# FFT transformation

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The basic premise of our project is that we are interested in figuring out why different instruments sound different, i.e. what makes instruments have different timbres.

Our process is to input some uncompressed audio file of an instrument playing at 440HZ (A4), process that file at a sampling rate of 44.1 kHz, run it through a self-implemented version of the Fast Fourier Transform, and look at the amplitude and frequncies of the resulting sine waves. At this stage, there are various things we can do with this output, some of which we will investigate.

Most use the Cooley-Tukey Fast Fourier Transform algorithm which is extremely optimized, however we decided to use a simpler algorithm that we found more understandable that utilizes recursion. As a result, our FFT algorithm is not very fast, but we think that it's more understandable and clearer to someone who has not had much experience with Discrete Fourier Transforms.

Demo