6.003: Signal Processing

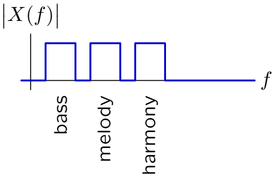
Filtering in Streaming Applications

15 April 2021

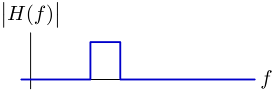
Filtering Music

Consider a song (contained in `am_synth.wav`), consisting of three separate "voices," each of which is band-limited:

- "bass": 40-170 Hz
- "melody": 170-370 Hz
- "harmony": 370-750 Hz



Consider the task of separating these three tracks, producing a new song consisting of, for example, only the "melody" part.



How can we do this?

Filtering Music

Now consider the same task, but with a recording of the same song played on guitars rather than on synthesized cosine waves (`am.wav`).

Predict how this same approach will perform on this recording.

And try it!

Filtering in a Streaming Application

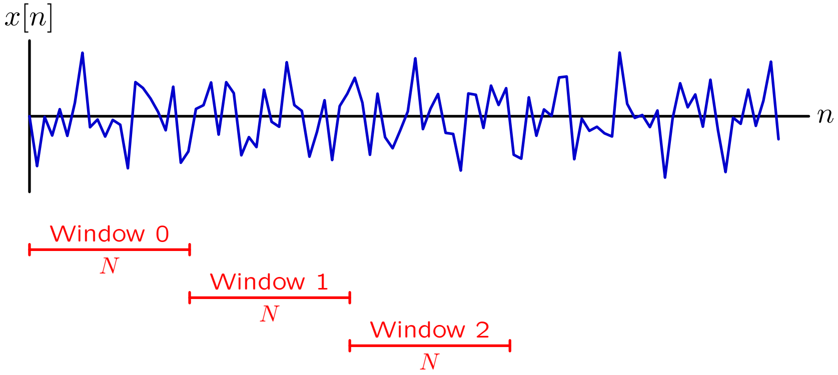
In many applications, we don't have the entire signal we want to process available to us at the start (we receive it a little bit at a time). Examples:

- a live speaker at an event
- streaming music online

How can we process these signals in a similar way, without access to the entire signal?

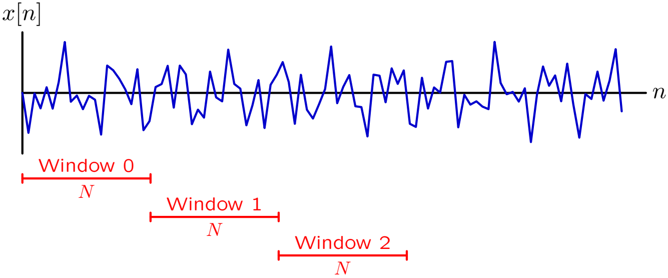
Algorithm 1

Short-time Fourier transforms are based on the analysis of a sequence of finite-length portions of an input signal.

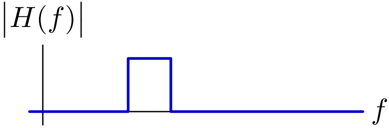


Algorithm 1

Chop the input signal into pieces that are each of length N.



Filter each piece by zeroing FFT components outside passband.

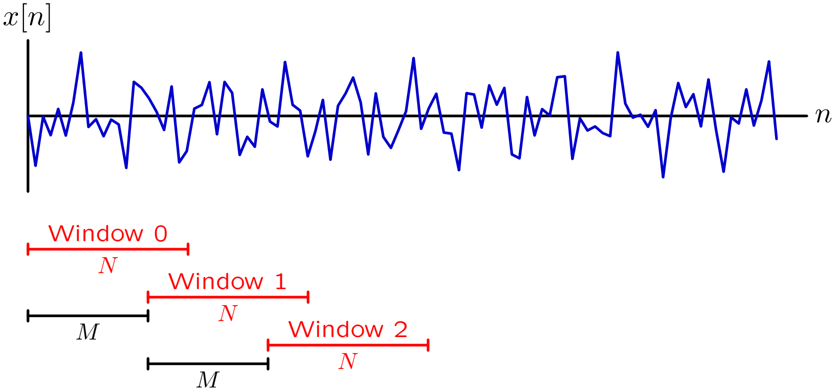


Compare original to this new result.

How effective is this algorithm? How can it be improved?

Overlap-Add Method

Algorithm 1's big problem can be fixed with overlapping windows.



How does overlapping help? How would you choose M and N?

Overlap-Add Method

Create a filter, but limit its unit sample response to some length L. Pad this unit sample response with some number M of zeros to create a unit sample response of length $N = L + M$.

Divide input signal into blocks of length M, which we pad with L zeros to produce a new window of length $N = M + L$.

Convert each length-N block to the frequency domain and multiply by the freqency-domain representation of the filter.

Convert this result back to the time domain. L partial values at the end of each block are added to L partial values at the beginning of the next block.

Overlap-Add: Graphical Depiction

Filter Design

Design a filter for the overlap-add method: $M = 6144$ and $N = 8192$. The filter should pass frequencies in the range $\Omega_l < \Omega < \Omega_h$.

Method 1: $N = 8192$

$$X[k] = \begin{cases} 1 & \text{if } N\frac{\Omega_l}{2\pi} \le |k| \le N\frac{\Omega_h}{2\pi} \\ 0 & \text{otherwise} \end{cases}$$

Method 2: $N = 2048$

$$X[k] = \begin{cases} 1 & \text{if } N\frac{\Omega_l}{2\pi} \le |k| \le N\frac{\Omega_h}{2\pi} \\ 0 & \text{otherwise} \end{cases}$$

Method 3: Start with method 2.
Then take inverse FFT; zero-pad to N=8192, and take FFT.

Method 4: Start with method 1.
Then take inverse FFT, apply rectangular window with width 2048, and take FFT.

Filter Design

What was wrong with the previous method? How can we fix it?

Overlap-Add Method

Importantly, we can process the first window without waiting for the entire song to be transmitted – very important for **streaming applications.**

But, it turns out that this method also tends to be more efficient in normal applications as well!