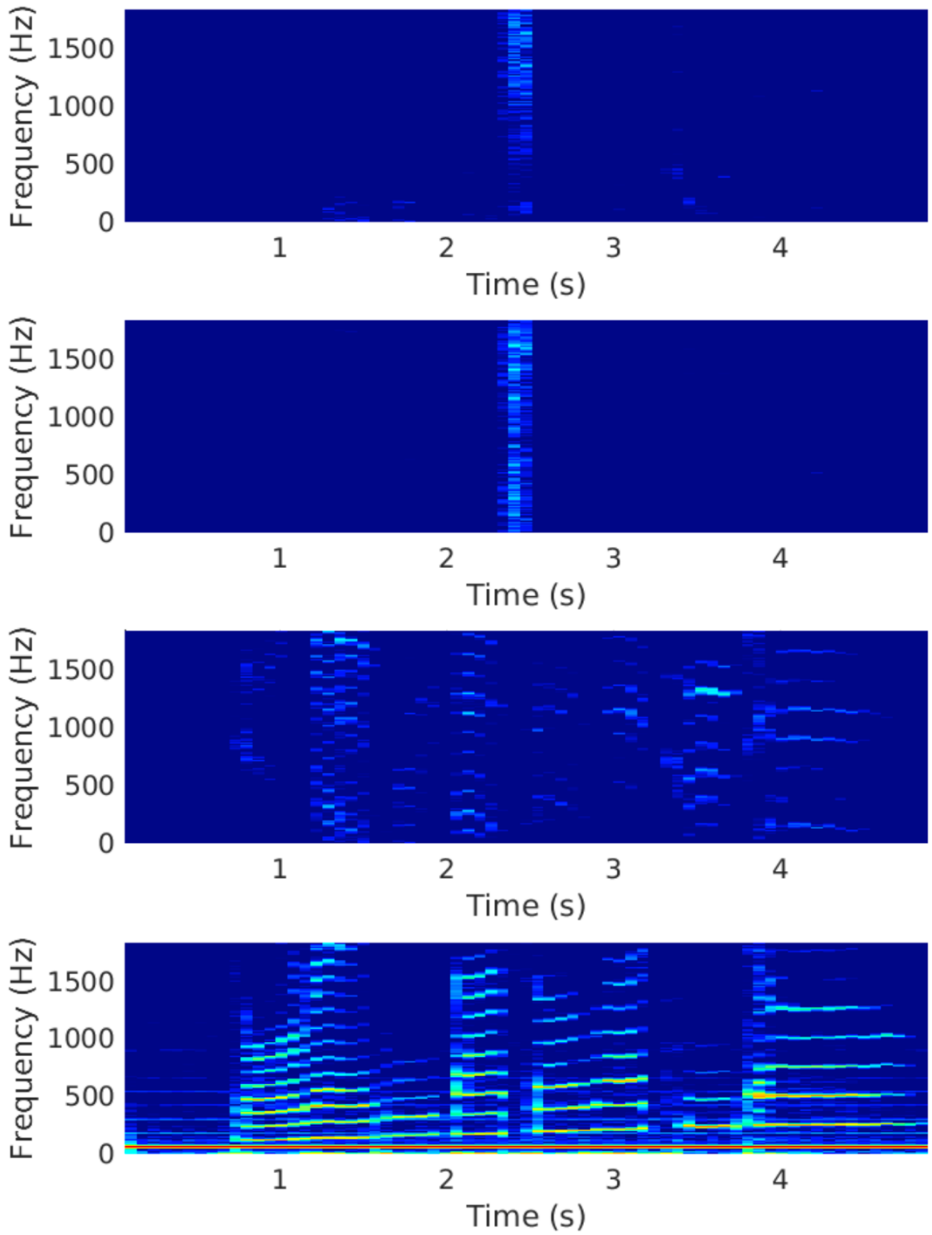

Figure 7: Downsampled versions of the filtered signals shown in Figure 6.

Reconstructed a synthesized signal from the four resulting signals from the previous section using the reverse process described for the analysis filterbank.

- Play your synthesized signal using the `sound` function and comment on what you hear.
- Save the resulting synthesized signal to a WAV file using the filename "team[[yourteamnumber]]-synthesized.wav" and include in your submission to CANVAS.
- Plot the spectrogram of synthesized signal to include in your report.

7. Project Submission

Each student team should submit the following via the ECE2312 CANVAS website:

- A project report in PDF format that answers all the questions indicated in this project handout. Additionally, the following elements should be included:
  - Coverpage with course number and title listed, names of all student members of submitting team, date of submission, project number.
  - Descriptive captions for all figures contained within report submission.
  - Sufficiently detailed responses to all questions, providing insights about the answers provided. Responses should be written in complete sentences/paragraphs, should be grammatically correct, and written using professional vocabulary.
  - Proper pagination and formatting (e.g., 1-inch margins, single-spaced lines, 11-point serif fonts).
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- All requested WAV files as indicated in this handout.
- Link to the Github/Gitlab containing all source code generated by the student team. This code should be in a condition that it can be executed by the teaching assistant to verify its functionality. Moreover, you will run and show this code on **March 2**.