

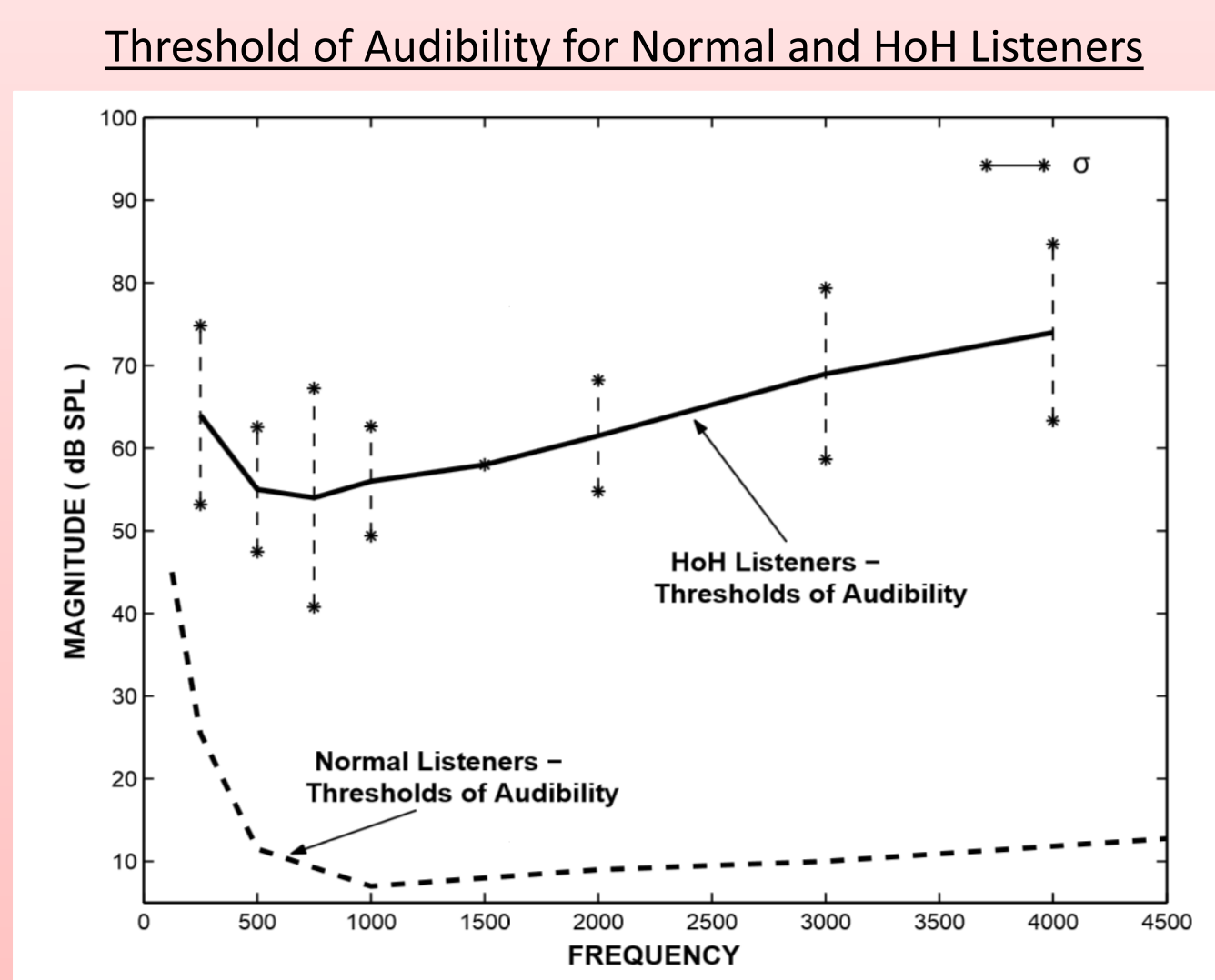
# Real-Time Hardware Implementation of Telephone Speech Enhancement Algorithm

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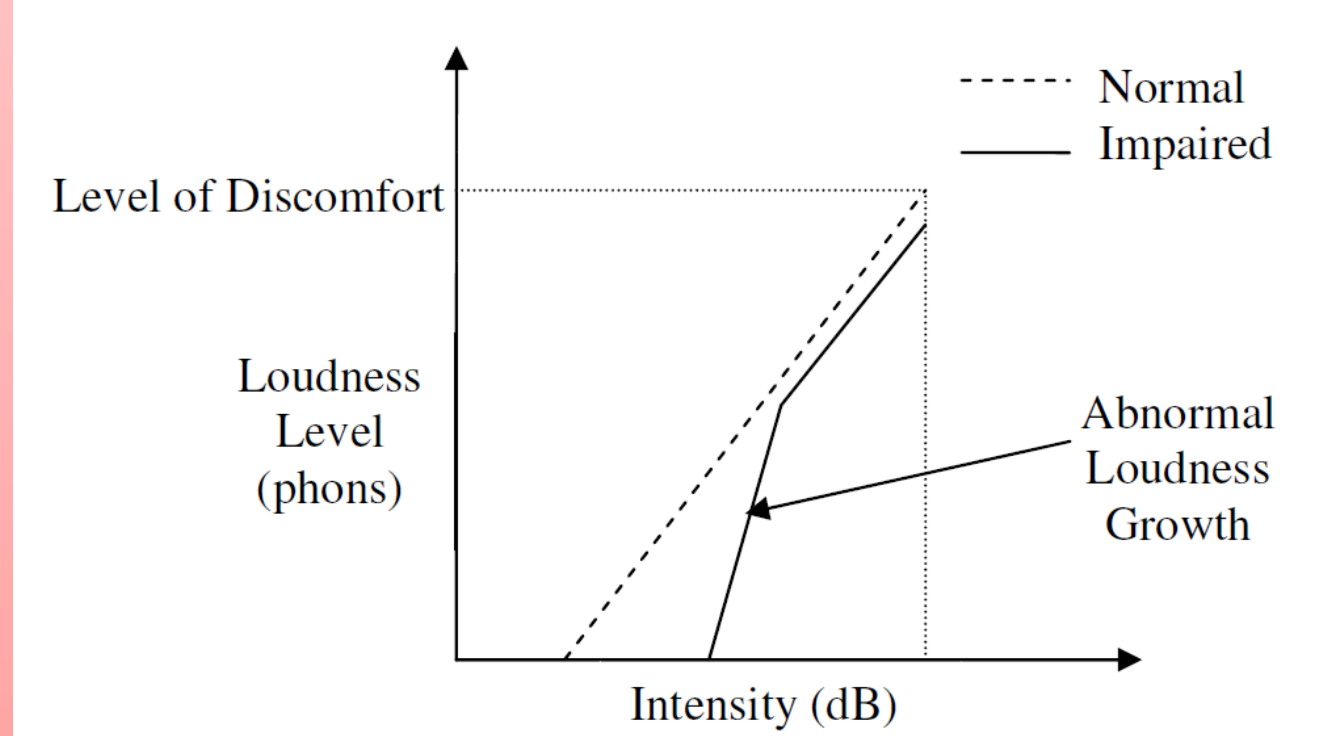
The Ohio State University

## The Problem

- Hard of Hearing (HoH) people often struggle with telephone communication
- The Question: How can telephone communication be improved for the HoH?
- Hearing aids can be uncomfortable during telephone use
- The threshold of hearing for the HoH is much higher than normal which results in reduced dynamic range
- Reduced dynamic range creates abnormal loudness growth in HoH people which leads to discomfort and even pain

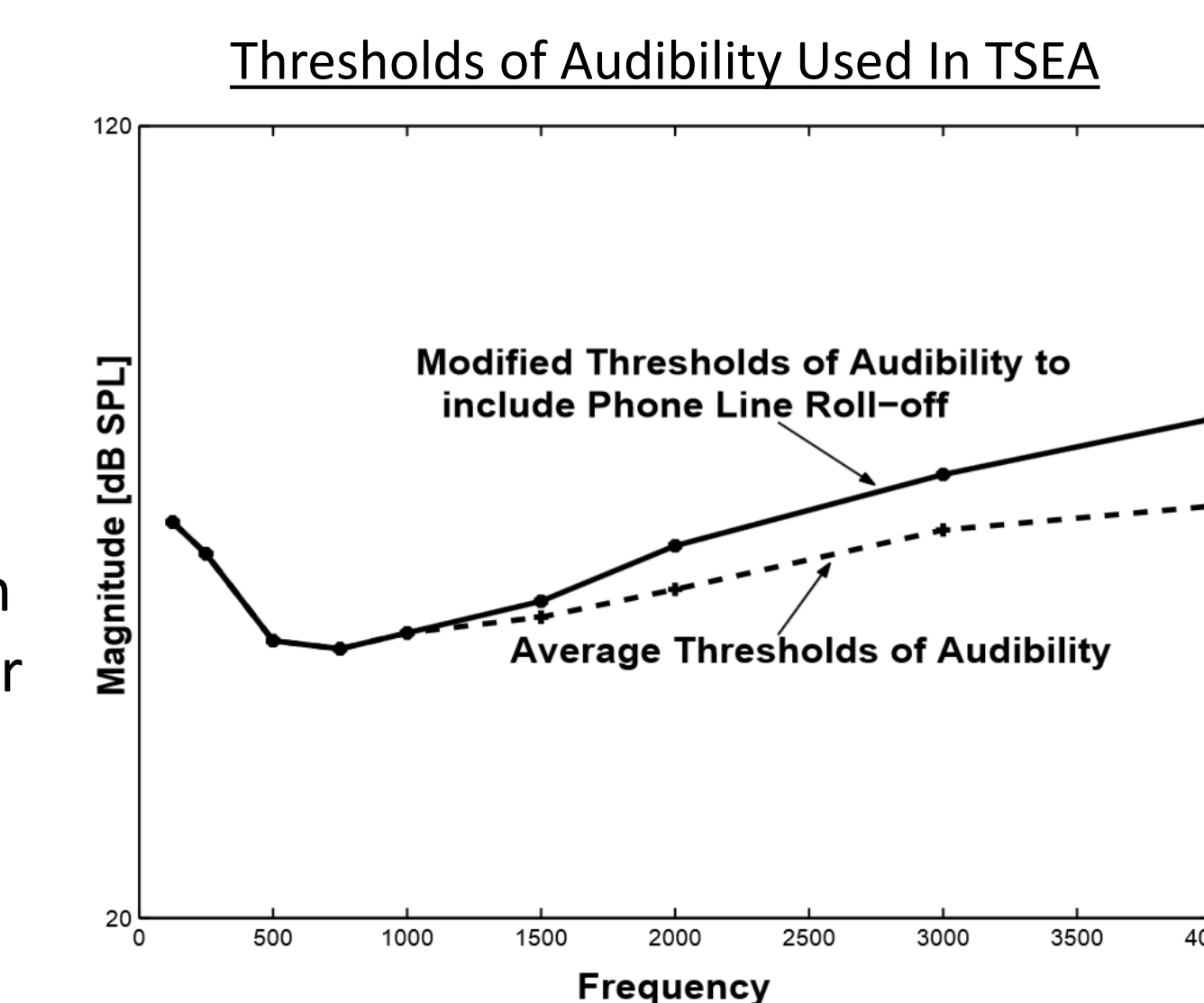


Reduced Dynamic Range Leading to Abnormal Loudness Growth

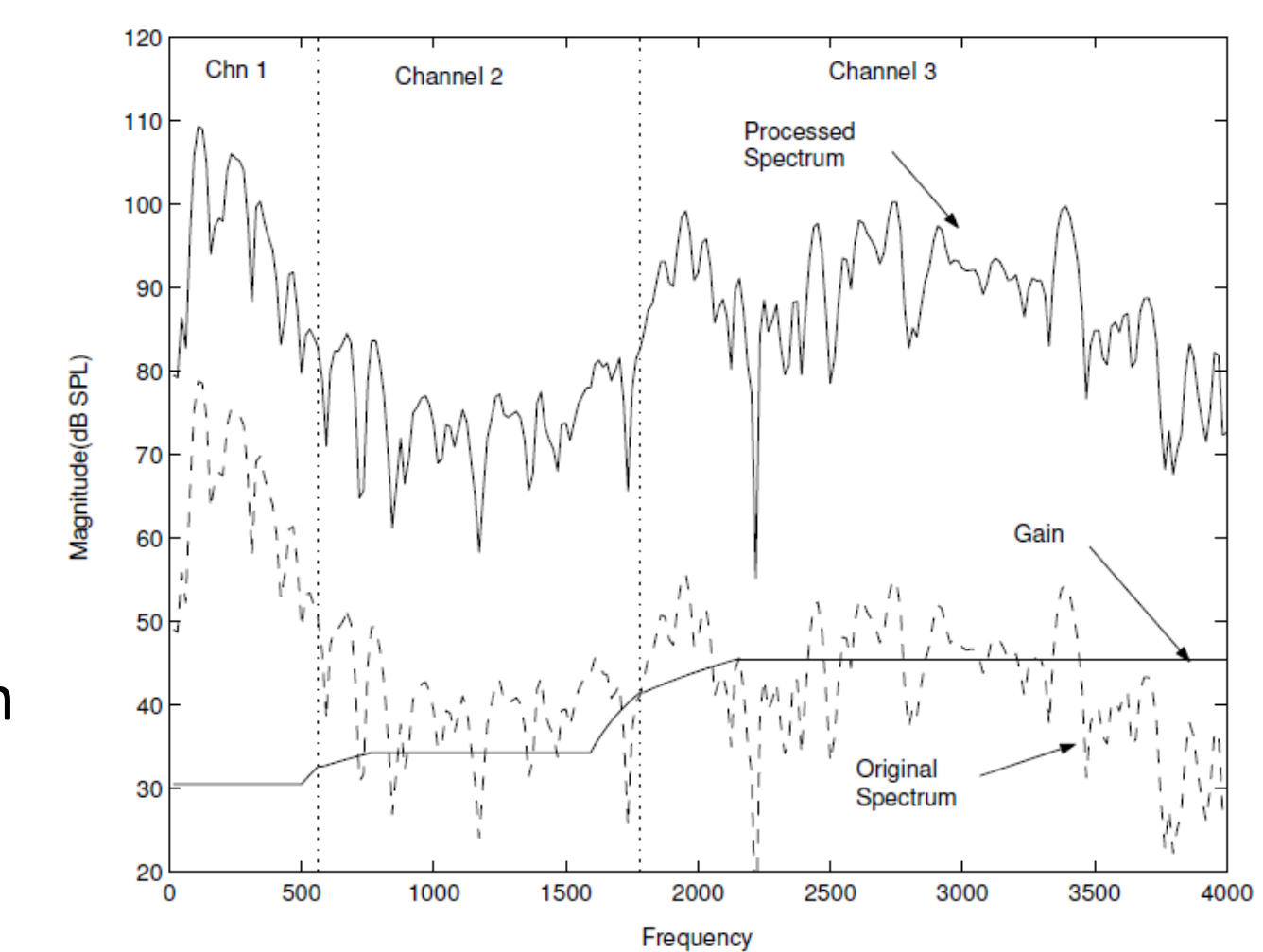


## The Solution

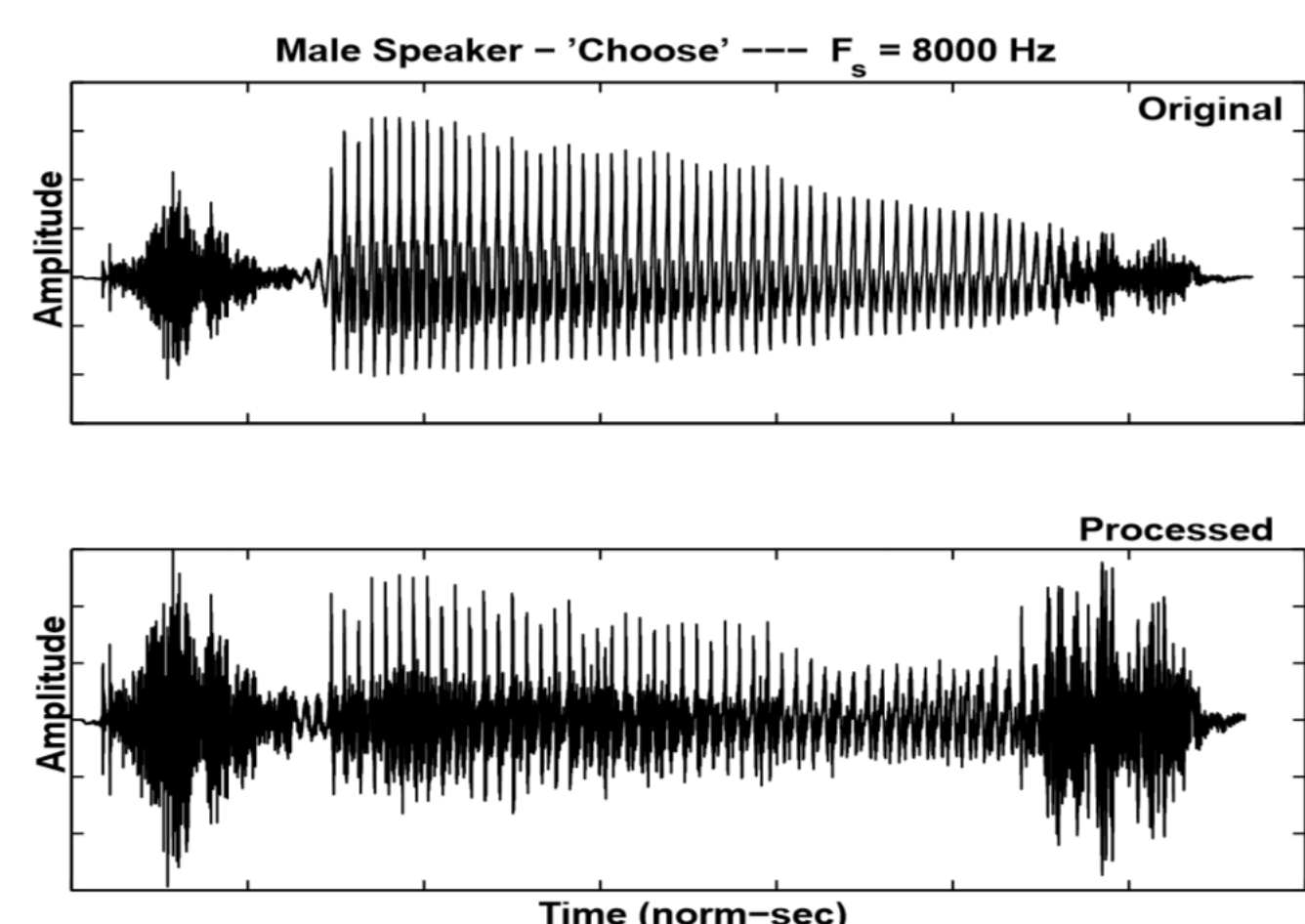
- The Telephone Speech Enhancement Algorithm (TSEA) developed by Natarajan
- This solution is unique in the fact that it preprocesses speech signals before transmission over the phone line
- The goal is to process speech signals so they lie within the HoH listener's dynamic range
- TSEA is a three-channel compression algorithm
- Dynamically varies channel boundaries based on formant frequency peaks
- Boosts higher frequencies important to speech perception that are limited by HoH audibility thresholds and reduced phone line bandwidth
- Gain calculations based on the dynamic range ratios of normal and HoH listeners
- Low amplitude consonant sounds are amplified while higher amplitude vowel sounds remain comfortably loud
- TSEA proven effective at improving speech perception



Sample Processed Signal Spectrum and Applied Gain



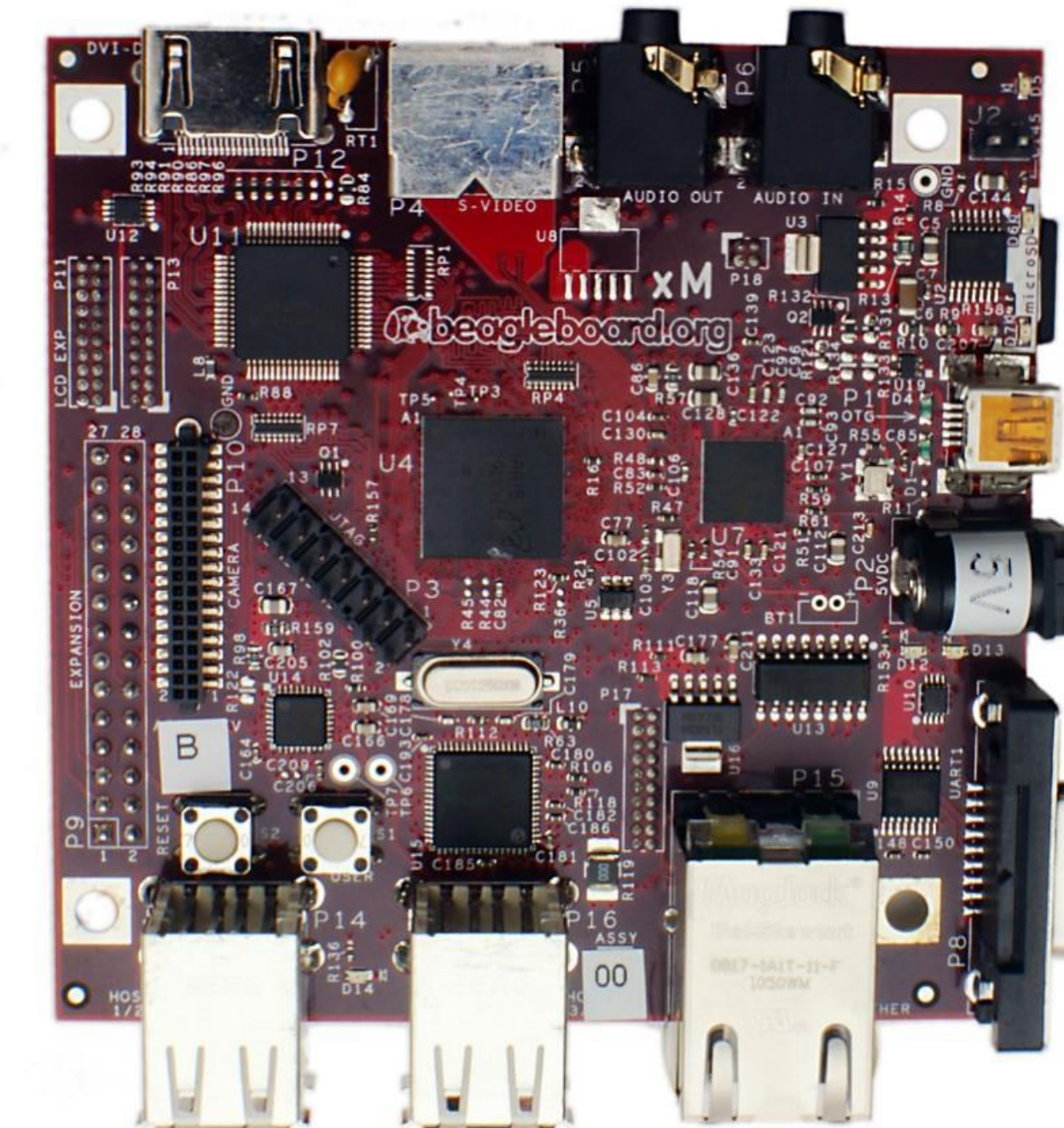
Processed Waveform Showing Consonant Boost



## The Hardware Implementation

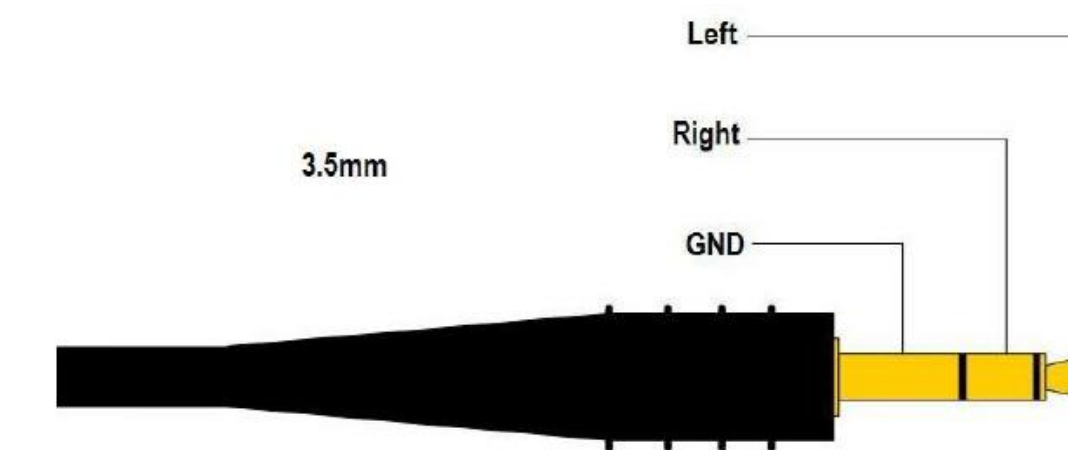
- This project implements the TSEA on the BeagleBoard-xM
- Audio input and output connectors on the BeagleBoard are used to transfer signals to and from the board
- Speech signals are sent into the board and processed using an adapted version of the original Simulink TSEA model developed by Kommatil
- Processed speech output is recorded via the output jack connector and used for analysis
- Now three versions of TSEA exist: the original MATLAB model, the Simulink model, and the BeagleBoard model

The BeagleBoard-xM Development Board

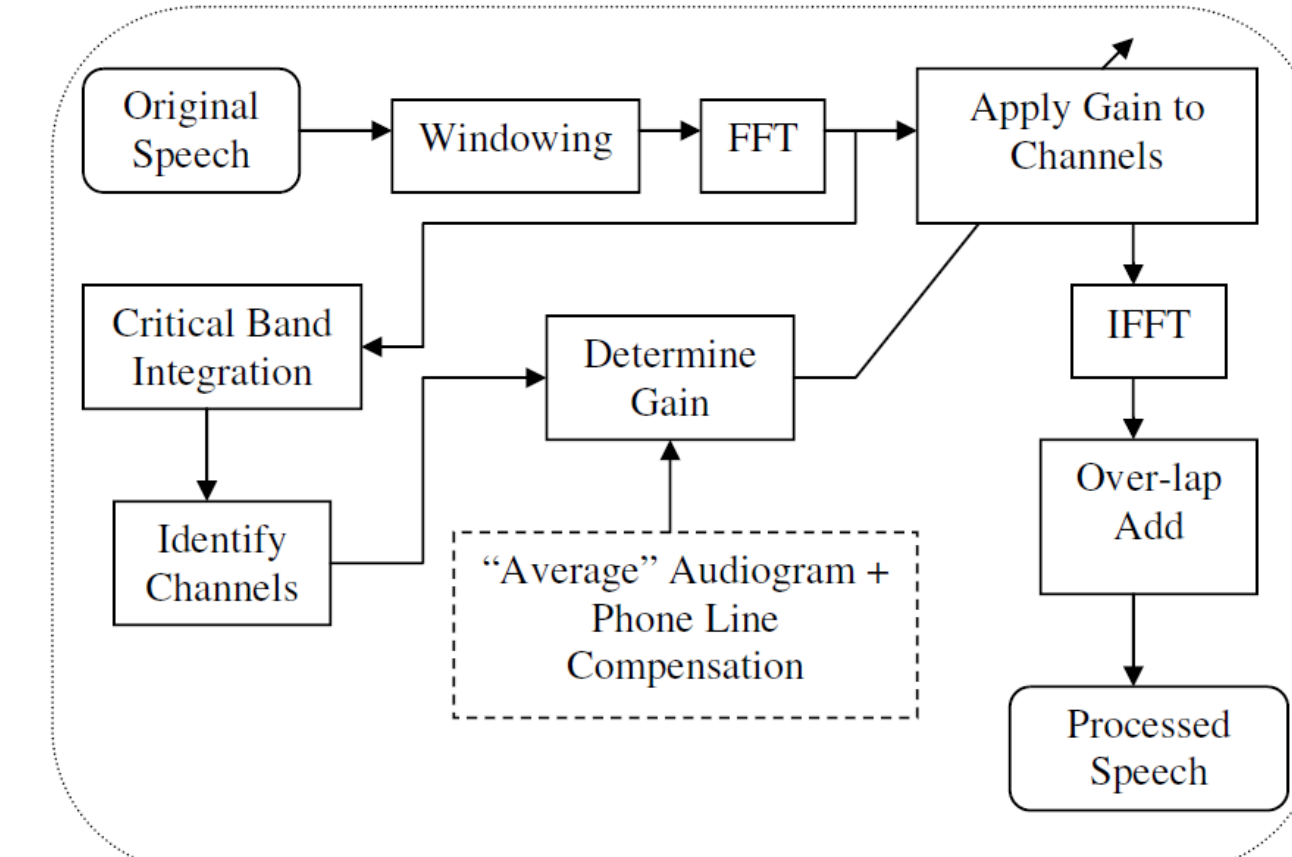


- A 1 GHz Texas Instruments processor provides the power to run the algorithm
- The BeagleBoard-xM has an onboard audio codec for audio processing at all standard sampling rates
- Stereo audio is converted to mono to simulate telephone speech signals

3.5-mm Audio Jack Connectors Used



Block Diagram of the TSEA Algorithm Used

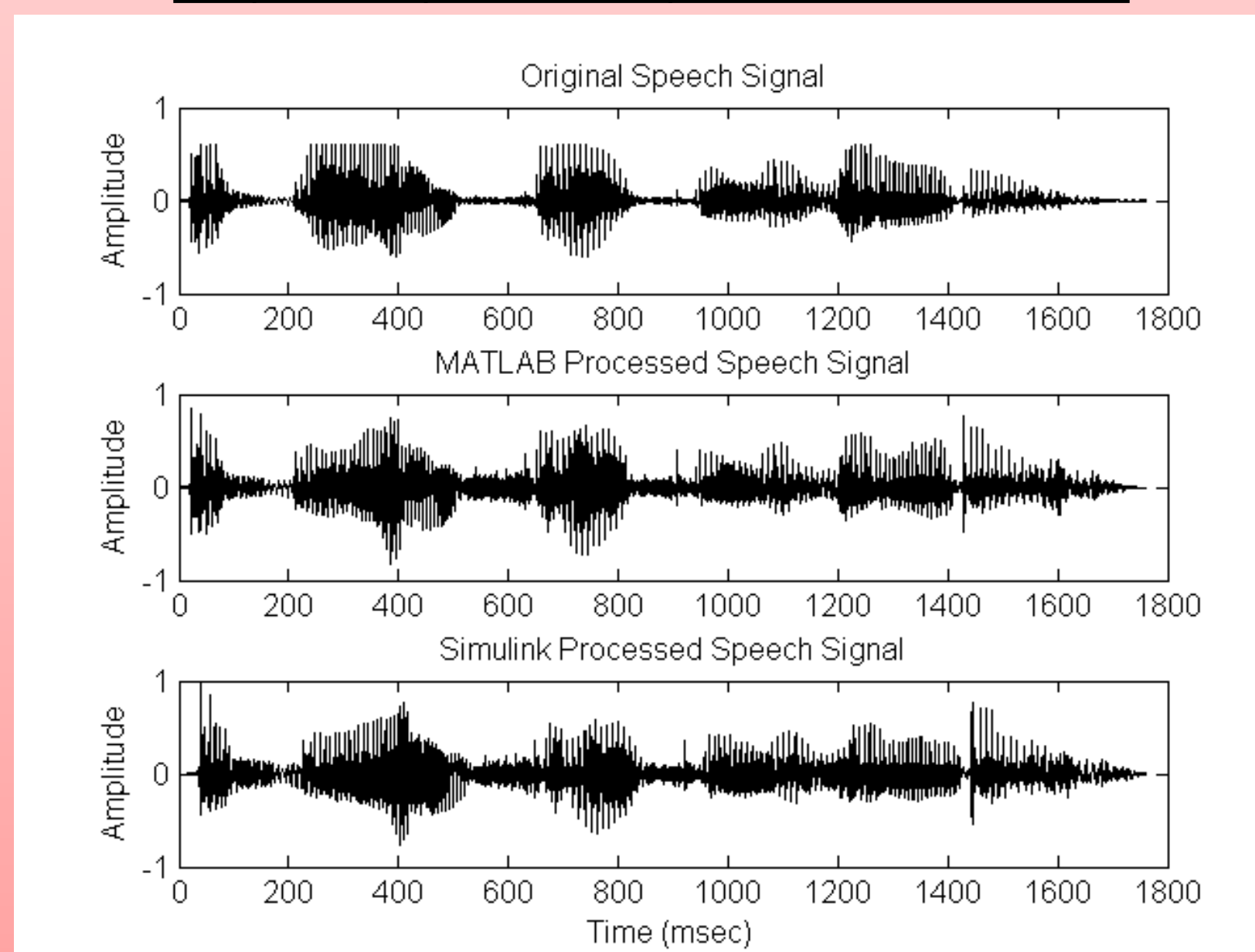


- The BeagleBoard model introduces internal audio artifacts into the signal
- Testing and analysis was performed to determine exactly what effects the BeagleBoard has on the algorithm

## Preliminary Testing

- The old original Simulink model was updated to the newest version of Simulink and compared to the original MATLAB version of TSEA
- Error calculations yielded an average of 2.42% RMS error and 13.51% maximum error between the MATLAB and Simulink versions of TSEA

Output Comparison of Updated Simulink Model



## Testing and Results

- Output of the BeagleBoard TSEA signals were compared to the Simulink version of TSEA
- Six speech signals (sampled at 8kHz) were processed to find an average error
- Errors caused due to analog filtering of the audio jacks and internal noise of the board
- The RMS error between the BeagleBoard and Simulink model can largely be attributed to the error from the internal noise and analog connection filters

Error Between BeagleBoard and Simulink Model Output

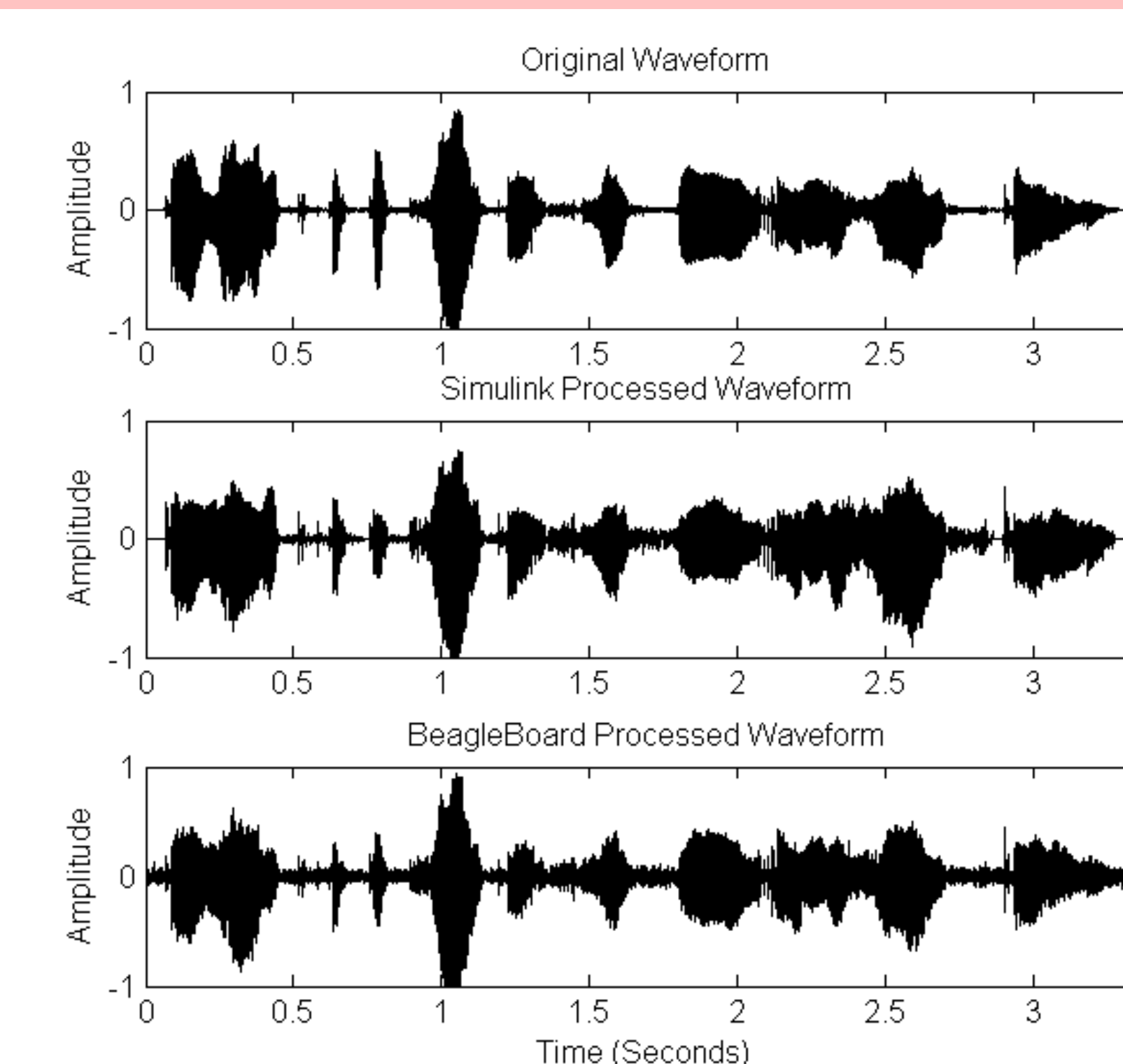
Speech Signal Test	RMS Error	Maximum Error
Test 1	4.93%	28.68%
Test 2	4.13%	24.00%
Test 3	4.24%	37.58%
Test 4	4.38%	29.44%
Test 5	4.74%	29.48%
Test 6	3.74%	27.24%
<b>Average</b>	<b>4.36%</b>	<b>29.40%</b>

Error From Internal Noise and Analog Connections

Sinusoid Frequency	RMS Error	Maximum Error
100 Hz	1.34%	4.08%
500 Hz	2.87%	6.44%
1000 Hz	4.50%	8.88%
2000 Hz	2.71%	7.42%
3500 Hz	3.66%	12.02%
<b>Average</b>	<b>3.02%</b>	<b>7.77%</b>

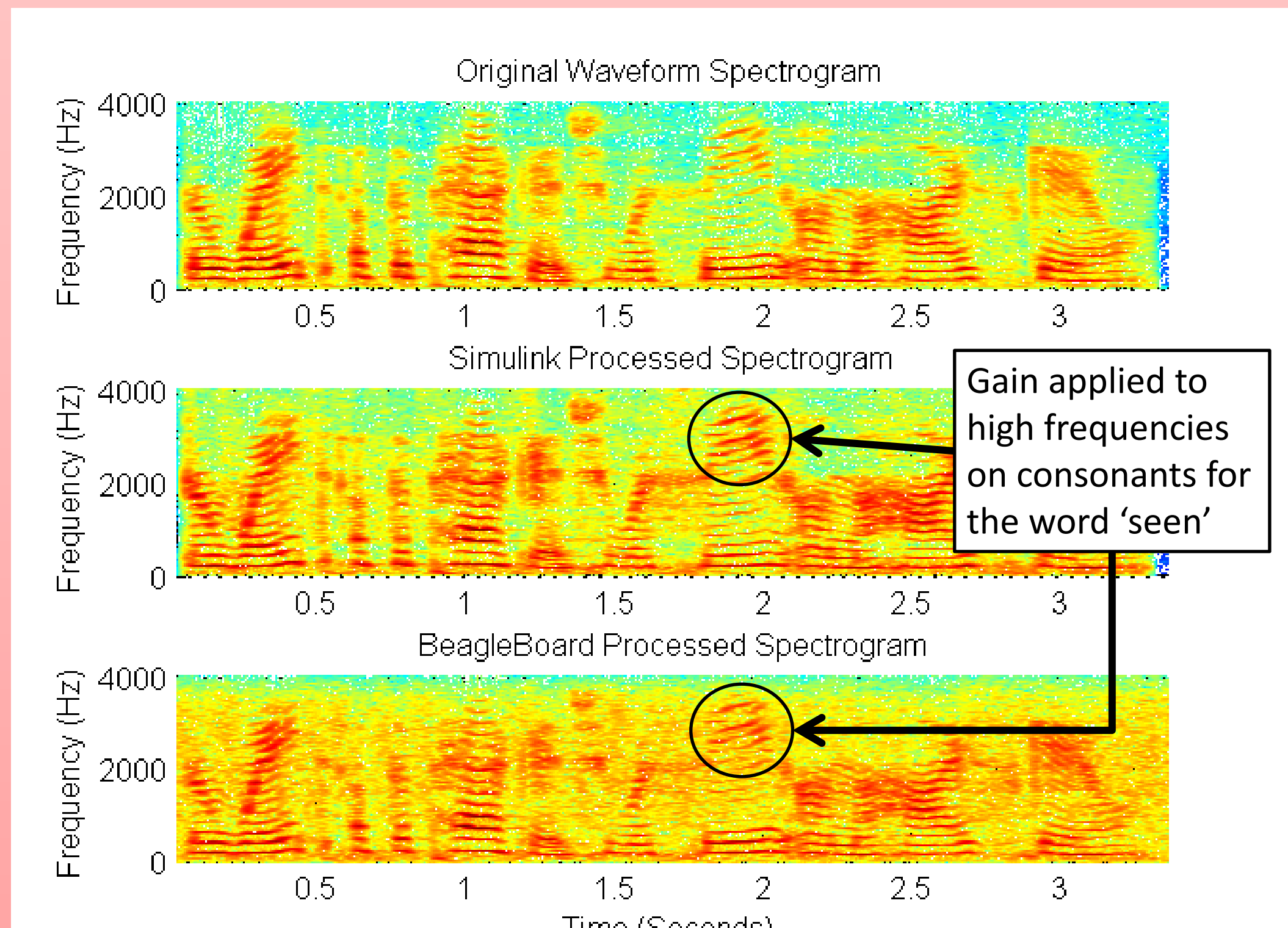
- Output of the BeagleBoard TSEA processed signals were compared to the original waveforms to verify the compression characteristics of TSEA

Waveform Output of Original, Simulink and BeagleBoard Model



- Spectrogram analysis was used to analyze the frequency components as a function of time and to validate applied gain to high frequencies in the algorithm

Spectrogram of Outputs Displaying Compression Characteristics



## Conclusion

- The Telephone Speech Enhancement Algorithm (TSEA) successfully implemented on the BeagleBoard-xM
- This hardware model could be the foundation used to preprocess telephone signals sent to hearing impaired listeners to help improve speech perceptibility
- Noise characteristics of the BeagleBoard-xM influence the model's behavior and performance

## Acknowledgements

I would like to thank Dr. Ashok Krishnamurthy and Dr. Lawrence Feth for acting as my advisors and guiding me through this project. Also, thank you to Dr. Bradley Clymer for helping with all the logistical efforts in making this project feasible. Without the initial development of TSEA by Harikrishna P. Natarajan, TSEA would not have come to fruition. The following work done by Resmi Kartha Komatitil was another huge contribution that brought TSEA to a real-time stage in Mathwork's Simulink.