Incorporating Auditory Processing into Undergraduate Signal Processing Courses to Enhance Student Learning

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***Abstract*— This paper describes the educational experience of developing lab components and projects designed to enhance understanding key signal processing concepts in undergraduate Signals and Systems courses. Students work with signal-processing algorithms commonly used in auditory models and speech-processing applications for hearing aids and cochlear implants. This approach provides an interactive way to understand digital filters, nonlinear systems, speech analysis and synthesis, Fourier transform, and modulation/demodulation. Assessment results show that students have a highly favorable view of these specially designed labs and projects.**

Index terms — signal processing, student learning, auditory system, cochlear implant, vocoder.

I. Introduction

Signal processing is a foundational subject in electrical engineering and other engineering majors, providing students with essential skills for analyzing and manipulating signals, systems, and data. The classic Signals and Systems courses cover the fundamental knowledge of signal operations, system properties, filters, modulation/demodulation, Fourier transform, etc., well in depth. However, engineering students typically understand engineering science concepts but lack the skill to apply what they learned to solve real-world engineering problems [1]. Incorporating labs and projects tied to real-world applications into educational curricula offers significant benefits for student learning [2][3]. The hands-on activities can enhance engagement and motivation by connecting theoretical concepts to practical applications, which helps them develop critical thinking and problem-solving skills.

In Signals and Systems courses, many real-world signals can be used to design labs and projects that provide students with hands-on experience. Audio signals, particularly speech, are ideal for teaching time-domain and frequency-domain analysis. Students can visualize waveforms, apply Fourier transform, and analyze spectrograms to grasp concepts like filter, spectrum, and spectrogram. These signal-processing techniques are also widely used in applications such as auditory modeling, hearing aids, and cochlear implants [4-6]. Working with speech signals allows students to immediately observe the effects of their processing techniques, deepening their understanding of fundamental signal-processing concepts. By showcasing how auditory processing algorithms are applied in real-world scenarios—like noise cancellation, speech recognition, and hearing enhancement—students can better understand their studies' relevance and impact on everyday technology.

This paper details a series of laboratory experiments and projects focused on processing audio signals, particularly in the context of auditory modeling and hearing devices. Section II outlines the general design of the labs and projects, emphasizing the connection between signal processing concepts and their real-world applications. It also describes the key components of each lab and project. Section III presents the assessment results from student surveys, while Section IV offers a short summary and conclusions.

II. Description of the Experimental Labs and Projects

The signals and systems course covers a range of essential concepts in signal processing. Figure 1 illustrates the overall design of the lab series, which helps students grasp the primary concepts introduced in the course. One lab begins by exploring linear and nonlinear systems, supported by experiments demonstrating the cochlea's nonlinear processing and the half-wave/full-wave rectification used in auditory processing. The second lab component focuses on filters and Fourier transform, with practical lab procedures on designing low-pass, band-pass, and high-pass filters to filter speech signals. Students also engage in filter design through two projects (Sections II C, and

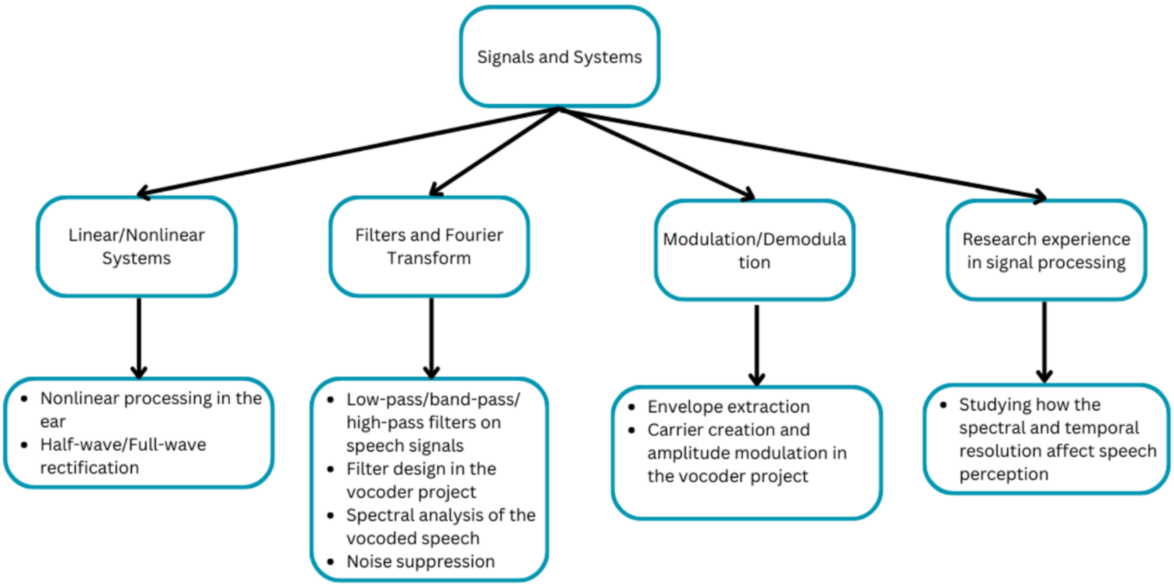

Fig. 1 Overall design of the lab series to connect the key signal processing concepts to real applications

D): developing a speech vocoder and designing noise suppression algorithms. Modulation and demodulation techniques are also introduced, with labs on envelope extraction, carrier creation, and amplitude modulation within the vocoder project. Additionally, students gain research experience by studying how spectral and temporal resolution impact speech perception in the speech vocoder project.

A. Nonlinear processing vs. linear processing

Understanding the differences between linear and nonlinear processing is crucial in signal processing. Nonlinear systems do not follow the principle of superposition. The response from nonlinear processing typically contains distortions and interactions between different frequency components, demonstrated by the presence of new frequency components (Fig. 2). The auditory system exhibits nonlinear behavior at various stages, ranging from the peripheral regions to the cortical level [7]. One great example is the Distortion Product Otoacoustic Emissions (DPOAEs) phenomenon [8], which occurs when the cochlea is simultaneously stimulated with two pure tones at different frequencies, f_1 and f_2. To illustrate how the combination tones ($f_1 + f_2$ and $f_2 - f_1$) are generated

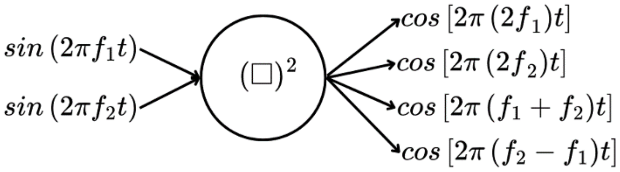

Fig. 2 Nonlinear processing in the auditory system.

due to nonlinear processing; in one lab, students are asked to mathematically derive how the combination tones can be generated by applying the squaring operation to the mixed signal. Next, students work on programming to create two tones at 1,000 Hz and 1,200 Hz and then play each separately. To experience the nonlinear processing in the ear, students write code to mix the two tones and play them simultaneously. Students are instructed to listen to the mixed tone and should hear a noticeable low-pitched tone at 200 Hz and a high-pitched tone at 2,200 Hz, which are not present in the mixed tone. Students not only get the chance to study the fundamental causes of nonlinear distortions but also learn about the practical applications of nonlinearity properties in human hearing screening using DPOAEs.

B. Digital filers and speech signal filtering

Digital filters and their design and implementation are the critical components of signals and systems undergraduate courses. Auditory models often include envelope extraction to simulate how the human ear processes sound. In hearing aids and cochlear implants, signal rectifiers are used to process speech signals to extract features and enhance speech intelligibility. Applying different filters to speech signals and exploring how they affect speech perception can provide an interactive way to understand digital filters' fundamentals.

Filters are commonly used in signal processing for hearing aids, cochlear implants, and other hearing devices. In the lab series, students design and apply three different types of filters (low-pass, band-pass, and high-pass) to process speech signals, allowing them to listen to the results and gain hands-on experience. They can vary the cutoff frequency of each filter to study how it affects speech perception performance. This practical approach helps students understand the impact of different filter parameters and techniques on real-world audio processing, reinforcing theoretical concepts through direct application.

In addition, students are given the coding task of extracting the envelope signal of a speech sound. First, the speech signal passes through either a half-wave or full-wave rectifier for envelope extraction, another example of nonlinear processing. Then, the filters at cutoff frequencies from 400 Hz down to 4 Hz will be applied to extract a smooth envelope signal. Plots can be created to demonstrate how the cutoff corner frequency can affect the smoothness of the envelope of a speech signal.

C. Filter design and speech denoising

One primary ABET(Accreditation Board for Engineering and Technology) student learning outcome is acquiring the ability to identify, formulate, and solve complex engineering problems by applying engineering, science, and mathematics principles. To meet this requirement, a short design project was incorporated into the labs of the undergraduate course. Students are tasked with designing and implementing digital filters to remove noise from the mixture of a clean speech signal and noise signal (Fig. 3). The tasks include:

- *Using FFT to find the dominant frequency range of the noise signal*
- *Determine the filter needed and its parameters*
- *Find the input-output difference equation that represents the filter*
- *Write a FOR loop in Matlab/Python to implement the filter by processing the noisy speech sample by sample*
- *Demonstrate how the cutoff frequency setting and the filter's order affect the speech quality after noise suppression.*

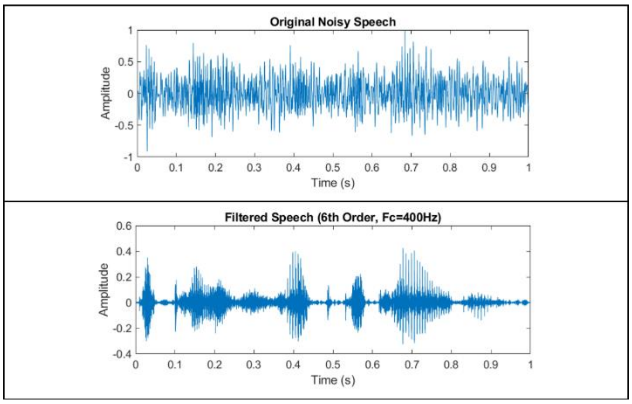

Fig. 3 The waveform of the noisy speech signal and the waveform of the expected output signal after noise removal

D. The sound vocoder project

A sound vocoder (Fig. 4) is a signal-processing algorithm used to analyze and synthesize audio signals for applications such as audio compression and cochlear implant simulations [9-11]. It incorporates many of the key signal-processing components typically covered in undergraduate Signals and Systems courses, including band-pass filters, low-pass filters, nonlinear rectifiers, envelope extraction, amplitude modulation, etc. (Fig. 4). Students will also be able to learn speech analysis, speech synthesis, and FFT-based spectrum and spectrogram

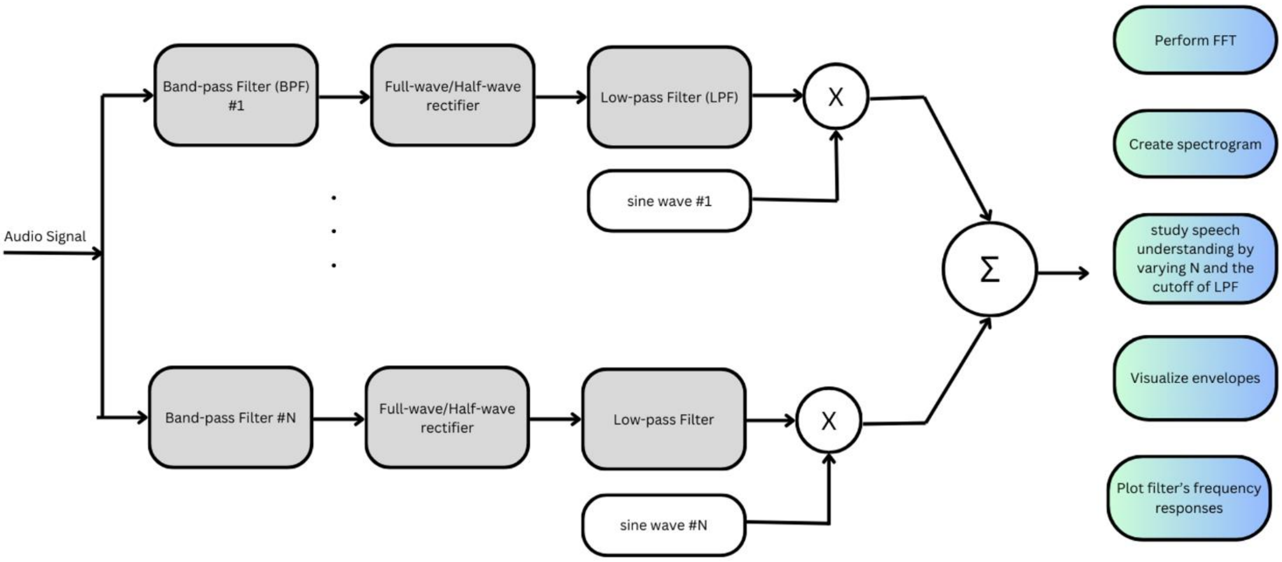

Fig. 4 Block diagram of a speech vocoder and expected leaning outcomes.

generation techniques. It can also help students develop better coding skills. The major requirements are:

- *Design and apply Butterworth band-pass and low-pass filters*
- *Extract envelopes from the filtered signal*
- *Modulate envelopes with sinusoidal carriers*
- *Rethenthesize a sound from N bands*
- *Create spectrograms to demonstrate the spectral contents of vocoded speech (Fig. 5)*
- *Study speech understanding with reduced spectral and temporal cues*

Students can also gain experience conducting research and analyzing data in the vocoder project. One of the assignments is to study how speech understanding performance can be affected by spectral resolution when varying the number of bands, *N*. Fig. 6 presents a sample result, demonstrating that the percentage of words correctly identified increases with the number of bands.

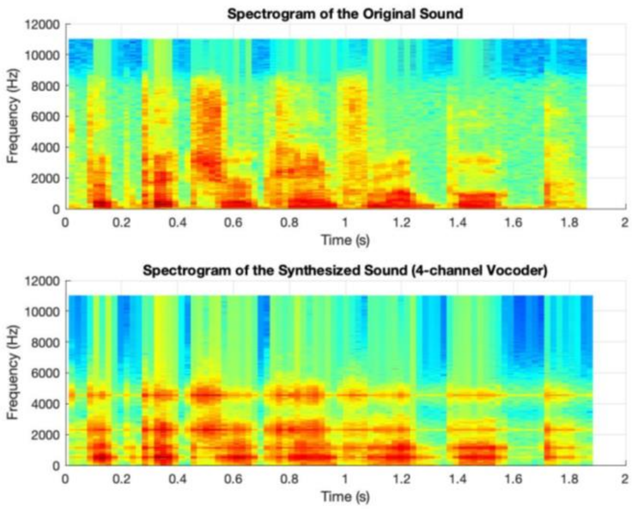

Fig. 5 Spectrograms of a speech signal and the vocoded speech.

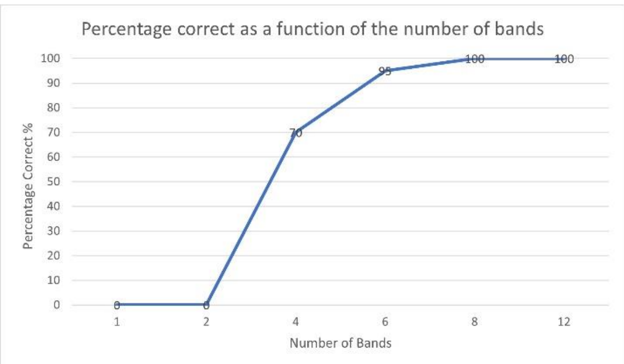

Fig.6 Study results to demonstrate the performance of speech understanding when varying the number of processing bands in the speech vocoder.

III. Assessment Results

To evaluate the effectiveness of the labs and projects involving auditory processing, students were asked to provide feedback at the end of the course. Out of 24 students, 15 responded to the survey questions, as shown in Table I. Over 93% of the students rated the labs and projects as good to excellent, indicating success in stimulating their interest in learning and enhancing their understanding of key signal processing concepts.

Table I Student rating scores on the labs and projects

Survey Questions	Good, very good, and excellent
The lab section as a whole was	100%
The content of the lab section was	93%
The interest level of lab sessions was	100%
The amount you learned in the lab sections was	93%
The relevance and usefulness of lab section content was	93%

IV. Summary and Conclusions

By integrating auditory processing into signal processing courses, students can gain a deeper understanding of theoretical concepts in digital signal processing through engaging examples and hands-on practices. It can stimulate students' interest in learning by illustrating how the auditory system processes sounds and allowing them to hear the effects of signal processing. This approach enhances student learning outcomes and better prepares students for careers in technical fields related to audio and speech signal processing.

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