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In [5]: import numpy as np
import matplotlib.pyplot as plt
import soundfile as sf
from scipy.signal import lfilter, freqz
from scipy.signal.windows import hamming
from scipy.linalg import solve
import os
import sys

# --- 1. Standard Vowel Formant Data ---
VOWEL_FORMANT_DATA = {
    # Vowel (IPA): [F1 (Hz), F2 (Hz)]
    'i': [240, 2400],
    'y': [235, 2100],
    'e': [390, 2300],
    'ø': [370, 1900],
    'ɛ': [610, 1900],
    'œ': [585, 1710],
    'a': [850, 1610],
    'æ': [820, 1530],
    'ɑ': [750, 940],
    'ɒ': [700, 760],
    'ʌ': [600, 1170],
    'ɔ': [500, 700],
    'χ': [460, 1310],
    'օ': [360, 640],
    'ɯ': [300, 1390],
    'ʊ': [250, 595]
}

# --- 2. Configuration Parameters ---
# STANDARD PARAMETERS FOR 16 kHz SPEECH ANALYSIS
LPC_ORDER = 18 # Ideal order for 16 kHz (16/1 + 2 = 18)
FRAME_SIZE = 512
HOP_SIZE = 160
STABILITY_GAMMA = 0.999 # Standard stability factor is now sufficient for P=18

# --- 3. Speech Signal Acquisition and Framing ---
def load_and_frame_signal(file_path):
    """Loads a WAV file and splits it into overlapping frames."""

    signal, fs = sf.read(file_path)

    if fs > 20000:
        print("\n*** WARNING: High Sampling Rate Detected! ***")
        print(f"The input file has Fs={fs} Hz. For accurate formant analysis,")

    if signal.ndim > 1:
        signal = signal[:, 0]

    # Pre-emphasis filter: H(z) = 1 - 0.97z^-1
    signal = lfilter([1, -0.97], [1], signal)
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num_samples = len(signal)
num_frames = 1 + int(np.floor((num_samples - FRAME_SIZE) / HOP_SIZE))

frames = []
for i in range(num_frames):
    start = i * HOP_SIZE
    end = start + FRAME_SIZE
    # Apply Hamming window
    frame = signal[start:end] * hamming(FRAME_SIZE)
    frames.append(frame)

return signal, fs, frames, num_frames

# --- 4. LPC Analysis (Autocorrelation Method with Stabilization) ---
def lpc_analysis(frame, order, gamma):
    """
    Computes LPC coefficients and gain using Levinson-Durbin with
    Bandwidth Expansion (gamma) for filter stability.
    """
    R = np.correlate(frame, frame, mode='full')[len(frame) - 1 :]
    R = R[: order + 1]

    # Bandwidth Expansion for Stability: R[i] = R[i] * gamma**i
    if gamma != 1.0:
        for i in range(1, order + 1):
            R[i] = R[i] * (gamma ** i)

    a = np.zeros(order + 1)
    a[0] = 1.0
    E = R[0]

    # Levinson-Durbin Recursion
    for m in range(1, order + 1):
        k = R[m]
        for i in range(1, m):
            k -= a[i] * R[m - i]

        if E == 0:
            k = 0
        else:
            k /= E

        a_new = np.zeros(order + 1)
        a_new[0] = 1.0
        a_new[m] = k
        for i in range(1, m):
            a_new[i] = a[i] - k * a[m - i]

        a = a_new
        E *= (1 - k**2)

    G = np.sqrt(E) if E > 0 else 0
    return a, G

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# --- 5. Signal Reconstruction ---
def reconstruct_signal(frames, lpc_coefficients_list, G_list, num_samples):
    """Reconstructs the entire speech signal from frame-wise LPC coefficients.
    reconstructed_signal = np.zeros(num_samples)

    for i in range(len(frames)):
        start = i * HOP_SIZE
        end = start + FRAME_SIZE

        a = lpc_coefficients_list[i]
        G = G_list[i]

        # Excitation: Using scaled random noise
        excitation = np.random.normal(0, G, FRAME_SIZE)

        # Synthesis filter: H(z) = G / A(z).
        reconstructed_frame = lfilter([G], a, excitation)

        # Overlap-add
        reconstructed_signal[start:end] += reconstructed_frame

    return reconstructed_signal

# --- 6. Formant Estimation ---
def estimate_formants(a, fs):
    """Estimates formant frequencies from the roots of the LPC polynomial A(z)
    roots = np.roots(a)
    poles = roots[np.imag(roots) > 0]

    formants_hz = np.arctan2(np.imag(poles), np.real(poles)) * fs / (2 * np.pi)
    bandwidths_hz = -np.log(np.abs(poles)) * fs / np.pi

    formants = sorted([(F, B) for F, B in zip(formants_hz, bandwidths_hz)], key=lambda x: x[0])

    # Final Relaxed Filter: Bandwidth limit set to 1000 Hz
    meaningful_formants = [(F, B) for F, B in formants if F < fs/2 and B < 1000]

    return [f[0] for f in meaningful_formants]

# --- 7. Main Execution and Visualization ---
def run_lpc_analysis_lab():

    # --- Flexible File Acquisition Loop ---
    file_path = None
    while file_path is None:
        user_input = input("\nEnter the path to your WAV file (ideally downsampled):")

        if not user_input.lower().endswith('.wav'):
            print("Error: The file must be a '.wav' file. Please ensure the extension is correct")
            continue

        if os.path.exists(user_input):

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        file_path = user_input
        break

    directory = os.path.dirname(user_input)
    filename = os.path.basename(user_input)

    found_case_insensitive = False
    if os.path.isdir(directory) or directory == '':
        check_dir = directory if directory else '.'
        try:
            for item in os.listdir(check_dir):
                if item.lower() == filename.lower():
                    file_path = os.path.join(check_dir, item)
                    found_case_insensitive = True
                    print(f"File found! Using path: {file_path}")
                    break
        except FileNotFoundError:
            pass

    if found_case_insensitive:
        break

    print(f"Error: File not found at '{user_input}'. Check the path and file name.")

print(f"\nAnalyzing file: {file_path}")

try:
    original_signal, fs, frames, num_frames = load_and_frame_signal(file_path)
except Exception as e:
    print(f"Fatal Error during signal loading: {e}")
    return

print(f"Loaded signal with sampling rate: {fs} Hz. Total frames: {num_frames}")

# --- LPC Analysis and Reconstruction ---
lpc_coeffs_list = []
G_list = []

for frame in frames:
    a, G = lpc_analysis(frame, LPC_ORDER, gamma=STABILITY_GAMMA)
    lpc_coeffs_list.append(a)
    G_list.append(G)

avg_lpc_coeffs = np.mean(lpc_coeffs_list, axis=0)
avg_G = np.mean(G_list)

reconstructed_signal = reconstruct_signal(frames, lpc_coeffs_list, G_list, fs)

# --- Formant Estimation ---
estimated_formants = estimate_formants(avg_lpc_coeffs, fs)
print(f"\nEstimated Formant Frequencies (Hz): {estimated_formants}")

# --- Visualization ---

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# Plot 1: Original and Reconstructed Signals (Saving to file)
plt.figure(figsize=(12, 6))
time_axis = np.linspace(0, len(original_signal) / fs, len(original_signal))

plt.subplot(2, 1, 1)
plt.plot(time_axis, original_signal, color='blue')
plt.title(f'Original Speech Signal Waveform: {os.path.basename(file_path)}')
plt.ylabel('Amplitude')
plt.grid(True)

plt.subplot(2, 1, 2)
plt.plot(time_axis, reconstructed_signal, color='red')
plt.title('Reconstructed Speech Signal Waveform (LPC Synthesis)')
plt.xlabel('Time (s)')
plt.ylabel('Amplitude')
plt.grid(True)

plt.tight_layout()
plt.savefig('waveform_comparison.png')
plt.show() # Display plot

# Plot 2: Formant Frequencies (Frequency Response) (Saving to file)
plt.figure(figsize=(10, 5))

w, h = freqz(avg_G, avg_lpc_coeffs, worN=2048, fs=fs)

plt.plot(w, 20 * np.log10(abs(h)), color='green')

for i, F in enumerate(estimated_formants):
    label = f'Estimated F{i+1}'
    plt.axvline(F, color='red', linestyle='--', linewidth=1, label=label)

plt.title('Frequency Response (LPC Spectral Envelope) with Formants')
plt.xlabel('Frequency (Hz)')
plt.ylabel('Magnitude (dB)')
plt.xlim(0, fs / 2)
plt.grid(True, which='both', linestyle='--', linewidth=0.5)
plt.legend()
plt.tight_layout()
plt.savefig('spectral_envelope.png')
plt.show() # Display plot

# --- Comparison Table ---
print("\n--- Comparison of Estimated Formants with Standard Vowel Data ---")

if len(estimated_formants) >= 2:
    est_F1 = estimated_formants[0]
    est_F2 = estimated_formants[1]

    best_match_vowel = None
    min_distance = float('inf')

    for vowel, data in VOWEL_FORMANT_DATA.items():

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expected_F1 = data[0]
expected_F2 = data[1]

distance = np.sqrt((est_F1 - expected_F1)**2 + (est_F2 - expected_F2)**2)

if distance < min_distance:
    min_distance = distance
    best_match_vowel = vowel

print(f"\nEstimated F1: {est_F1:.2f} Hz | Estimated F2: {est_F2:.2f} Hz")
print(f"Closest Standard Vowel Match: /{best_match_vowel}/ (Distance: {min_distance:.2f} Hz)")

print("\n| Vowel (IPA) | Expected F1 (Hz) | Expected F2 (Hz) |")
print("|-----|-----|-----|")

sorted_vowels = sorted(VOWEL_FORMANT_DATA.items(), key=lambda x: x[1][0])
for vowel, data in sorted_vowels:
    print(f"| {vowel:11} | {data[0]:16.0f} | {data[1]:16.0f} |")

print("|-----|-----|-----|")
print(f"| {'Estimated':11} | {est_F1:16.2f} | {est_F2:16.2f} | <- You can compare these values to the ones above")
else:
    print("Not enough formants (F1, F2) were estimated for a proper comparison")

# --- 8. Save Reconstructed Signal to a WAV file ---
output_filename = os.path.splitext(os.path.basename(file_path))[0] + "_reconstructed.wav"

try:
    max_original_amp = np.max(np.abs(original_signal))
    max_reco_amp = np.max(np.abs(reconstructed_signal))

    if max_reco_amp > 0:
        scaled_reconstructed_signal = reconstructed_signal / max_reco_amp
    else:
        scaled_reconstructed_signal = reconstructed_signal

    sf.write(output_filename, scaled_reconstructed_signal, fs)
    print(f"\nSuccessfully saved reconstructed signal to: {output_filename}")

except Exception as e:
    print(f"\nError saving reconstructed file: {e}")

# --- Run the Lab ---
if __name__ == "__main__":
    run_lpc_analysis_lab()

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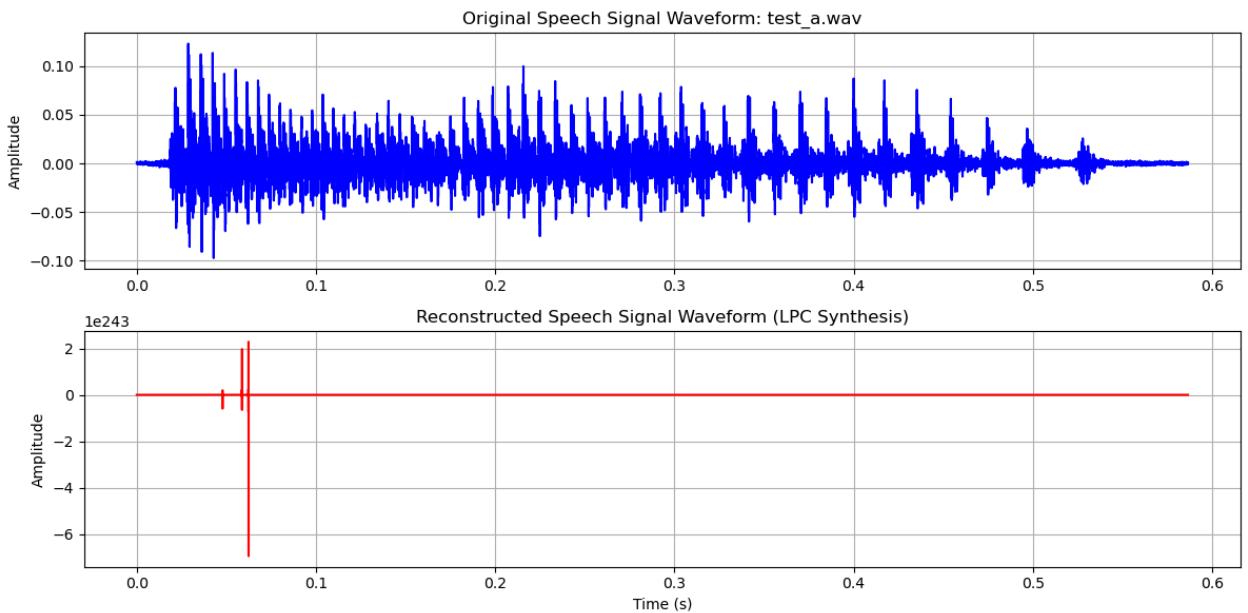
Analyzing file: test_a.wav

*** WARNING: High Sampling Rate Detected! ***

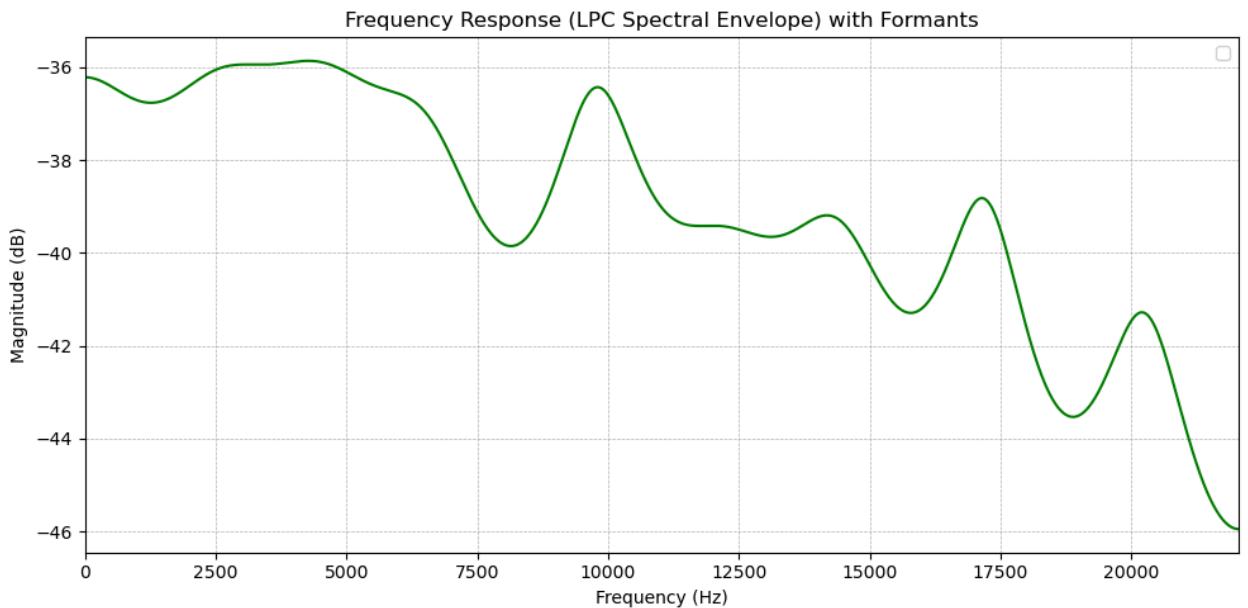
The input file has $F_s=44100$ Hz. For accurate formant analysis, please downsample the file to 16000 Hz.

Loaded signal with sampling rate: 44100 Hz. Total frames: 159

Estimated Formant Frequencies (Hz): []



C:\Users\Joel\AppData\Local\Temp\ipykernel_29360\254851941.py:250: UserWarning: No artists with labels found to put in legend. Note that artists whose label start with an underscore are ignored when legend() is called with no argument.
plt.legend()



--- Comparison of Estimated Formants with Standard Vowel Data ---
Not enough formants (F1, F2) were estimated for a proper comparison. Please ensure your input file is a clean, sustained vowel sampled at approximately 16000 Hz.

Successfully saved reconstructed signal to: test_a_reconstructed.wav

In []: