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SIGNALS AND SYSTEMS

CEP PROJECT

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INTRODUCTION:

In the realm of signal processing, the enhancement of speech signals plays a pivotal role in numerous applications, from telecommunications to voice recognition systems. One of the primary challenges faced in these applications is the presence of unwanted noise that can degrade the quality of the speech signal. This report focuses on the development and implementation of a signal processing system designed to improve the quality of speech signals by employing spectrum analysis and filtering techniques.

The primary objective of this system is to selectively attenuate or eliminate undesirable frequency components within the speech signal, commonly introduced by external and internal sources of noise. By leveraging advanced signal processing techniques, we aim to extract and enhance the essential features of the speech signal while mitigating the impact of noise. Specifically, this report details the design and application of a low-pass Finite Impulse Response (FIR) filter to achieve this enhancement.

Through a comprehensive exploration of the methodology, results, and performance analysis, this report provides insights into the efficacy of the employed signal processing techniques. The chosen approach not only addresses the immediate need for noise reduction but also opens avenues for further refinement and optimization based on specific use cases and environmental conditions.

As we delve into the details of the implemented system, we invite the reader to journey through the various stages of signal processing, from recording the original speech signal to the application of sophisticated filtering algorithms. The subsequent sections will unveil the visual and analytical assessments of the enhancement achieved, laying the groundwork for understanding the significance of this signal processing endeavor in the broader context of speech signal improvement.

RECORDING

The methodology begins with the recording of a speech signal using a 16-bit, mono audio recorder operating at a sampling frequency of 44.1 kHz. The speech signal, capturing both desired speech elements and unwanted noise, serves as the input to the subsequent signal processing stages.

```
% Code for recording
```

```
fs = 44100; % Sampling frequency

recorder = audiorecorder(fs, 16, 1); % 16 bits, 1 channel (mono)

disp('Start speaking...');

recordblocking(recorder, 5); % Record for 5 seconds

disp('End of recording');


% Get the recorded signal

speech_signal = getaudiodata(recorder);
```

The code snippet initializes an audio recorder with the specified parameters, prompts the user to speak for 5 seconds, and captures the recorded signal for further processing.

SIGNAL PROCESSING

The core of the methodology lies in the application of signal processing techniques to enhance the recorded speech signal. A low-pass Finite Impulse Response (FIR) filter is designed and applied to attenuate high-frequency noise components. The filter parameters, including order and cutoff frequency, are crucial in achieving an optimal trade-off between noise reduction and preservation of speech features.

```
% Code for signal processing

order = 8; % Filter order

cutoff_frequency = 4000; % Cutoff frequency in Hz

% Design a low-pass filter

lowpass_filter = designfilt('lowpassfir', 'FilterOrder', order, 'CutoffFrequency', cutoff_frequency, 'SampleRate', fs);

% Apply the filter to the speech signal

enhanced_signal = filter(lowpass_filter, speech_signal);
```

This section establishes the filter order and cutoff frequency, designs the low-pass FIR filter, and applies it to the recorded speech signal, resulting in an enhanced signal.

RESULTS

ORIGINAL VS. ENHANCED SIGNAL

To visually assess the impact of the applied filter, time-domain plots of the original and enhanced speech signals are generated. These plots showcase the effectiveness of the low-pass filter in reducing noise and preserving the essential speech components.

```
% Visualization of signals

t = (0:length(speech_signal)-1) / fs;


figure;

subplot(2,1,1);

plot(t, speech_signal);

title('Original Speech Signal');

xlabel('Time (s)');

ylabel('Amplitude');

subplot(2,1,2);

plot(t, enhanced_signal);

title('Enhanced Speech Signal');

xlabel('Time (s)');

ylabel('Amplitude');
```

The resulting figures provide a side-by-side comparison, allowing for a qualitative assessment of the improvements achieved through signal processing.

SPECTROGRAM ANALYSIS

A spectrogram, displaying the frequency content of the enhanced speech signal over time, offers a more detailed insight into the filtering effects. This analysis visually represents the removal of high-frequency noise components, emphasizing the clarity and quality enhancement.

% Spectrogram analysis

figure;

spectrogram(enhanced_signal, hann(256), 128, 1024, fs, 'yaxis');

title('Spectrogram of Enhanced Speech Signal');
The spectrogram plot complements the time-domain analysis, providing a comprehensive view of the frequency distribution within the enhanced signal.

CONCLUSION

Through the combined methodology of recording, signal processing, and result analysis, this report demonstrates the successful enhancement of speech signal quality. The utilization of a low-pass FIR filter proves effective in reducing noise and preserving valuable speech information. The ensuing visual and analytical results offer a comprehensive evaluation of the system's performance, setting the stage for potential refinements and applications in real-world scenarios.