PROJECT
Presentation

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Digital Signal Processing
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How signal of interest are physically generated?

1. Human voice

Human voices are produced in vocal cords when they come together and then vibrate as air passes through them during exhalation of air from the lungs. This vibration produces the sound wave for your voice.

2. Instruments

Most instruments make sound when they vibrate or there is something hitting it. The vibrate instruments are like Guitar, Violin etc. but most of the instruments make the air around the object vibrate and the air around it also vibrate until it enters your ear. That is why you hear can hear instruments as sounds.

How can we separate vocal from instrumental?

How can we separate sound?

Source separation, blind signal separation (BSS) or blind source separation, is the separation of a set of source signals from a set of mixed signals, without the aid of information (or with very little information) about the source signals or the mixing process. It is most commonly applied in digital signal processing and involves the analysis of mixtures of signals; the objective is to recover the original component signals from a mixture signal.

Many music/voice separation methods typically first identify the vocal/non-vocal segments and then use a variety of techniques to separate the lead vocals from the background music. Musical pieces are often composed of repeating background on voice which does not show a regular repeating structure.

Study related solution/knowledge from the internet

From our research of related projects about separating instrumental from vocal. We found that most methods will be related to Sound Source Separation and Fast Fourier Transform (FFT). We choose to do this through Matlab by using Fast Fourier Transform to separate frequency range and keep choosing it until we can find the suitable range that can hear instrumental clearly. As we kept researching about this, this prove to be a challenge and complicated topic when we do the coding but it is possible to do it.

Method

We use any song that has the vocal and instrumental

Method of collecting the signals of interest

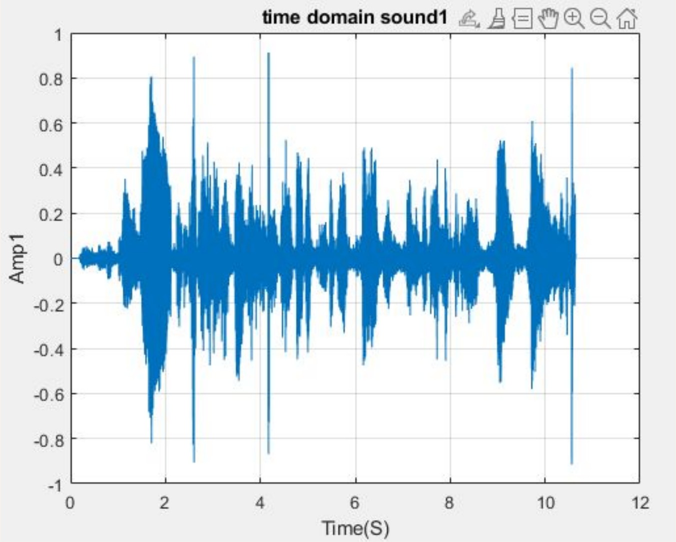
- We collected our samples from website or we recorded our voices while playing music.
- Choose some parts of song to analyze.
- We use MATLAB to do coding to separate instrumental from vocal.
    - We will use FFT to do the separation.
- We use MATLAB to analyze output data.

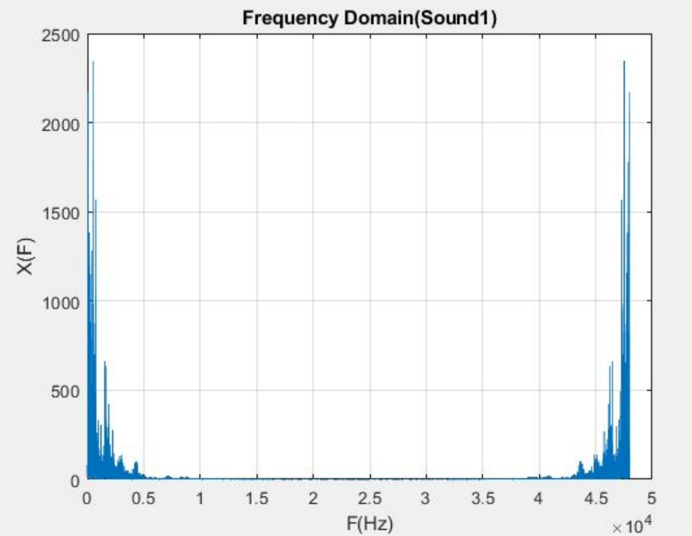
What is Fast Fourier Transformation (FFT)?

The "Fast Fourier Transform" (FFT) is an important measurement method in the science of audio and acoustics measurement. It converts a signal into individual spectral components and thereby provides frequency information about the signal.

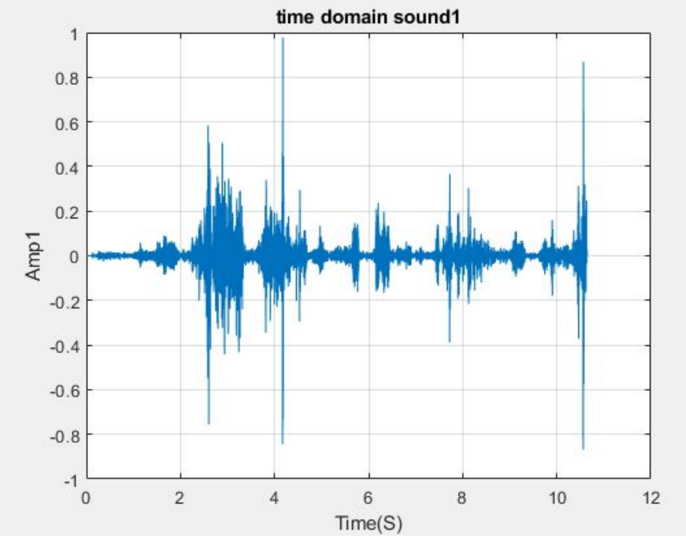
In other word, the FFT is an optimized algorithm for the implementation of the "Discrete Fourier Transformation" (DFT). A signal is sampled over a period of time and divided into its frequency components. These components are single sinusoidal at distinct frequencies each with their own amplitude and phase.

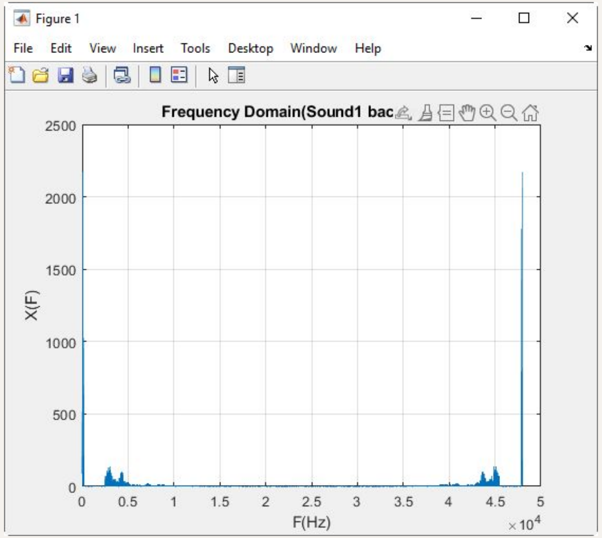
Sound1 (Original)



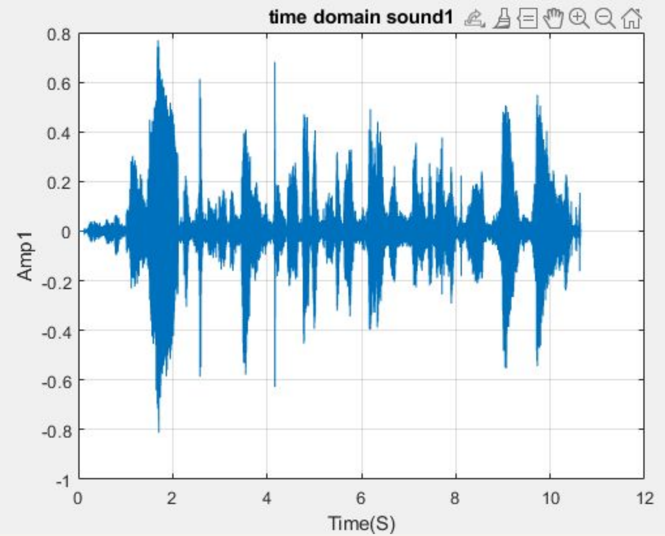


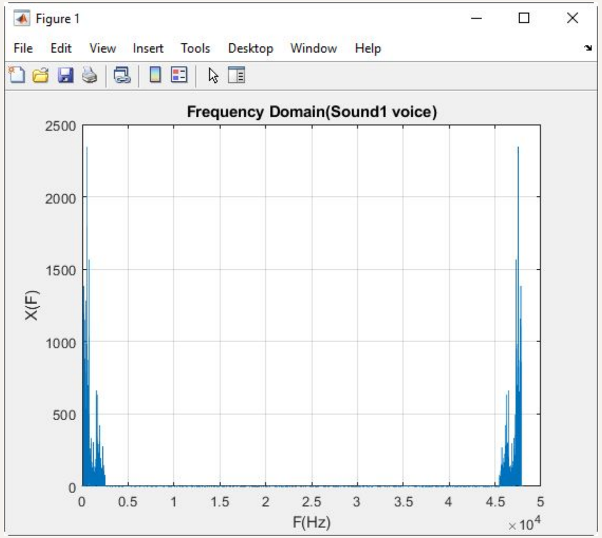
Sound_Background1



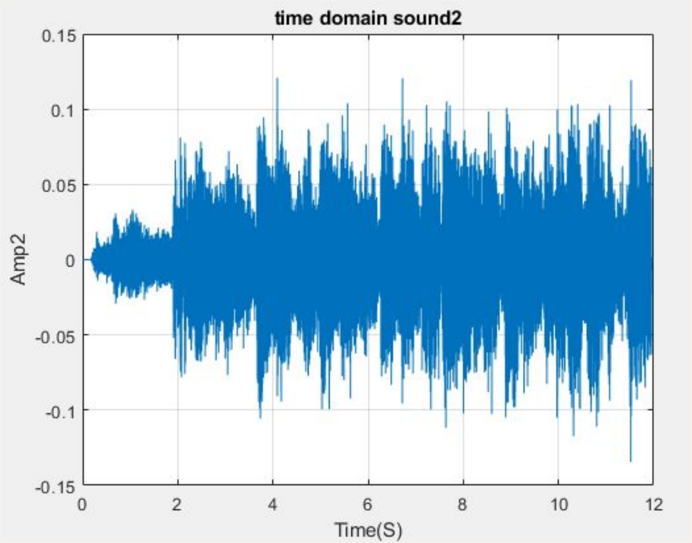


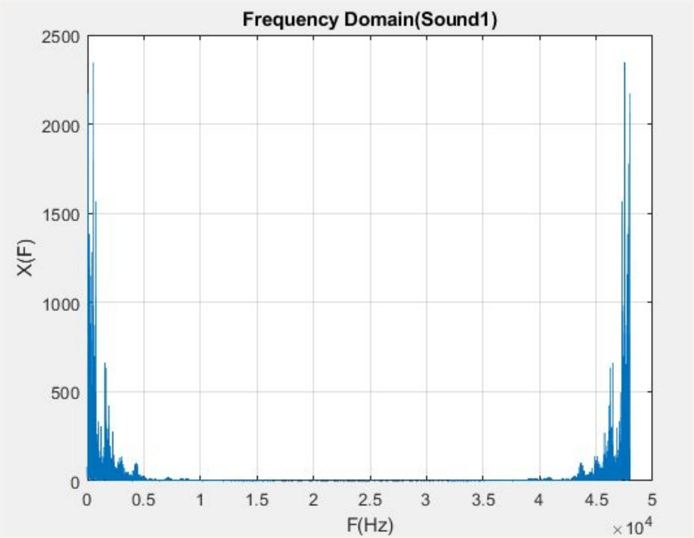
Sound_voice1



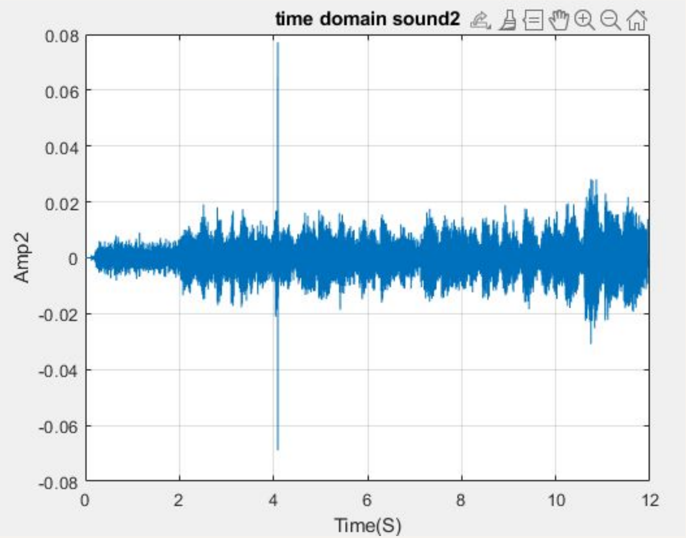


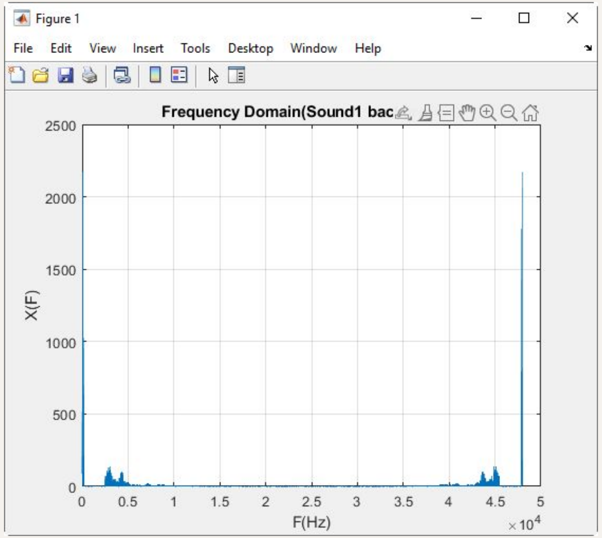
Sound2 (Original)



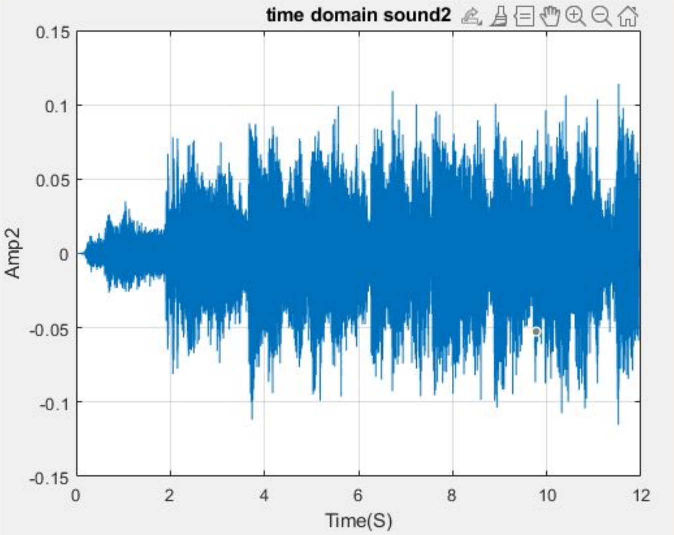


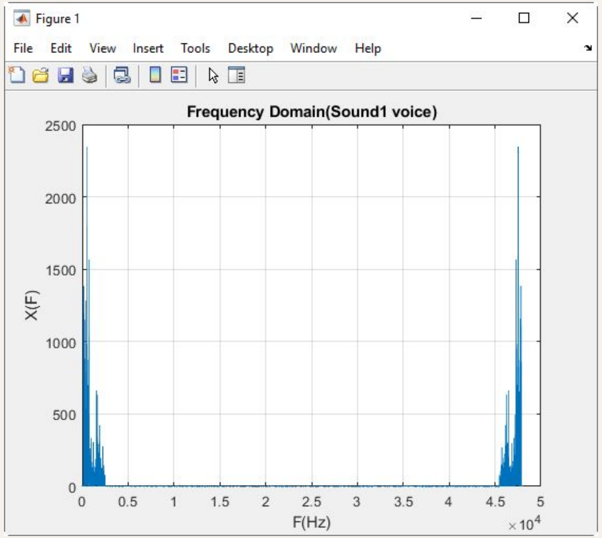
Sound_Background2





Sound_voice2





Code

```
[audio_in,audio_freq_sampl] = audioread('sound.wav');
length_audio = length(audio_in);
df = audio_freq_sampl/length_audio;
frequency_audio = -audio_freq_sampl/2:df:audio_freq_sampl/2-df;
FFT_audio_in = fftshift(fft(audio_in)/length(fft(audio_in)));
lower_threshold = 150;
upper_threshold = 2500;
val = abs(frequency_audio)<upper_threshold & abs(frequency_audio)>lower_threshold;
FFT_ins = FFT_audio_in(:,1);
```

```
FFT_voc = FFT_audio_in(:,1);
FFT_ins(val) = 0;
FFT_voc(~val) = 0;
FFT_a = ifftshift(FFT_audio_in);
```

Conclusion

Should we use this method to separate vocal and instrumental?

Ans: No because it is very impractical to use MATlab. You have to blindly choose frequency range. So you can pin point the range for both vocal and instrumental. The another reason is MATlab makes the audio quality drop by half, so it will sound very bad and it doesn't really separate 100%. It just trying to drown other part of sounds rather than separating it.

Reference

- https://www.nti-audio.com/en/support/know-how/fast-fourier-transform-fft
- https://pseeth.github.io/public/papers/seetharaman_2dft_waspaa2017.pdf
- Fast Fourier transform - MATLAB fft (mathworks.com)
- Signal separation - Wikipedia