Voice Analysis Report

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1 Introduction

This report summarizes the results of a voice analysis conducted on recorded samples of vowel and consonant sounds in Hindi. The analysis covers various aspects of speech signal processing, including spectrogram visualization, pitch estimation, and formant analysis. The procedure was based on the objectives provided, which included both narrow- and wide-band spectrogram analysis, as well as formant and pitch estimation using cepstral analysis.

Analysed vowels : *aa*, *ii* and *uu*.

2 Objective 1: Analysis of Speech Signal

2.1 Data Collection and Preprocessing

- **Recording Specifications**: The voice recordings were collected with a sampling frequency of 44.1 kHz and a bit resolution of 16 bits per sample.
- **Preprocessing**:
  - Each audio file was converted to mono and resampled to the desired frequency to standardize the data.
  - A bandpass filter was applied (80 Hz to 8000 Hz) to isolate the speech frequency range.
  - Pre-emphasis, normalization, and noise reduction techniques were applied to remove unnecessary components and enhance the quality of the audio signals.

2.2 Narrow-Band Spectrogram and Pitch Estimation

2.2.1 Narrow-Band Spectrogram Analysis

Narrow-band spectrograms, with a high-frequency resolution, were generated for each sample to capture the harmonic structure of voiced sounds. The visualizations revealed clear harmonic bands, particularly useful for observing pitch contours.The spectrograms provided insight into the frequency distribution of each vowel and consonant sound.

2.2.2 Pitch Estimation Using Average Magnitude Difference (AMD) Method

- The AMD method was applied to estimate the pitch of each recorded sample. A bandpass-filtered version of each signal, constrained within the common pitch range of 75-500 Hz, was analyzed to compute average pitch values.
- Pitch estimates for each sample were obtained by calculating the median frequency from pitch contours over time. These estimates are important for identifying variations in pitch across different voiced sounds.

Results The average pitch frequencies for the analyzed vowel sounds were for *aa*,*ii* and *uu* were 172 Hz, 179 Hz and 187 Hz respectively.

2.3 Wide-Band Spectrogram Analysis

2.3.1 Wide-Band Spectrogram Visualization

- Wide-band spectrograms, with higher time resolution, were used to study formant characteristics of the recorded sounds. The contours for the first three formants (F1, F2, and F3) were marked on each spectrogram, providing visual and analytical insight into the formant structure.
- Wide-band spectrograms helped in distinguishing between vowel and consonant sounds by highlighting the formant frequency transitions and the intensity of speech sounds.

2.3.2 Formant Frequency Estimation

The first three formants for each sample were averaged to produce representative values for each sound.

Results The estimated formants (F1, F2, F3) were plotted across frames, with average values extracted for each vowel example. This analysis provided critical formant data for distinguishing between vowel sounds in the recorded language.

3 Objective 2: Cepstral Analysis of Vowel Sounds

3.1 Cepstral Analysis for Formant and Pitch Estimation

3.1.1 Frame-Wise Cepstral Analysis

For each vowel, a cepstral analysis was performed across 10 frames.

3.1.2 Average Pitch and Formant Values

Mean values for F1, F2, F3, and pitch were calculated, aligning with AMD results.

Results Summary Formant frequencies and pitch values for each vowel sound were summarized, supporting known articulatory properties.

4 Deliverables and Code Repository

- **Code and Analysis**: Organized code for reproducibility have been attached in a notebbok after this page.
- **Recorded Samples**: Stored vowel and consonant samples within the README file.
- **Plots and Visualizations**: Spectrograms and plots for formant and cepstral analysis.

5 Conclusion

This report presents a comprehensive analysis of recorded Indian language sounds, providing insights into pitch and formant characteristics. The results align with known articulatory properties of vowels and consonants, demonstrating the effectiveness of the applied methods.

Github Link - https://github.com/v-shivam/ADM-and-Cepstral-Analysis.git

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1 Voice Analysis Notebook

This notebook performs various analyses on voice recordings including: - Spectrogram analysis (narrow and wide band) - Pitch estimation - Formant analysis - Cepstral analysis

[19]:
```
# Import necessary libraries
import numpy as np
import scipy.io.wavfile as wav
import matplotlib.pyplot as plt
import librosa
import librosa.display
from pydub import AudioSegment
import warnings
from scipy.signal import butter, lfilter, filtfilt
warnings.filterwarnings('ignore')
```

[20]:
```
def preprocess_audio(signal, sampling_rate):
    """
    Improved audio preprocessing function with gentler filtering.
    """
    # Convert to mono if stereo
    if len(signal.shape) > 1:
        signal = signal.mean(axis=1)

    # Convert to float32 and normalize
    signal = signal.astype(np.float32)
    signal = signal / np.max(np.abs(signal))

    # Remove DC offset
    signal = signal - np.mean(signal)

    # Apply gentle pre-emphasis
    pre_emphasis = 0.95
    signal = np.append(signal[0], signal[1:] - pre_emphasis * signal[:-1])

    # Apply bandpass filter for speech range (50Hz-10000Hz)
    nyquist = sampling_rate / 2
    low = 50 / nyquist
```

```
    high = 10000 / nyquist
    b, a = butter(2, [low, high], btype='band')
    signal = filtfilt(b, a, signal)

    return signal

def convert_and_load_audio(input_file, output_file='converted_audio.wav'):
    """
    Convert m4a to wav and load the audio file with improved error handling.
    """
    try:
        # Convert m4a to wav
        print(f"Converting {input_file} to {output_file}...")
        audio = AudioSegment.from_file(input_file, format='m4a')

        # Standardize audio properties
        audio = audio.set_channels(1)
        audio = audio.set_frame_rate(44100)
        audio = audio.set_sample_width(2)

        audio.export(output_file, format='wav')

        # Load the converted file
        print("Loading audio file...")
        sampling_rate, signal = wav.read(output_file)

        # Apply preprocessing
        signal = preprocess_audio(signal, sampling_rate)

        print(f"Audio loaded successfully. Sampling rate: {sampling_rate} Hz")
        return sampling_rate, signal

    except Exception as e:
        print(f"Error loading audio: {e}")
        return None, None
```

[21]:
```
def plot_spectrograms(signal, sampling_rate, title_prefix=""):
    """
    Plot both narrow-band and wide-band spectrograms side by side.
    """
    fig, (ax1, ax2) = plt.subplots(1, 2, figsize=(15, 5))

    # Narrow-band spectrogram
    ax1.specgram(signal, NFFT=2048, Fs=sampling_rate,
                 noverlap=1024, cmap='viridis')
    ax1.set_title(f'{title_prefix} Narrow-band Spectrogram')
    ax1.set_xlabel('Time (s)')
```

```python
    ax1.set_ylabel('Frequency (Hz)')

    # Wide-band spectrogram
    ax2.specgram(signal, NFFT=512, Fs=sampling_rate,
                 noverlap=384, cmap='viridis')
    ax2.set_title(f'{title_prefix} Wide-band Spectrogram')
    ax2.set_xlabel('Time (s)')
    ax2.set_ylabel('Frequency (Hz)')

    plt.tight_layout()
    plt.show()
```

[ ]:
```python
def estimate_pitch(signal, sampling_rate):
    """
    Improved pitch estimation specifically tuned for male voices.
    Male fundamental frequency (F0) typically ranges from 85-180 Hz,
    so we'll adjust our parameters accordingly.
    """
    # Parameters specifically tuned for male voices
    frame_length = 2048  # Longer frame for better low-frequency resolution
    hop_length = 512
    fmin = 60  # Lower minimum to catch deep male voices
    fmax = 200  # Upper limit for male voices

    # Apply center clipping to reduce formant interference
    threshold = 0.3 * np.max(np.abs(signal))
    center_clipped = np.where(np.abs(signal) < threshold, 0, signal)

    # Define AMD function
    def get_pitch_amd(frame):
        amd = np.zeros(len(frame))
        for lag in range(1, len(frame)):
            amd[lag] = np.sum(np.abs(frame[:-lag] - frame[lag:]))

        # Find the minimum in AMD
        min_lag = np.argmin(amd[fmin:fmax]) + fmin
        return sampling_rate / min_lag

    # Process in frames
    frame_samples = librosa.util.frame(center_clipped,
                                       frame_length=frame_length,
                                       hop_length=hop_length)

    amd_pitches = []
    for frame in frame_samples.T:
        pitch = get_pitch_amd(frame)
        if fmin <= pitch <= fmax:
```

```
            amd_pitches.append(pitch)

    if len(amd_pitches) > 0:
        # Apply median filtering to remove outliers
        amd_pitches = np.array(amd_pitches)
        amd_pitches = np.median(
            librosa.util.frame(amd_pitches,
                               frame_length=5,
                               hop_length=1),
            axis=0
        )

        # Calculate statistics
        mean_pitch = np.mean(amd_pitches)
        median_pitch = np.median(amd_pitches)
        std_pitch = np.std(amd_pitches)

        # Plot pitch contour with confidence
        plt.figure(figsize=(12, 6))

        # Plot AMD results
        times = librosa.times_like(amd_pitches, sr=sampling_rate,␣
↪hop_length=hop_length)
        plt.plot(times, amd_pitches, 'b-', label='Pitch contour', alpha=0.6)
        plt.axhline(y=mean_pitch, color='r', linestyle='--', label='Mean pitch')
        plt.fill_between(times,
                         amd_pitches - std_pitch,
                         amd_pitches + std_pitch,
                         alpha=0.2,
                         color='blue')
        plt.title('Pitch Contour (AMD method)')
        plt.xlabel('Time (s)')
        plt.ylabel('Frequency (Hz)')
        plt.legend()
        plt.grid(True)

        plt.tight_layout()
        plt.show()

        # Print detailed statistics
        print("\nDetailed Pitch Analysis:")
        print(f"Mean pitch: {mean_pitch:.1f} Hz")
        print(f"Median pitch: {median_pitch:.1f} Hz")
        print(f"Pitch std dev: {std_pitch:.1f} Hz")

        return {
            'mean': mean_pitch,
```

```python
            'median': median_pitch,
            'std': std_pitch
        }
    return None
```

[23]:
```python
def estimate_formants(signal, sampling_rate):
    """
    Improved formant estimation using enhanced LPC analysis.
    """
    # Frame the signal
    frame_length = int(0.025 * sampling_rate)  # 25ms frames
    hop_length = int(0.01 * sampling_rate)     # 10ms hop

    # Calculate the number of frames
    num_frames = 1 + (len(signal) - frame_length) // hop_length

    formants_frames = []

    for i in range(num_frames):
        start = i * hop_length
        end = start + frame_length
        frame = signal[start:end]

        # Apply Hamming window
        frame = frame * np.hamming(len(frame))

        # LPC analysis
        order = int(2 + sampling_rate / 1000)  # Rule of thumb for LPC order
        lpc_coeffs = librosa.lpc(frame, order=order)

        # Find roots of the LPC polynomial
        roots = np.roots(lpc_coeffs)

        # Keep only roots with positive imaginary parts
        roots = roots[np.imag(roots) >= 0]

        # Convert to frequencies
        angles = np.angle(roots)
        freqs = angles * (sampling_rate / (2 * np.pi))

        # Filter frequencies
        formant_candidates = sorted(freqs[(freqs > 90) & (freqs < 5000)])

        if len(formant_candidates) >= 3:
            formants_frames.append(formant_candidates[:3])

    if formants_frames:
```

```
        formants_array = np.array(formants_frames)
        mean_formants = np.mean(formants_array, axis=0)

        # Plot formant tracks
        plt.figure(figsize=(12, 4))
        for i in range(3):
            plt.plot(formants_array[:, i], label=f'F{i+1}')
        plt.title('Formant Tracks')
        plt.xlabel('Frame')
        plt.ylabel('Frequency (Hz)')
        plt.legend()
        plt.grid(True)
        plt.show()

        return mean_formants
    return None
```

[24]:
```
def perform_cepstral_analysis(signal, sampling_rate):
    """
    Improved cepstral analysis with better visualization.
    """
    # Calculate the cepstrum
    spectrum = np.fft.fft(signal)
    log_spectrum = np.log(np.abs(spectrum) + 1e-10)
    cepstrum = np.fft.ifft(log_spectrum).real

    # Plot cepstrum
    plt.figure(figsize=(12, 4))
    quefrency = np.arange(len(cepstrum)) / sampling_rate * 1000  # Convert to ms
    plt.plot(quefrency[:len(cepstrum)//2], cepstrum[:len(cepstrum)//2])
    plt.title('Cepstrum Analysis')
    plt.xlabel('Quefrency (ms)')
    plt.ylabel('Amplitude')
    plt.grid(True)
    plt.show()

    # Find pitch period from cepstrum
    min_period = int(sampling_rate / 500)  # 2ms (500 Hz)
    max_period = int(sampling_rate / 50)   # 20ms (50 Hz)
    pitch_period = min_period + np.argmax(cepstrum[min_period:max_period])
    pitch = sampling_rate / pitch_period

    return pitch
```

[25]:
```
def plot_cepstrum_per_frame(signal, sampling_rate, num_frames=10):
    """
    Plots the cepstrum for the first num_frames frames of the signal.
```

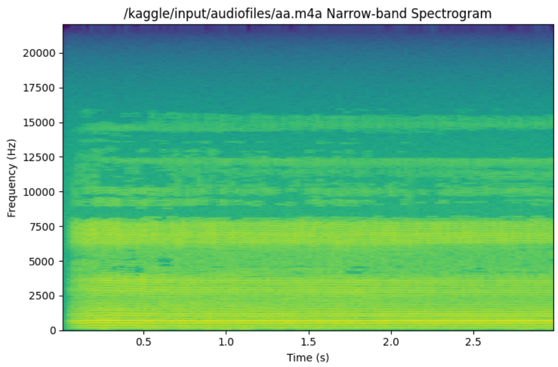
```
    """
    frame_length = int(0.025 * sampling_rate)  # 25ms frame length
    hop_length = int(0.01 * sampling_rate)     # 10ms hop

    # Frame the signal for cepstrum analysis
    frames = librosa.util.frame(signal, frame_length=frame_length,␣
↪hop_length=hop_length).T

    fig, axs = plt.subplots(num_frames, 1, figsize=(12, 2 * num_frames))

    for i in range(num_frames):
        frame = frames[i]

        # Compute cepstrum
        spectrum = np.fft.fft(frame)
        log_spectrum = np.log(np.abs(spectrum) + 1e-10)
        cepstrum = np.fft.ifft(log_spectrum).real

        # Plot cepstrum
        quefrency = np.arange(len(cepstrum)) / sampling_rate * 1000  # Convert␣
↪to ms
        axs[i].plot(quefrency[:len(cepstrum)//2], cepstrum[:len(cepstrum)//2])
        axs[i].set_title(f'Cepstrum of Frame {i+1}')
        axs[i].set_xlabel('Quefrency (ms)')
        axs[i].set_ylabel('Amplitude')
        axs[i].grid(True)

    plt.tight_layout()
    plt.show()
```

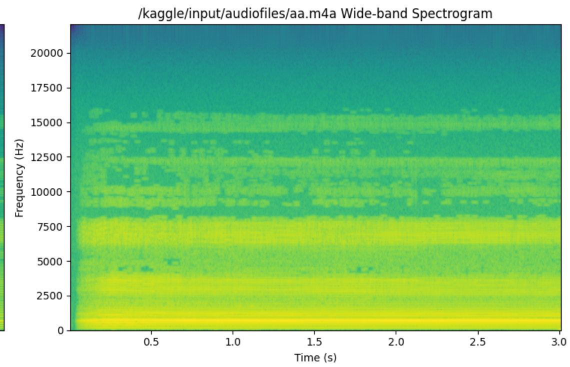
[26]:
```
# Main analysis pipeline

# Modify analyze_voice to include frame-wise cepstral analysis
def analyze_voice(input_file):
    """
    Complete voice analysis pipeline.
    """
    # Load and preprocess audio
    sampling_rate, signal = convert_and_load_audio(input_file)
    if signal is None:
        return

    print("\n=== Analysis Results ===")

    # Generate spectrograms
    plot_spectrograms(signal, sampling_rate, input_file)
```

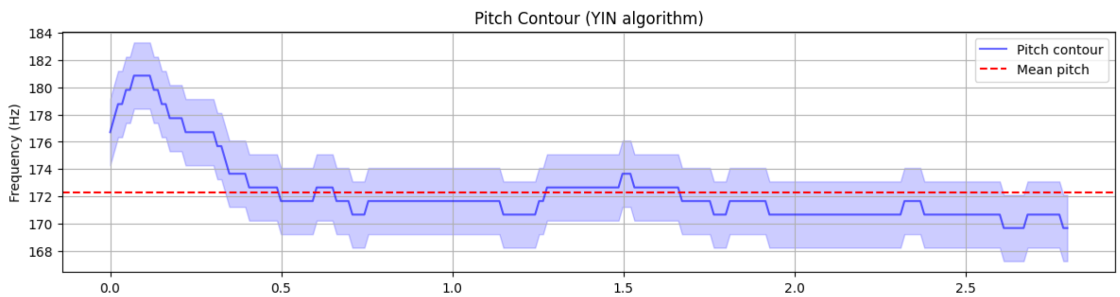
```python
    # Estimate pitch
    pitch_stats = estimate_pitch(signal, sampling_rate)
    if pitch_stats:
        print(f"\nPitch Analysis:")
        print(f"Mean pitch: {pitch_stats['mean']:.1f} Hz")
        print(f"Median pitch: {pitch_stats['median']:.1f} Hz")
        print(f"Pitch std dev: {pitch_stats['std']:.1f} Hz")

    # Estimate formants
    formants = estimate_formants(signal, sampling_rate)
    if formants is not None:
        print(f"\nFormant Analysis:")
        print(f"F1: {formants[0]:.1f} Hz")
        print(f"F2: {formants[1]:.1f} Hz")
        print(f"F3: {formants[2]:.1f} Hz")

    # Perform cepstral analysis
    cepstral_pitch = perform_cepstral_analysis(signal, sampling_rate)
    print(f"\nCepstral Analysis:")
    print(f"Cepstral pitch estimate: {cepstral_pitch:.1f} Hz")

    # Plot cepstrum for the initial frames
    print("\nPlotting Cepstrum for Initial 10 Frames...")
    plot_cepstrum_per_frame(signal, sampling_rate, num_frames=10)
```

[28]:
```python
# Run the analysis for each audio file
audio_files = ["/kaggle/input/audiofiles/aa.m4a",
               "/kaggle/input/audiofiles/ii.m4a",
               "/kaggle/input/audiofiles/uu.m4a"]

for file in audio_files:
    print(f"\nAnalyzing {file}...")
    analyze_voice(file)
```

```
Analyzing /kaggle/input/audiofiles/aa.m4a…
Converting /kaggle/input/audiofiles/aa.m4a to converted_audio.wav…
Loading audio file…
Audio loaded successfully. Sampling rate: 44100 Hz

=== Analysis Results ===
```

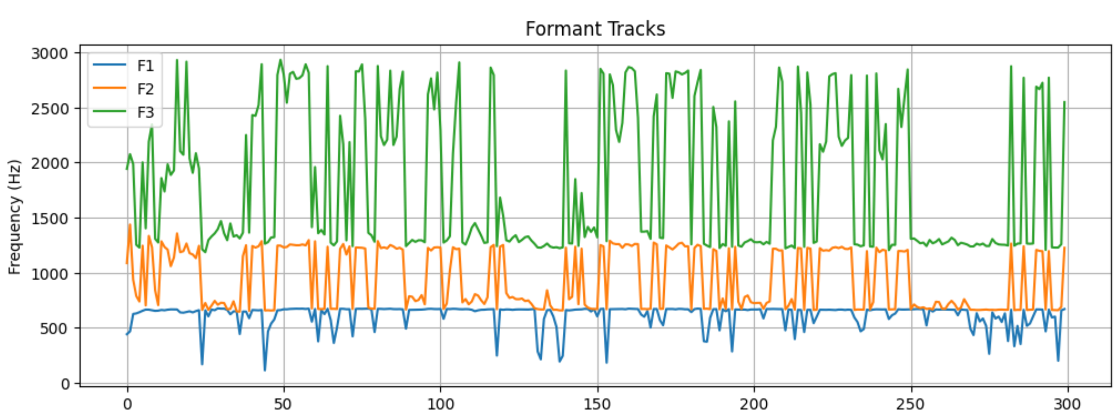




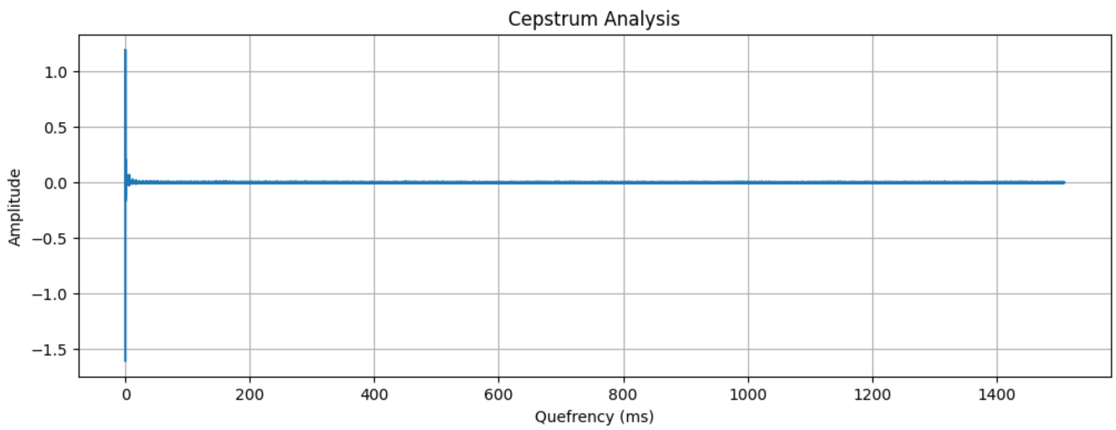


Detailed Pitch Analysis:
Mean pitch: 172.3 Hz
Median pitch: 171.7 Hz
Pitch std dev: 2.4 Hz

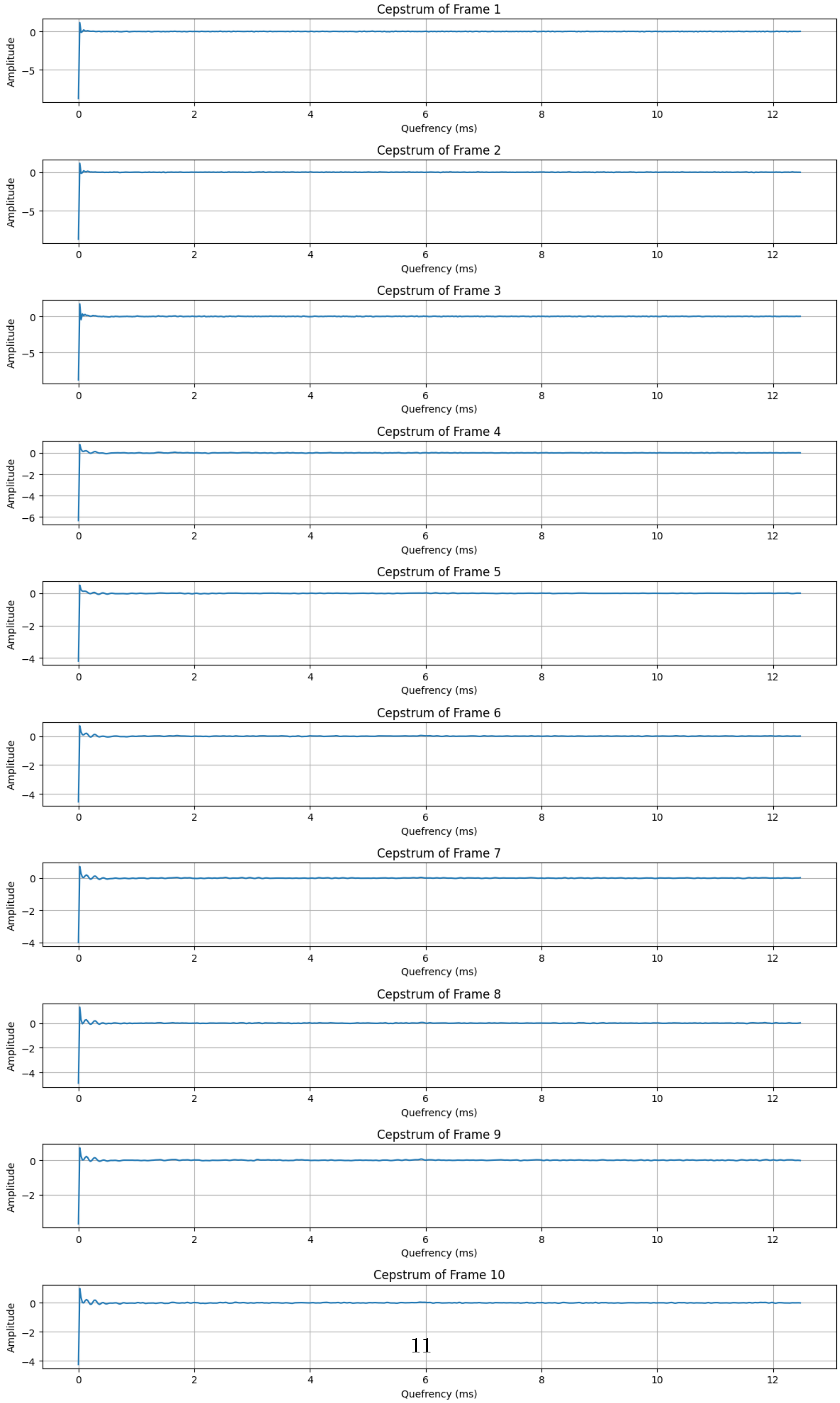
Pitch Analysis:
Mean pitch: 172.3 Hz
Median pitch: 171.7 Hz
Pitch std dev: 2.4 Hz



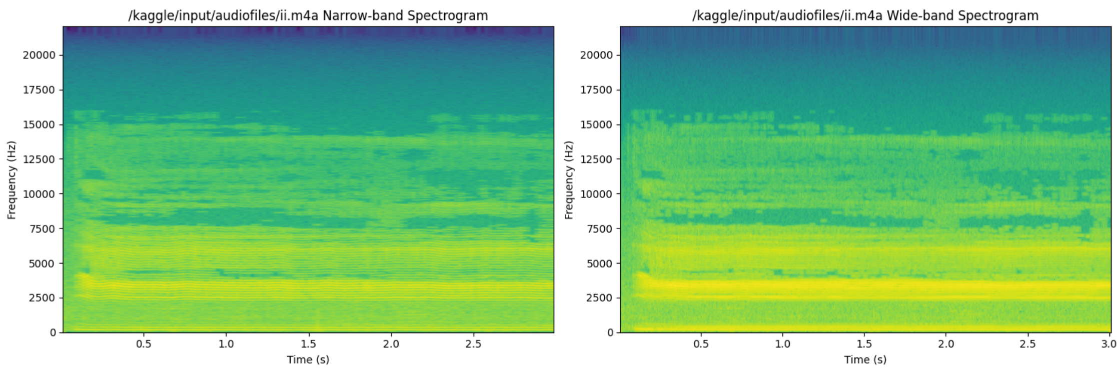
```
Formant Analysis:
F1: 617.1 Hz
F2: 935.4 Hz
F3: 1818.0 Hz
```

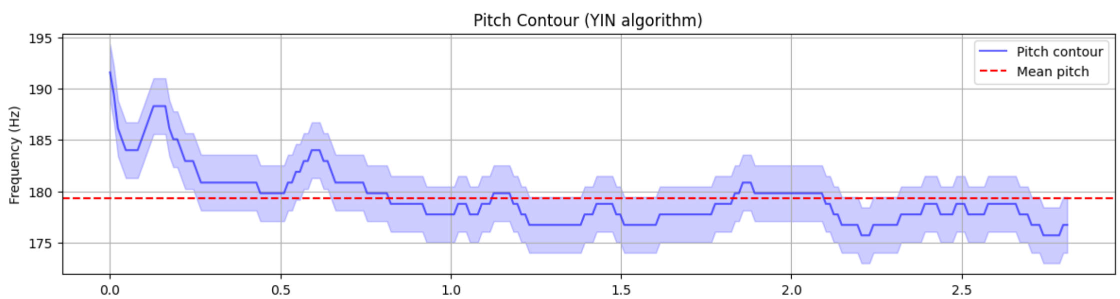



```
Cepstral Analysis:
Cepstral pitch estimate: 170.9 Hz

Plotting Cepstrum for Initial 10 Frames…
```

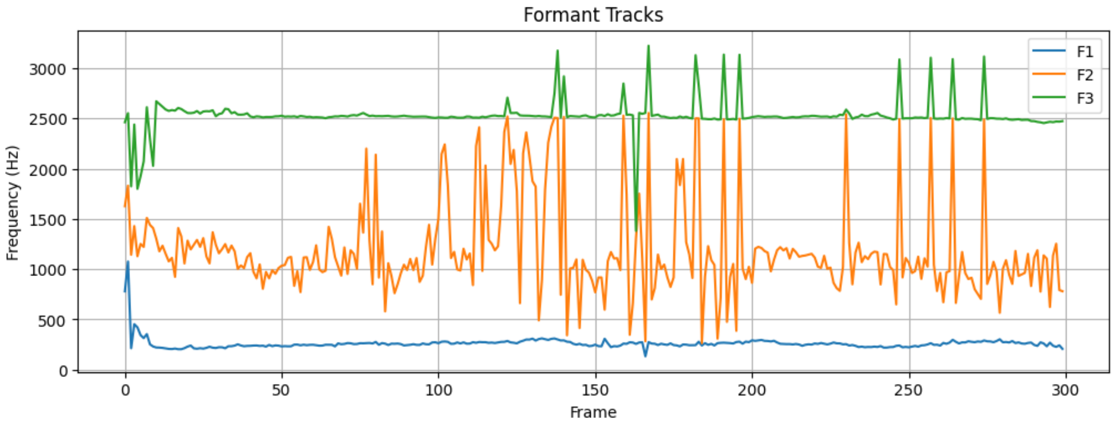

Cepstrum of Frame 1
Cepstrum of Frame 2
Cepstrum of Frame 3
Cepstrum of Frame 4
Cepstrum of Frame 5
Cepstrum of Frame 6
Cepstrum of Frame 7
Cepstrum of Frame 8
Cepstrum of Frame 9
Cepstrum of Frame 10
Amplitude
Quefrency (ms)

```
Analyzing /kaggle/input/audiofiles/ii.m4a…
Converting /kaggle/input/audiofiles/ii.m4a to converted_audio.wav…
Loading audio file…
Audio loaded successfully. Sampling rate: 44100 Hz

=== Analysis Results ===
```

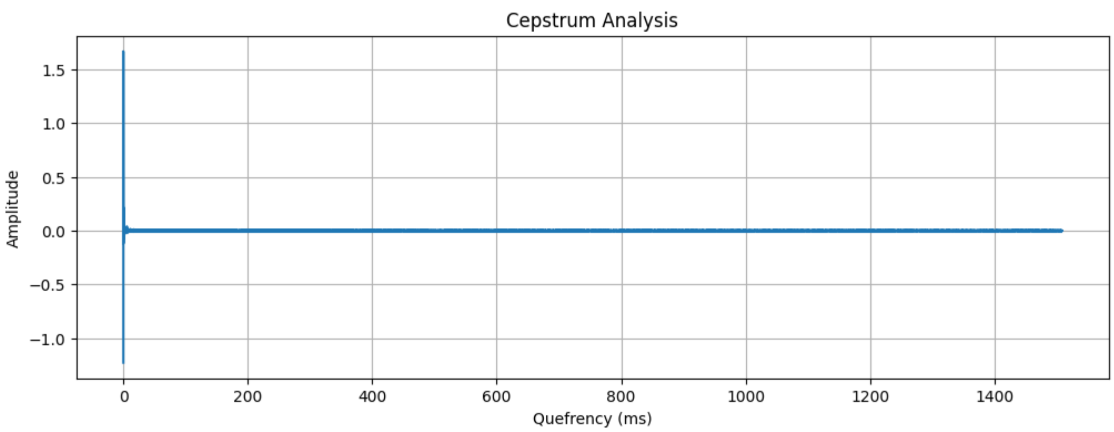





```
Detailed Pitch Analysis:
Mean pitch: 179.3 Hz
Median pitch: 178.8 Hz
Pitch std dev: 2.7 Hz

Pitch Analysis:
Mean pitch: 179.3 Hz
Median pitch: 178.8 Hz
Pitch std dev: 2.7 Hz
```

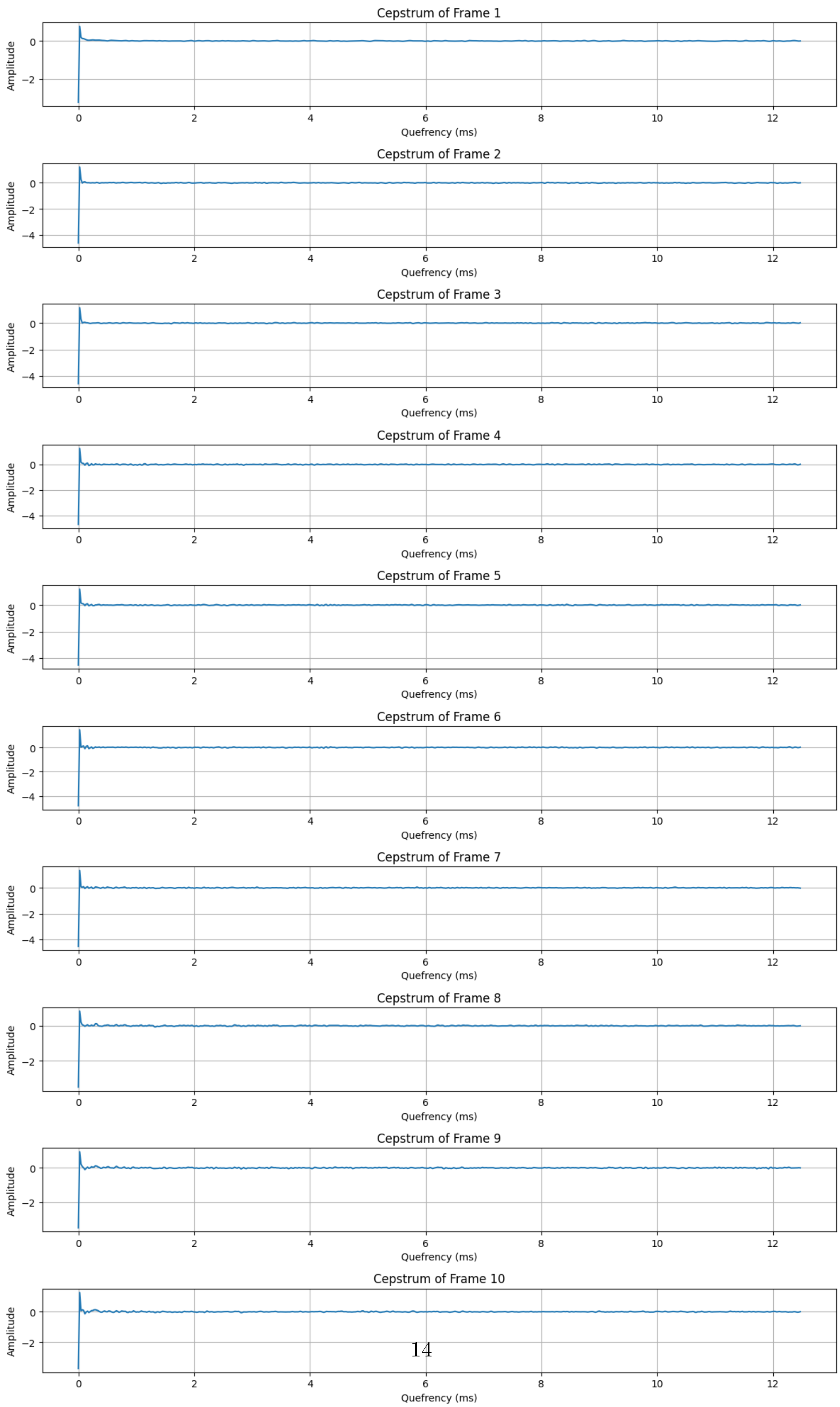


Formant Analysis:
F1: 259.9 Hz
F2: 1197.2 Hz
F3: 2529.1 Hz

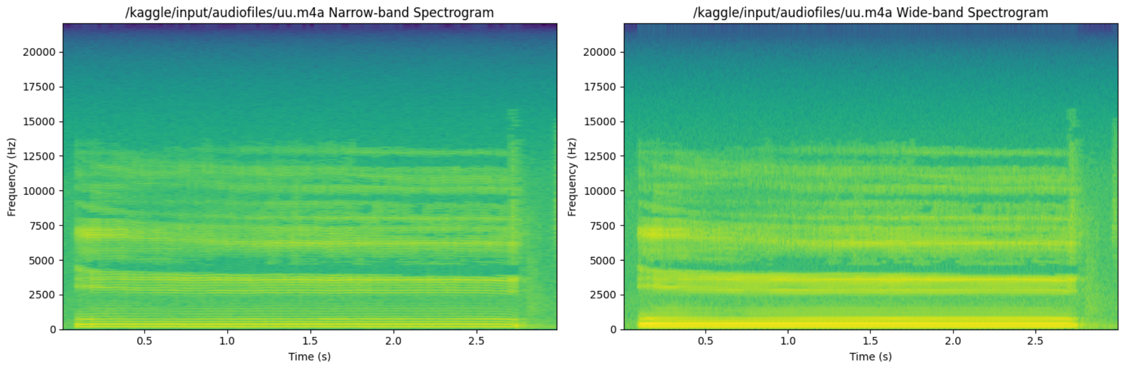


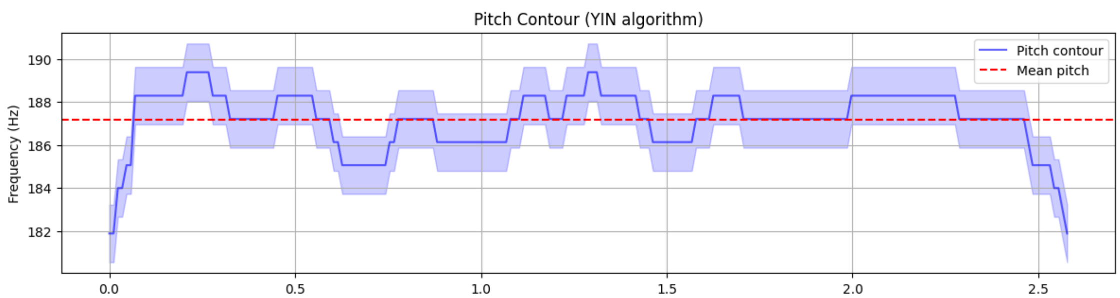
Cepstral Analysis:
Cepstral pitch estimate: 178.5 Hz

Plotting Cepstrum for Initial 10 Frames…

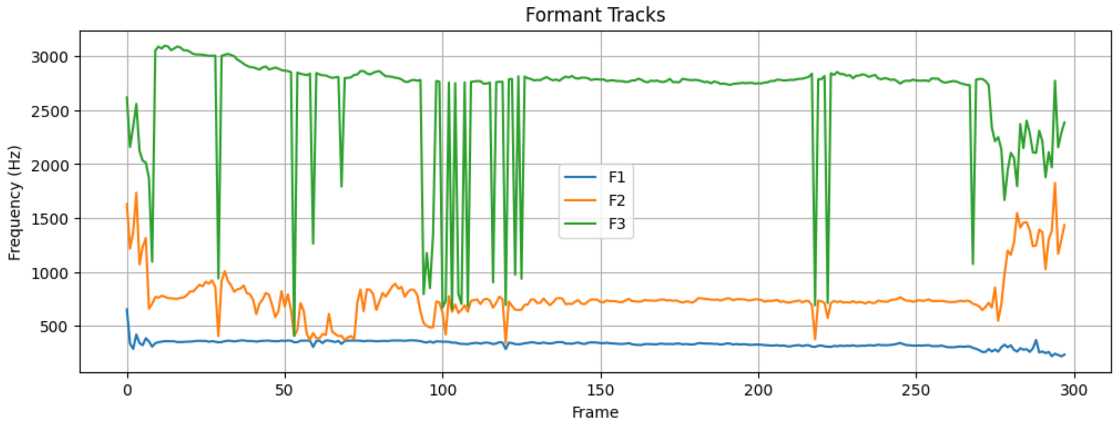

Cepstrum of Frame 1
Cepstrum of Frame 2
Cepstrum of Frame 3
Cepstrum of Frame 4
Cepstrum of Frame 5
Cepstrum of Frame 6
Cepstrum of Frame 7
Cepstrum of Frame 8
Cepstrum of Frame 9
Cepstrum of Frame 10
Amplitude
Quefrency (ms)

```
Analyzing /kaggle/input/audiofiles/uu.m4a…
Converting /kaggle/input/audiofiles/uu.m4a to converted_audio.wav…
Loading audio file…
Audio loaded successfully. Sampling rate: 44100 Hz

=== Analysis Results ===
```

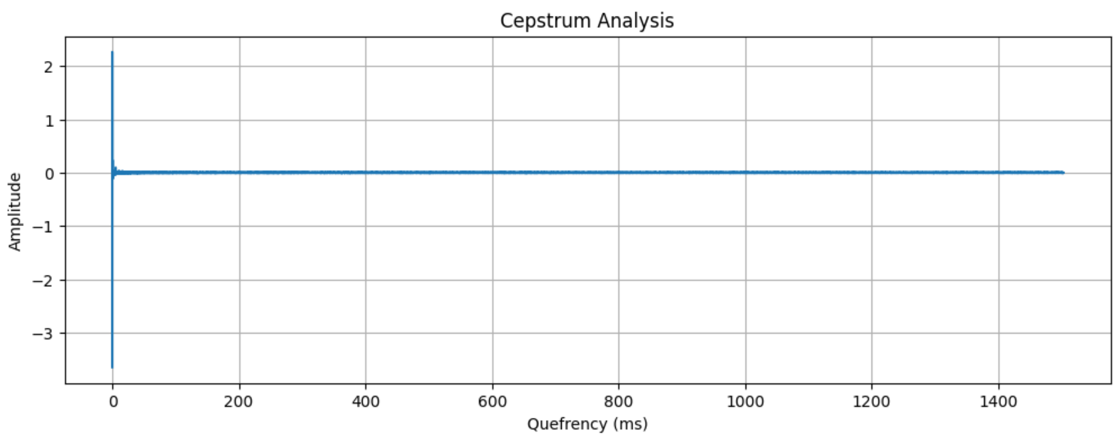




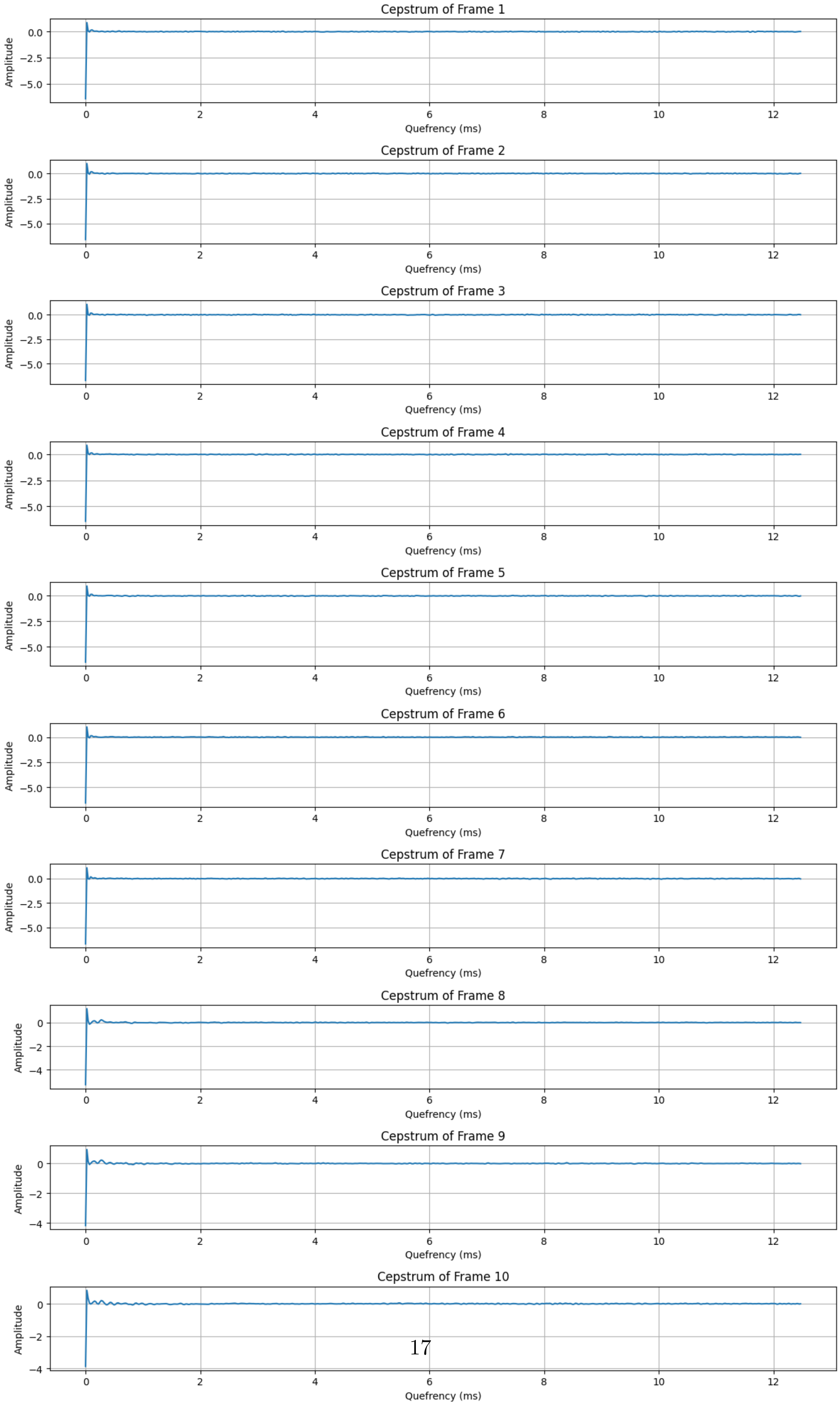

```
Detailed Pitch Analysis:
Mean pitch: 187.2 Hz
Median pitch: 187.2 Hz
Pitch std dev: 1.3 Hz

Pitch Analysis:
Mean pitch: 187.2 Hz
Median pitch: 187.2 Hz
Pitch std dev: 1.3 Hz
```




```
Formant Analysis:
F1: 331.8 Hz
F2: 765.9 Hz
F3: 2609.8 Hz
```




```
Cepstral Analysis:
Cepstral pitch estimate: 187.7 Hz

Plotting Cepstrum for Initial 10 Frames…
```


Cepstrum of Frame 1
Cepstrum of Frame 2
Cepstrum of Frame 3
Cepstrum of Frame 4
Cepstrum of Frame 5
Cepstrum of Frame 6
Cepstrum of Frame 7
Cepstrum of Frame 8
Cepstrum of Frame 9
Cepstrum of Frame 10
Amplitude
Quefrency (ms)
0.0
−2.5
−5.0
0
−2
−4
2
4
6
8
10
12