RIBBIT: A Cost-Effective iOS Hearing Aid App

TCU College of Science & Engineering
reSEaRCh Science and Engineering Research Center

Authors: Duy Dang, Robert Kern, Esteban Kleckner

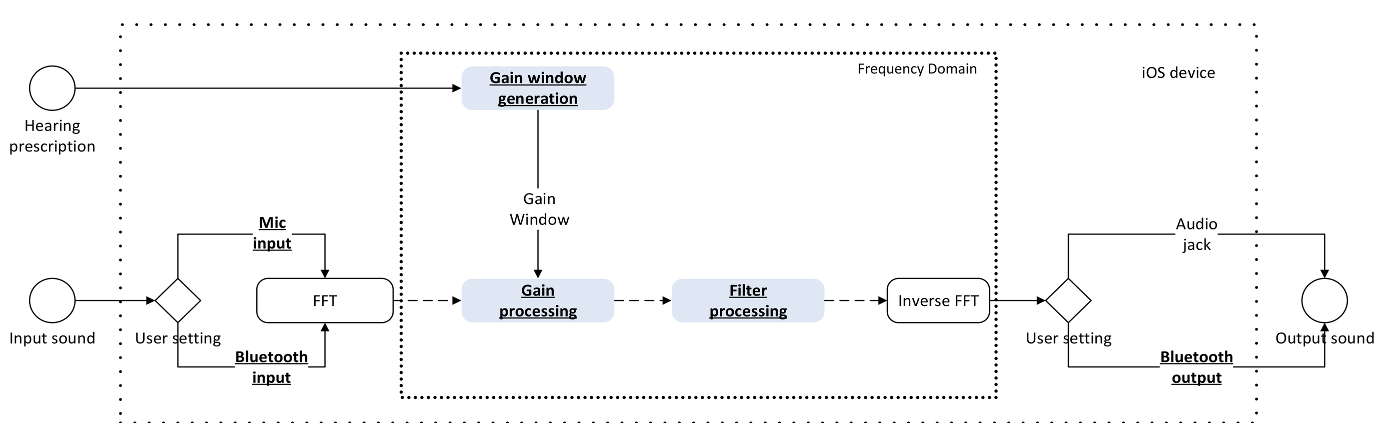
Faculty Advisors: Liran Ma Ph.D. (CS) and Donnell Payne Ph.D. (CS)

Background

Motivation:
- To provide a solution for hearing-impaired individuals who cannot afford a traditional hearing aid

Goals:
- Design and implement an affordable, and user-friendly hearing aid on Smart Devices, such as an iPhone

Ribbit Architecture



- Two inputs to the iOS app: sound and hearing prescription
- Sound transformed from the time domain to the frequency domain with Fast Fourier Transform (FFT)
- Based on the prescription, appropriate gain and filtering algorithms are applied in the frequency domain
- Sound transformed back to the time domain with the Inverse FFT and output

QR-Based Prescription Generation

Online hearing prescription to QR-code converter

Prescription Data

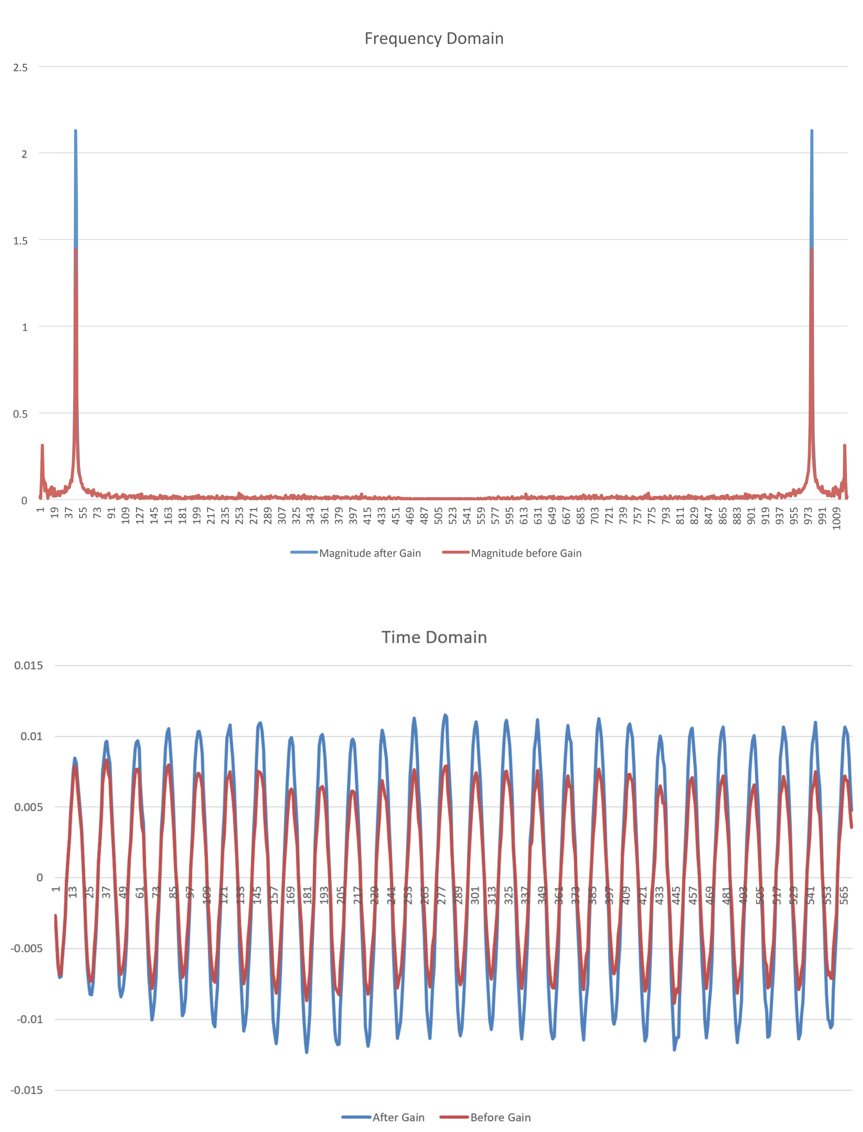
Mode	Indoors
Patient	Bob Marley
Prescriber	Frank Sinatra
Date	3/30/16
Debug	True

	Left Ear	Right Ear
[125]	5	5
[250]	20	15
[500]	20	20
[1000]	20	25
[2000]	36	40
[4000]	70	65
[8000]	80	75

Generate QR

- A convenient, secure method to load prescription
- QR code generation is done in a browser
- Read in by the phone, using the camera

Gain Processing

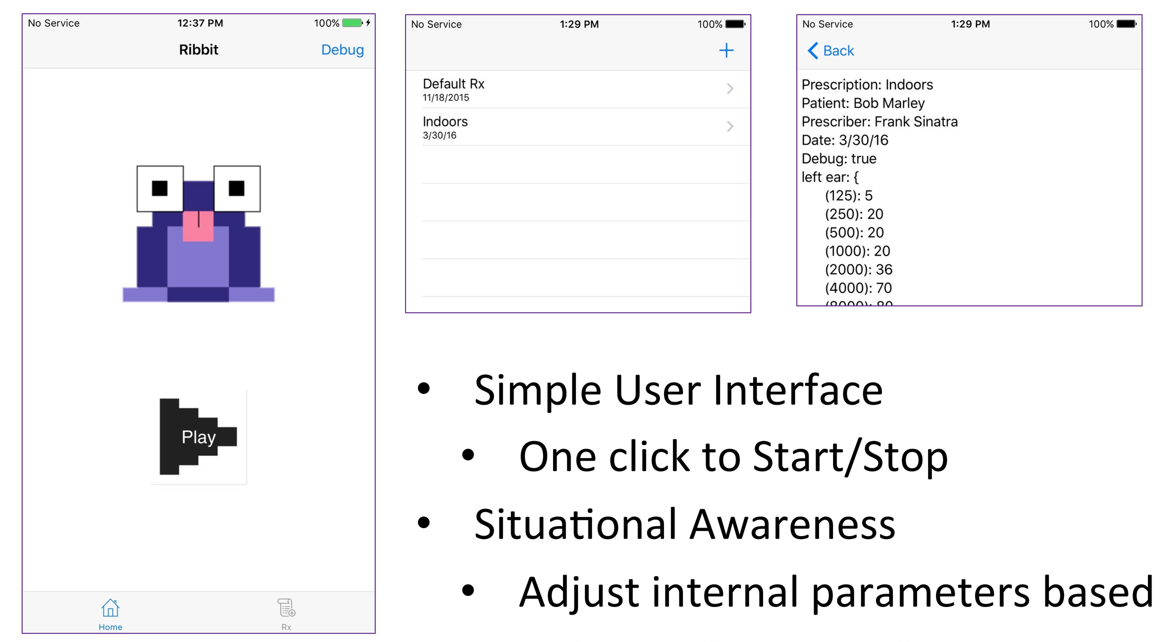


- Hearing-impairment- loss of the ability to hear/detect some (usually higher) frequencies at the normal loudness
- Gain requirements:
  - Boost a certain frequency by a specific loudness level (dB)
  - Calculate the dB values to increase each individual frequency
- Gain processing steps:
  - Compare the prescription of a normal-hearing person to that of the user
  - Generate a gain window based on the above comparison
  - Multiply the signal buffer by the gain window

Filter Processing

- Filtering the input sound removes noise and sound artifacts that otherwise impact the user's ability to fully understand/comprehend speech
- The filtering process utilizes the vDSP framework; Apple's Digital Signal Processing library. Specifically, the Fast Fourier Transform (FFT)
- The FFT is used to break the signal into its constituent parts, then apply a filter window and recombine these parts into a coherent signal
- Left top: An example filter window designed to reduce amplitudes between 4 and 24 kHz—a range that doesn't provide information necessary for understanding speech
- Left middle: An example of frequencies that would be removed from an input signal
- Left bottom: The differences between the input and output signal

The Ribbit Interface



- Simple User Interface
  - One click to Start/Stop
- Situational Awareness
  - Adjust internal parameters based on different "scenarios"

Conclusions and Future Work

- Using a Smart Device as a hearing aid shows promise
  - The app is able to finish processing sound samples within 22ms
  - Sound quality achieves acceptable levels
  - By tailoring the application to the user, we offer a more personal experience
- Future enhancements would include:
  - The elimination of sound artifacts and the inclusion of smart technologies, such as location sensing
  - The opportunity to test our application in a setting where human trials are possible

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