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Final Project



Report Structure

- Introduction: A general description of the area of your project and why you're doing it.
- **Problem Specification:** A clear and succinct technical description of the problem you're addressing. Formulating a general problem (e.g., transcribing music) into a well-defined technical goal (e.g., reporting a list of estimated fundamental periods at each time frame) is often the most important part of a project.
- **Data:** What are the real-world and/or synthetic signals you are going to use to develop and evaluate your work?
- Evaluation Criteria: How are you going to measure how well your project performs? The best criteria are objective, quantitative, and discriminatory. You want to be able to demonstrate and measure improvements in your system.
- **Approach:** A description of how you went about trying to solve the problem. Sometimes you can make a nice project by contrasting two or more different approaches.



Report Structure

- **Results and Analysis:** What happened when you evaluated your system using the data and criteria introduced above? What were the principal shorfalls? (This may require you to choose or synthesize data that will reveal these shortcomings.) Your analysis of what happened is one of the most important opportunities to display your command of signal processing concepts.
- **Development:** If possible, you will come up with ideas about how to improve the shortcomings identified in the previous section, and then implement and evaluate them. Did they, in fact, help? Were there unexpected side-effects?
- Conclusions: What did you learn from doing the project? What did you demonstrate about how to solve your problem?
- References: Complete list of sources you used in completing your project, with explanations of what you got from each.



- **DTMF decoder** convert a recording of 'touch tones' from a real telephone into the corresponding set of digits.
- Channel equalization. This is a classic signal processing problem, where the signal has been subject to some unknown filter, and the goal is to infer what that filter was, then invert its effects.
- **Signal denoising**. Signals get corrupted by noise all kinds of ways by electrical interference, by mechanical damage of recording media, or simply because there were unwanted sounds present during the original recording.
- Speech endpointing find the beginning and end of each speech phrase or utterance in a recording (which may include background noise)



- Speech/music discrimination classify example fragments as speech, music, or some other class
- **Pitch extraction**. This is a widespread problem in speech and music processing: identifying the local periodicity of a sound with a perceived pitch. Autocorrelation is the classic method, but it makes a lot of common errors, so there are many approaches to improving it.
- Modeling musical instruments. It turns out that many musical instruments can be modeled with surprisingly good quality by relatively simple signal processing networks.

- **Timescale modification**. In the very first lecture I played an example of speech that had been slowed down *without* lowering the pitch.
- Sound visualization. We've seen the spectrogram as an example of rendering a sound as an image. However, there are very many parameters to vary, with pros and cons to each variation. This project would choose a particular goal, say a clear of a certain kind of sound in a variety of backgrounds, then investigate the best possible processing to facilitate that display.

Steganography/watermarking. There are various motivations for 'hiding' data in a soundfile without making the alteration audible. One is watermarking, so that a sound can be recognized as 'valid' without knowing its content ahead of time. Another is embedding copyright markers that cannot easily be removed by counterfeiters. This project will investigate some mechanisms for encoding data in sound, and examine the limits of how much data can be included, what degradation to sound quality is entailed, and how hard it is to remove, or to simulate, the marking.

- Artificial reverberation. I mentioned that one use of allpass filters is in the simulation of room reverberation. This project will build a simulated room reverberator and investigate several enhancements to increase the realism. There are several good papers to start from.
- Compression. Audio signal compression is a current hot topic. This project would involve implementing one or more simple compression schemes, and investigating how they perform for different kinds of signals, as well as the kinds of distortion they introduce.

- Time-delay angle-of-arrival estimation. We use our two ears to be able to detect the direction from which a sound arrives. The strongest cue is probably the slight time differences that occur due to the finite speed of sound traveling to each side. Cross-correlation can reveal this time difference, and indicate the azimuth from which sounds occur.
- **Doubletalk detection**. In speech processing we frequently assume that the signal contains just a single voice; in many cases this is not true, for instance when people interrupt one another on a telephone call or in a meeting. Separating these voices is hard, but we would like at least to be able to detect when it is happening, so we know not to attempt normal processing.

- Synthesizing 3D sound. Since we have some understanding of how the ear uses binaural (stereo) cues to infer the direction of different sound sources, we should be able to construct artificial sounds including those cues that will appear to come from particular directions
- Cross-synthesis. An interesting effect in electronic music synthesis is to somehow 'combine' two sounds into a single sound that appears to have properties of both sources. This project will implement some variants of how this can be done.





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