

# Speech Playback Using PIC32

Anthony Linley (arl228@cornell.edu) Advisor: Dr. Bruce Land

## Process Overview

Floating point audio sample

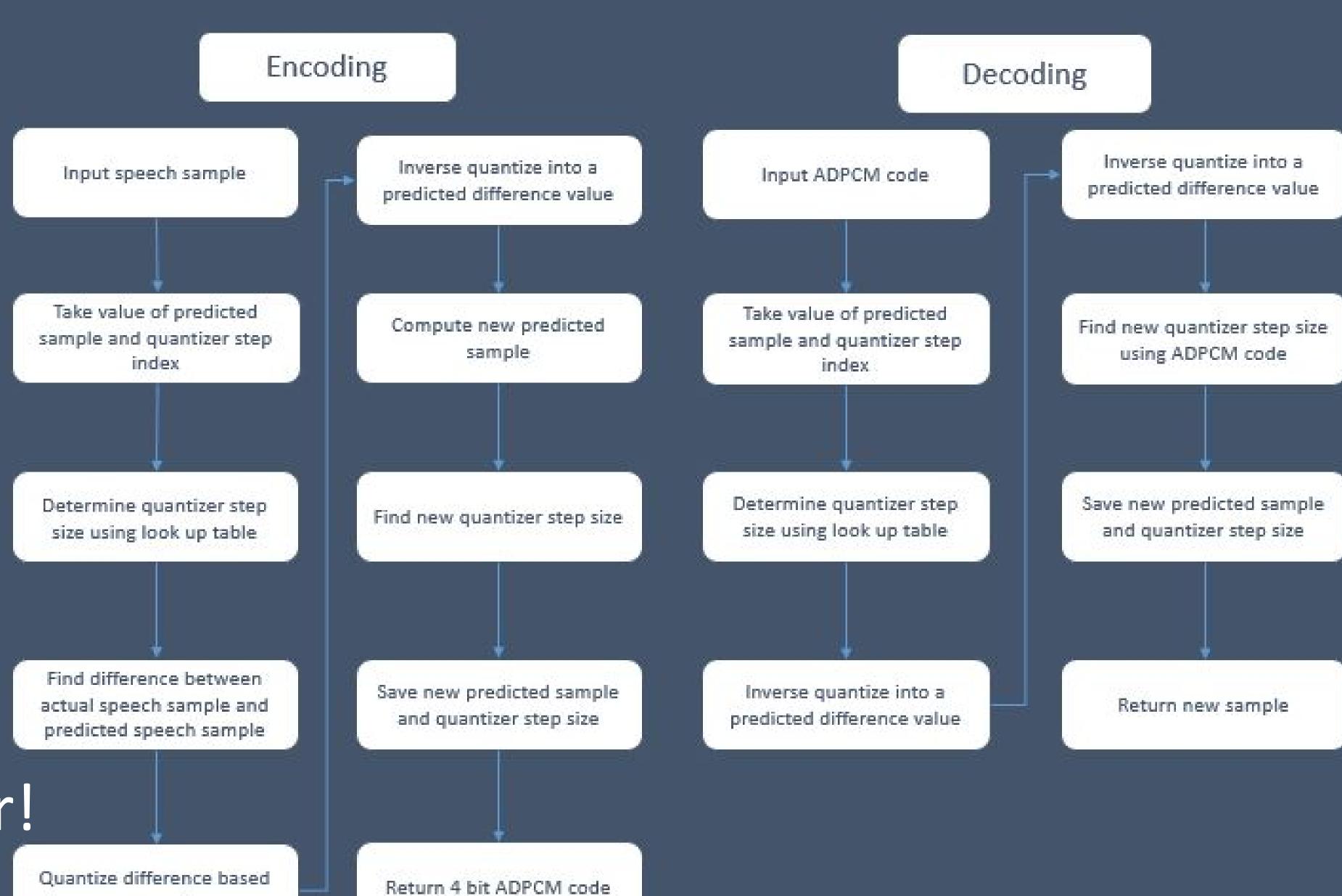
Encode Sample

Decode Sample

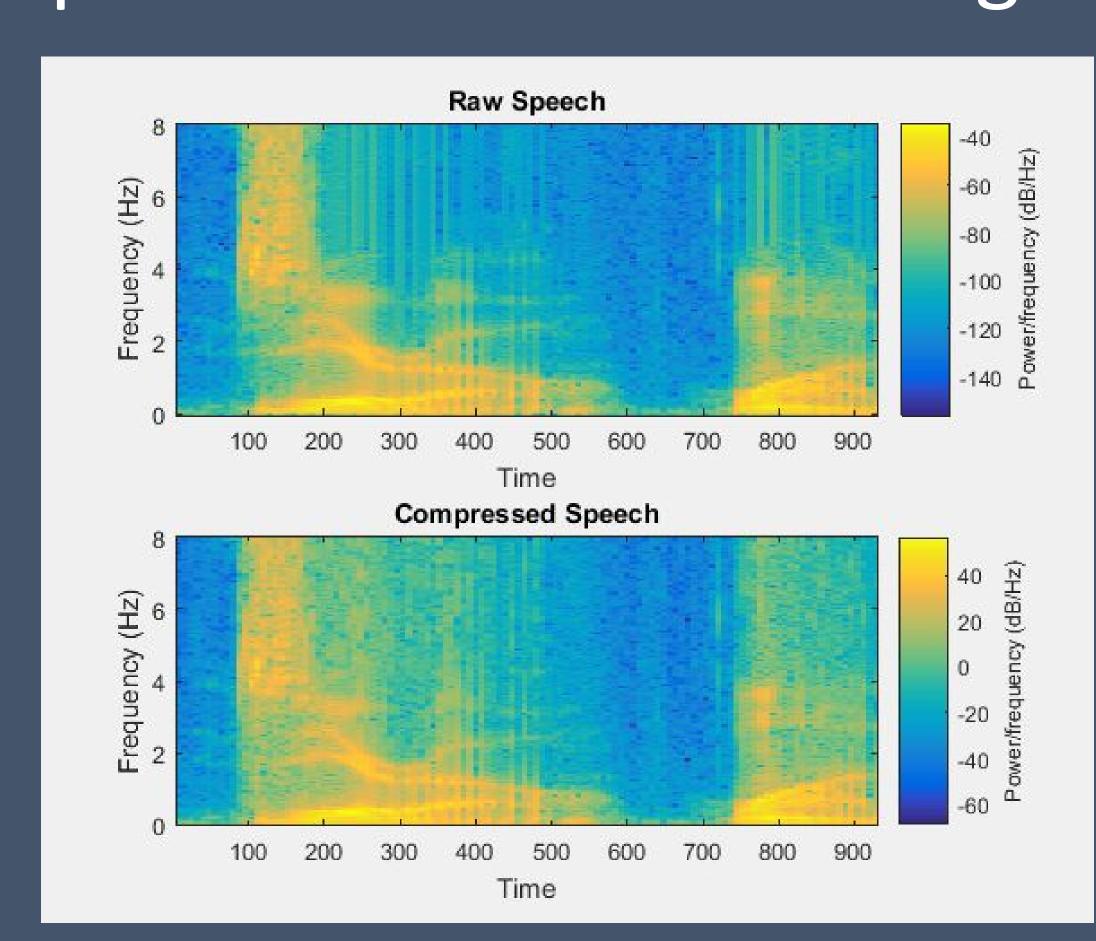
Output Sample through DAC

#### Sound Waves are BIG!

The goal of this project was to implement adaptive differential pulse code modulation, or ADPCM, on a PIC32 microcontroller in an attempt to process and play back speech audio data. The implementation used is based off Microchip's simplified ADPCM algorithm.



## Spectral Content Unchanged



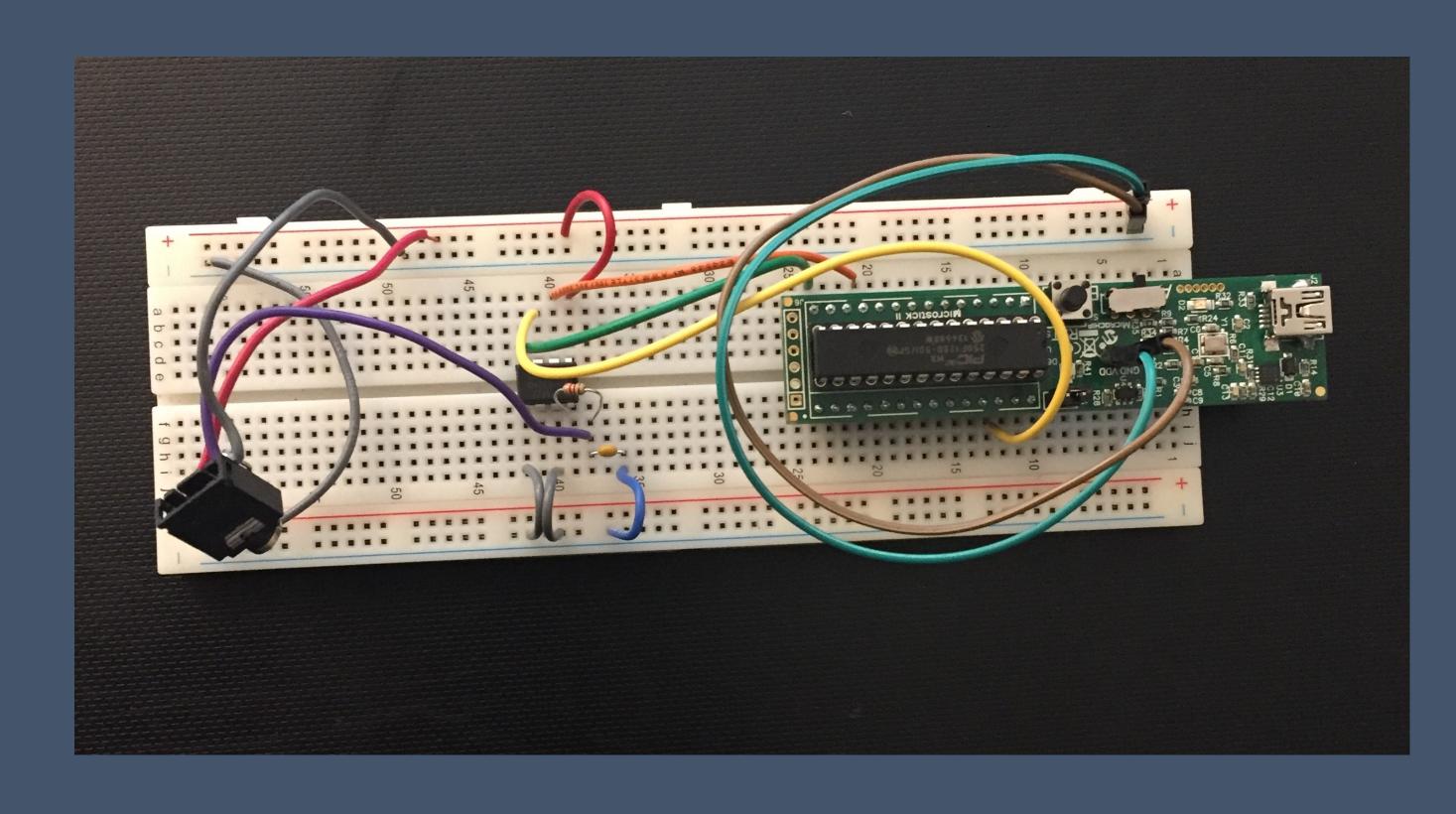
The goal of this project was to see if the PIC32 was capable of implementing speech playback utilizing ADPCM. The PIC32 was successful in being able to play back the audio based on raw numerical data from an audio file utilizing this signal processing technique. The entire encoding and decoding process took between 540-581 cycles to complete, which used about a 1/5 of the CPU.

# But They Can Be Smaller!

Quantize difference based on step size

ADPCM is a signal encoding process that takes audio data in and produces digital signals as an output. By only recording the differences between a sample and a predicted sample, the predictor can adjust itself appropriately which allows for signals to be produced at lower bit rates than when utilizing standard pulse code modulation.

## System Setup



## References

[1] Richey, Rodger Adaptive Differential Pulse Code Modulation using PICmicro™ Microcontrollers 2002