

Objective

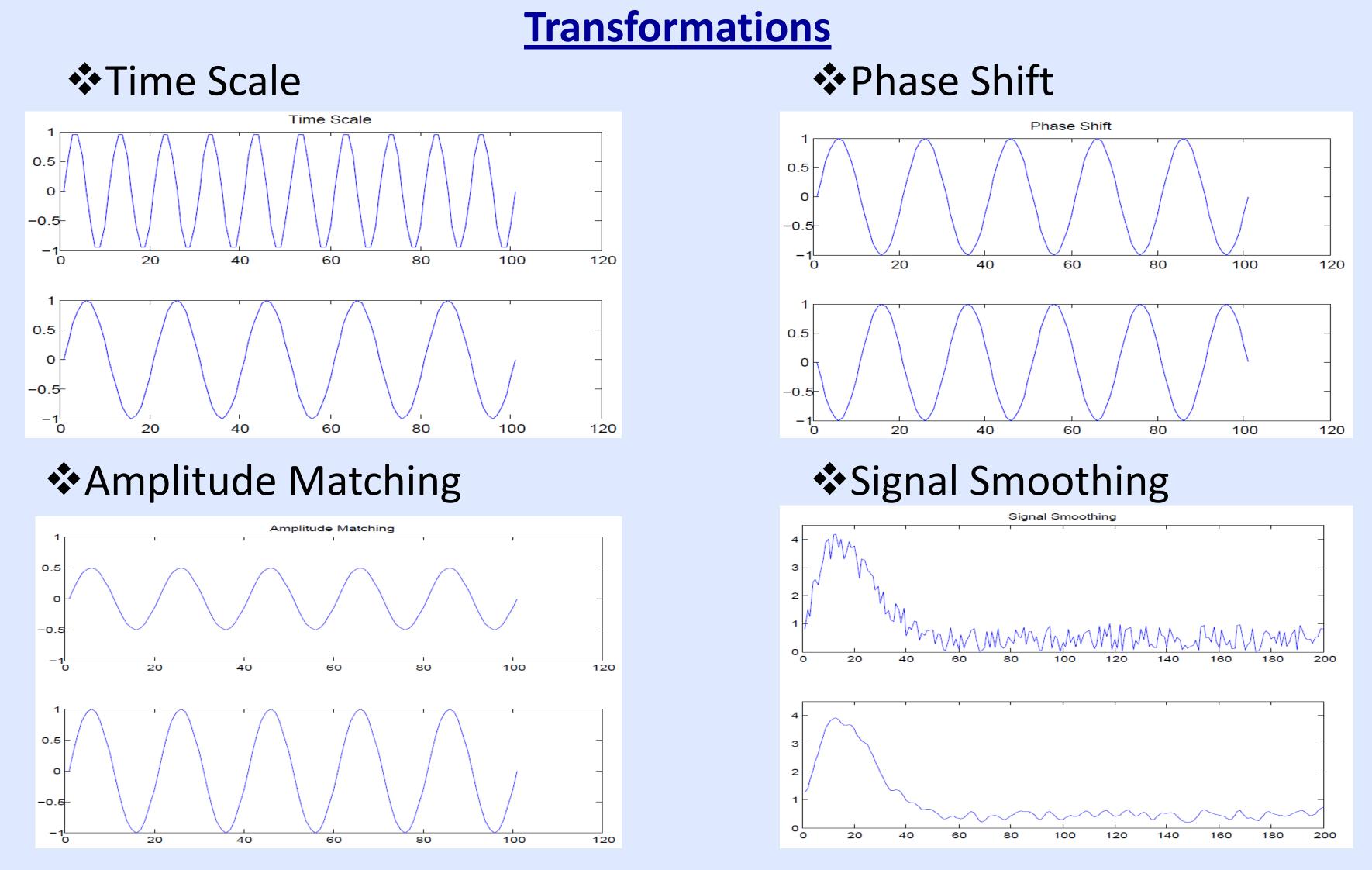
Create a system than takes an input song and separates its sources into different tracks while making the process as automatic and accurate as possible.

Background

Remixing and re-mastering almost always require the original separated audio tracks in a recording, but many times these are unavailable. This project attempts to solve the classical and notorious source separation problem in audio. The problems in separation arise from different instruments having overlapping frequencies in their sounds, but by using linearity we can decompose an audio track into a superposition of its parts.

Techniques Used

- Sample a note from the desired instrument to transform into the rest of the song
- Section the song into equal sections in order to extract the properties of an instrument from a section.
- *Match transformations of samples of the desired instrument playing in the song



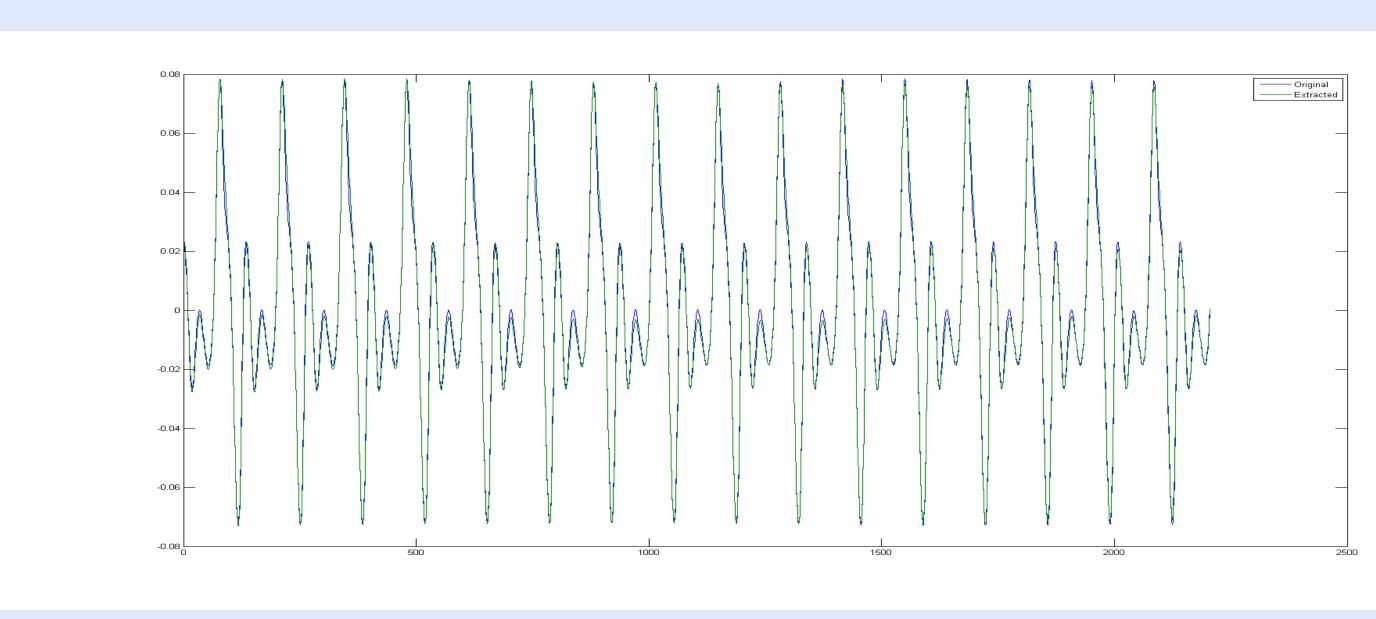
Digital Soloer

Iman Aboutaleb, Ahmad Aldabbagh, Arya Bandari, Mohammed Sarraj, Giovanni Zhang **EECS 452 - Winter 2012**

Hardware Used







In this case and for many with simple instruments (no effects) that play notes (single or chords) on beat, the extraction is almost identical to the original.

Future Work

We would like to thank Professor Alfred Hero, Dr. Kurt Metzger, Mr. Chao Yuan and Mr. Charlie Yan for their efforts and support.



TMS320C5515 DSP Evaluation Module

Features:

- Low power 16-bit processors to conserve energy
- 240 MIPS performance
- ✤320KB on-chip memory
- Hardware accelerator for FFT computation

Adaptive note lengths versus sectioning

Envelope detection versus constant amplitude matching Recursive iterations of extraction after some instruments have been removed to minimize residual error

Analyzing and simulating different effects such as reverb, delay, filter cutoffs, etc.

Acknowledgments

