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**Repository GitHub:** [Github](#)

Notebook ini mencakup seluruh penggeraan:

1. Rekaman & Analisis Suara Multi-Level
  2. Noise Reduction dengan Filtering
  3. Pitch Shifting & Audio Manipulation
  4. Audio Processing Chain (target loudness: **-16 LUFS**)
  5. Music Analysis & Remix
- 

## Import dan Setup Folder

```
In [3]: import os, math, io, copy, itertools, json, warnings
from pathlib import Path

import numpy as np
import matplotlib.pyplot as plt
import librosa, librosa.display
import scipy.signal as sps
import soundfile as sf

from pydub import AudioSegment
import pyloudnorm as pyln

warnings.filterwarnings('ignore')

BASE_DIR = Path.cwd()
AUDIO_DIR = BASE_DIR / 'audio'
AUDIO_DIR.mkdir(exist_ok=True)

print('BASE_DIR:', BASE_DIR)
print('AUDIO_DIR:', AUDIO_DIR)
```

```
BASE_DIR: d:\Coding\Sistem-Teknologi-Multimedia\Worksheet-3
AUDIO_DIR: d:\Coding\Sistem-Teknologi-Multimedia\Worksheet-3\audio
```

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## Utilities (I/O, DSP Helpers, Filtering)

```
In [4]: # Fungsi untuk membaca file audio dan mengembalikan data dalam bentuk array numpy
def load_audio(path, sr=None, mono=True):
    y, sr_ = librosa.load(path, sr=sr, mono=mono)
    return y.astype(np.float32), sr_
```

```

# Fungsi untuk menyimpan data audio (array numpy) menjadi file .wav
def save_wav(path, y, sr):
    sf.write(str(path), y, sr)
    return str(path)

# Fungsi untuk mengubah data numpy menjadi AudioSegment (format pydub)
# Berguna untuk pemrosesan tambahan seperti fade in/out, gabung audio, dll
def np_to_audiosegment(y, sr):
    y = np.asarray(y)
    y16 = np.int16(np.clip(y, -1.0, 1.0) * 32767) # konversi ke 16-bit PCM
    if y16.ndim == 1:
        channels = 1
        raw = y16.tobytes()
    else:
        if y16.shape[1] in (1,2):
            channels = y16.shape[1]
            raw = y16.reshape((-1,), order='C').tobytes()
        else:
            channels = 1
            raw = y16[:,0].tobytes()
    return AudioSegment(data=raw, sample_width=2, frame_rate=sr, channels=channels)

# Fungsi kebalikan dari np_to_audiosegment, yaitu mengubah AudioSegment menjadi array numpy
def audiosegment_to_np(seg):
    samples = np.array(seg.get_array_of_samples()).astype(np.int16)
    if seg.channels == 1:
        y = samples.astype(np.float32) / 32768.0
        return y, seg.frame_rate
    else:
        y = samples.reshape((-1, seg.channels)).astype(np.float32) / 32768.0
        return y, seg.frame_rate

# Fungsi untuk menampilkan waveform dan spektrogram dari sinyal audio
def plot_waveform_and_spectrogram(y, sr, title=''):
    fig, ax = plt.subplots(2, 1, figsize=(12, 6))
    # Menampilkan waveform (amplitudo terhadap waktu)
    times = np.arange(len(y)) / sr
    ax[0].plot(times, y)
    ax[0].set_title(f'Waveform {title}')
    ax[0].set_xlabel('Time (s)')
    ax[0].set_ylabel('Amplitude')
    ax[0].grid(True, alpha=0.3)

    # Menampilkan spektrogram (frekuensi terhadap waktu dalam skala log)
    S = np.abs(librosa.stft(y, n_fft=2048, hop_length=512))**2
    S_db = librosa.power_to_db(S, ref=np.max)
    img = librosa.display.specshow(S_db, sr=sr, hop_length=512, x_axis='time', y_axis='log')
    ax[1].set_title(f'Spectrogram (log-f) {title}')
    fig.colorbar(img, ax=ax[1], format='%.2f dB')
    plt.tight_layout(); plt.show()

# Fungsi untuk menerapkan filter Butterworth (Low-pass, high-pass, atau band-pass)
def butter_filter(y, sr, ftype='low', cutoff=1000.0, order=6, band=None):
    nyq = 0.5 * sr # frekuensi Nyquist
    if ftype == 'low':
        Wn = cutoff / nyq
        b, a = sps.butter(order, Wn, btype='low', analog=False)
    elif ftype == 'high':
        Wn = cutoff / nyq
        b, a = sps.butter(order, Wn, btype='high', analog=False)

```

```

    elif ftype == 'band':
        low = band[0] / nyq
        high = band[1] / nyq
        b, a = sps.butter(order, [low, high], btype='bandpass', analog=False)
    else:
        raise ValueError('Invalid ftype')
    return sps.filtfilt(b, a, y) # menggunakan filtfilt agar hasil lebih halus

# Fungsi untuk mempercepat atau memperlambat audio tanpa mengubah pitch
def time_stretch(y, rate):
    return librosa.effects.time_stretch(y, rate=rate)

# Fungsi untuk mengubah tinggi nada (pitch) tanpa mengubah durasi audio
def pitch_shift(y, sr, n_steps):
    return librosa.effects.pitch_shift(y, sr=sr, n_steps=n_steps)

# Fungsi untuk menormalkan loudness audio berdasarkan standar LUFS
def lufs_normalize(y, sr, target_lufs=-16.0):
    meter = pyln.Meter(sr)
    loudness = meter.integrated_loudness(y)
    gain = target_lufs - loudness
    y_norm = pyln.normalize.loudness(y, loudness, target_lufs)
    return y_norm, loudness, gain

# Fungsi untuk normalisasi puncak amplitudo (peak normalization)
def peak_normalize(y, peak=0.99):
    max_abs = np.max(np.abs(y)) + 1e-9
    return (y / max_abs) * peak

# Fungsi untuk melakukan dynamic range compression
# Menekan bagian keras dan menyeimbangkan dinamika audio
def compressor(y, sr, threshold_db=-20.0, ratio=4.0, attack_ms=10.0, release_ms=100.0):
    win = int(sr * 0.05)
    if win < 1: win = 1
    pad = (win//2, win - win//2)
    y_pad = np.pad(y, pad, mode='reflect')
    rms = np.sqrt(sps.convolve(y_pad**2, np.ones(win)/win, mode='valid'))
    rms_db = 20*np.log10(np.maximum(rms, 1e-9))
    over_db = np.maximum(rms_db - threshold_db, 0.0)
    gain_reduction_db = over_db * (1.0 - 1.0/ratio)

    # Pengaturan attack dan release time
    a = np.exp(-1.0 / (sr * (attack_ms/1000.0)))
    r = np.exp(-1.0 / (sr * (release_ms/1000.0)))
    gr_sm = np.zeros_like(gain_reduction_db)

    # Perhitungan gain reduction dengan smoothing
    for i, g in enumerate(gain_reduction_db):
        if i == 0:
            gr_sm[i] = g
        else:
            if g > gr_sm[i-1]:
                gr_sm[i] = a * gr_sm[i-1] + (1-a) * g
            else:
                gr_sm[i] = r * gr_sm[i-1] + (1-r) * g

    gr_lin = 10 ** (-gr_sm/20.0)

    # FIX: Pastikan ukuran sama dengan menggunakan minimum length
    min_len = min(len(y), len(gr_lin))

```

```

y_out = y[:min_len] * gr_lin[:min_len]
y_out *= 10 ** (makeup_db/20.0)

# Pad jika y_out lebih pendek dari y original
if len(y_out) < len(y):
    y_out = np.pad(y_out, (0, len(y)-len(y_out)))
return y_out

# Fungsi untuk menghilangkan bagian sinyal dengan level di bawah ambang batas te
def noise_gate(y, threshold_db=-50.0):
    win = 1024
    eps = 1e-9
    y2 = y.copy()
    for i in range(0, len(y), win):
        frame = y[i:i+win]
        rms = np.sqrt(np.mean(frame**2) + eps)
        db = 20*np.log10(rms + eps)
        if db < threshold_db:
            y2[i:i+win] = 0.0 # nol-kan frame yang terlalu pelan
    return y2

# Fungsi untuk memotong bagian diam (silence) di awal dan akhir rekaman
def trim_silence(y, top_db=40):
    yt, idx = librosa.effects.trim(y, top_db=top_db)
    return yt, idx

# Fungsi untuk memperkirakan tempo (BPM) dari sinyal audio
def estimate_bpm(y, sr):
    tempo = librosa.beat.tempo(y=y, sr=sr, hop_length=512, aggregate=np.median)
    return float(tempo[0])

# Fungsi untuk mendeteksi nada dasar (key) musik dari sinyal audio
def estimate_key(y, sr):
    chroma = librosa.feature.chroma_cqt(y=y, sr=sr)
    chroma = chroma.mean(axis=1)
    chroma = chroma / (np.max(chroma) + 1e-9)

# Template profil mayor dan minor untuk perbandingan
major = np.array([6.35, 2.23, 3.48, 2.33, 4.38, 4.09, 2.52, 5.19, 2.39, 3.66, 2.29, 2.8
minor = np.array([6.33, 2.68, 3.52, 5.38, 2.60, 3.53, 2.54, 4.75, 3.98, 2.69, 3.34, 3.1
major = major / major.max()
minor = minor / minor.max()
keys = ['C', 'C#', 'D', 'D#', 'E', 'F', 'F#', 'G', 'G#', 'A', 'A#', 'B']

# Bandingkan skor tiap key dengan profil mayor/minor
best_score = -1.0
best = ('C', 'major')
for i in range(12):
    maj_score = np.dot(chroma, np.roll(major, i))
    min_score = np.dot(chroma, np.roll(minor, i))
    if maj_score >= min_score and maj_score > best_score:
        best_score = maj_score; best = (keys[i], 'major')
    elif min_score > maj_score and min_score > best_score:
        best_score = min_score; best = (keys[i], 'minor')
return best

```

No	Nama Fungsi	Tujuan / Kegunaan Utama	Keterangan Singkat
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No	Nama Fungsi	Tujuan / Kegunaan Utama	Keterangan Singkat
1	<code>load_audio()</code>	Membaca file audio	Mengembalikan data audio dan sampling rate
2	<code>save_wav()</code>	Menyimpan audio ke file .wav	Menggunakan library <code>soundfile</code>
3	<code>np_to_audiosegment()</code>	Konversi NumPy → AudioSegment	Untuk efek seperti fade atau crossfade
4	<code>audiosegment_to_np()</code>	Konversi AudioSegment → NumPy	Kebalikan dari fungsi sebelumnya
5	<code>plot_waveform_and_spectrogram()</code>	Visualisasi sinyal audio	Menampilkan waveform dan spectrogram
6	<code>butter_filter()</code>	Filtering audio (low/high/band-pass)	Menghapus noise di frekuensi tertentu
7	<code>time_stretch()</code>	Mengubah kecepatan audio	Tidak mengubah pitch
8	<code>pitch_shift()</code>	Mengubah tinggi nada (pitch)	Tidak mengubah durasi
9	<code>lufs_normalize()</code>	Normalisasi loudness (LUFS)	Menyeimbangkan volume sesuai persepsi manusia
10	<code>peak_normalize()</code>	Normalisasi berdasarkan puncak amplitudo	Menjaga level tertinggi audio di bawah clipping
11	<code>compressor()</code>	Meratakan dinamika suara	Mengecilkan bagian keras dan memperjelas bagian lembut
12	<code>noise_gate()</code>	Menghilangkan noise saat jeda	Menutup bagian dengan volume di bawah threshold
13	<code>trim_silence()</code>	Memotong bagian diam	Membersihkan awal dan akhir rekaman
14	<code>estimate_bpm()</code>	Deteksi tempo lagu (BPM)	Mengukur kecepatan ketukan
15	<code>estimate_key()</code>	Deteksi nada dasar musik	Mengidentifikasi key mayor/minor

## Soal 1. Rekaman dan Analisis Suara Multi-

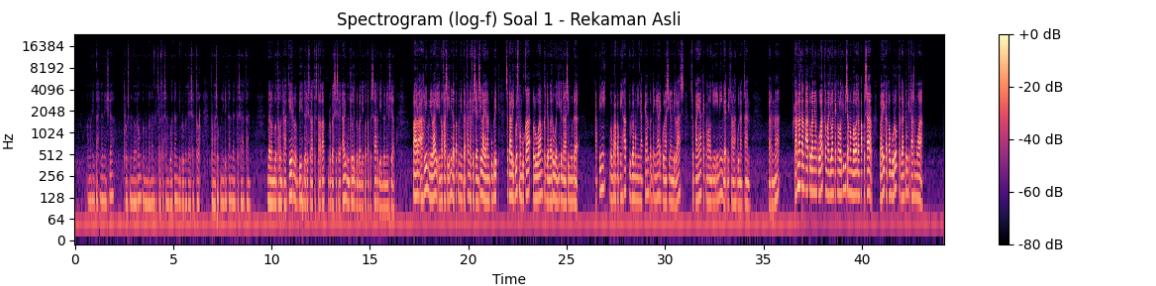
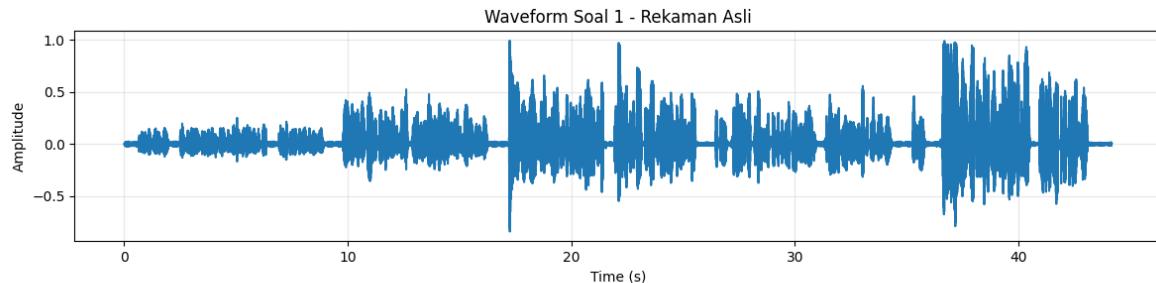
# Level

```
In [10]: path_soal1 = AUDIO_DIR / 'Berita.wav'
assert path_soal1.exists(), f'File tidak ditemukan: {path_soal1}. Silakan taruh
y1, sr1 = load_audio(path_soal1, sr=None, mono=True)
print(f'Loaded: {path_soal1.name}, durasi: {len(y1)/sr1:.2f}s, sr: {sr1}')

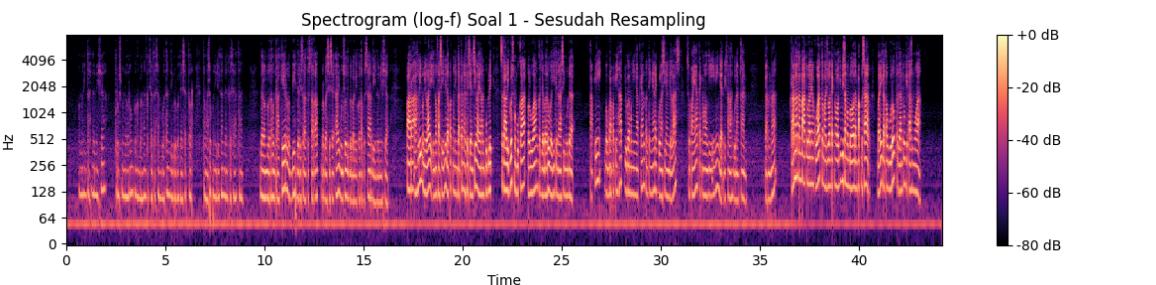
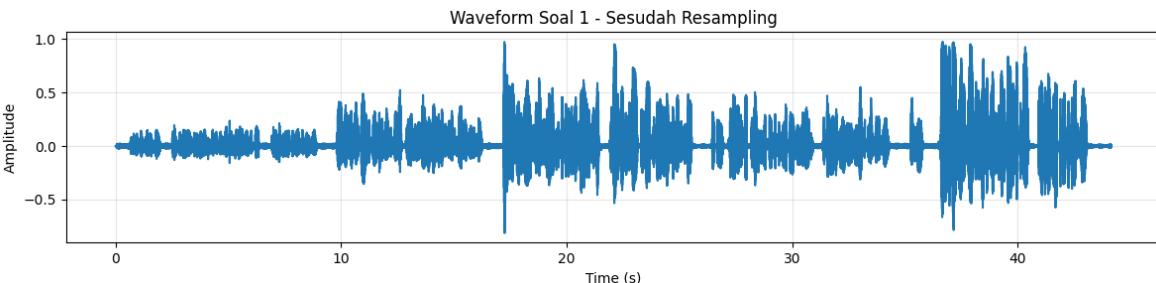
plot_waveform_and_spectrogram(y1, sr1, title='Soal 1 - Rekaman Asli')

target_sr = 16000
# Use scipy resampling to avoid resampy dependency
y1_rs = librosa.resample(y=y1, orig_sr=sr1, target_sr=target_sr, res_type='soxr_
save_wav(AUDIO_DIR / 'soal1_resampled.wav', y1_rs, target_sr)
print('Sesudah resampling:', len(y1_rs)/target_sr, 'detik @', target_sr, 'Hz')
plot_waveform_and_spectrogram(y1_rs, target_sr, title='Soal 1 - Sesudah Resampli
```

Loaded: Berita.wav, durasi: 44.16s, sr: 48000



Sesudah resampling: 44.16 detik @ 16000 Hz



Setelah dibandingkan, bentuk waveform dari rekaman asli dan hasil resampling kelihatan hampir sama yang mana urutan dan dinamika suaranya ga berubah. Bedanya cuma ada di bagian frekuensi: rekaman asli punya spektrum sampai sekitar 16 kHz, sedangkan hasil resampling cuma sampai 4 kHz karena frekuensi samplingnya diturunkan jadi 16 kHz.

Akibatnya, suara jadi sedikit kehilangan detail di High Frequency, tapi bagian utama suara manusia (sekitar 100–3000 Hz) tetap jelas (Bisa dilihat pada spectrogram intensitas warna pada frekuensi tersebut cukup terang)

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## Soal 2. Noise Reduction dengan Filtering

```
In [16]: path_soal2 = AUDIO_DIR / '1015.wav'
assert path_soal2.exists(), f'File tidak ditemukan: {path_soal2}. Letakkan rekaman'
y2, sr2 = load_audio(path_soal2, sr=None, mono=True)
print(f'Loaded: {path_soal2.name}, durasi: {len(y2)/sr2:.2f}s, sr: {sr2}')

plot_waveform_and_spectrogram(y2, sr2, title='Soal 2 - Audio Original (Noisy)')

# High-Pass Filter - Paling efektif untuk menghilangkan noise rendah (hum, rumble)
print("\n==== HIGH-PASS FILTER (Menghilangkan Noise Rendah) ===")
hp_cutoffs = [300, 500, 1000]

for c in hp_cutoffs:
    y_hp = butter_filter(y2, sr2, ftype='high', cutoff=c, order=6)
    save_wav(AUDIO_DIR / f'soal2_highpass_{c}Hz.wav', y_hp, sr2)
    plot_waveform_and_spectrogram(y_hp, sr2, title=f'Soal 2 - High-pass {c}Hz')
    print(f"✓ Saved: soal2_highpass_{c}Hz.wav")

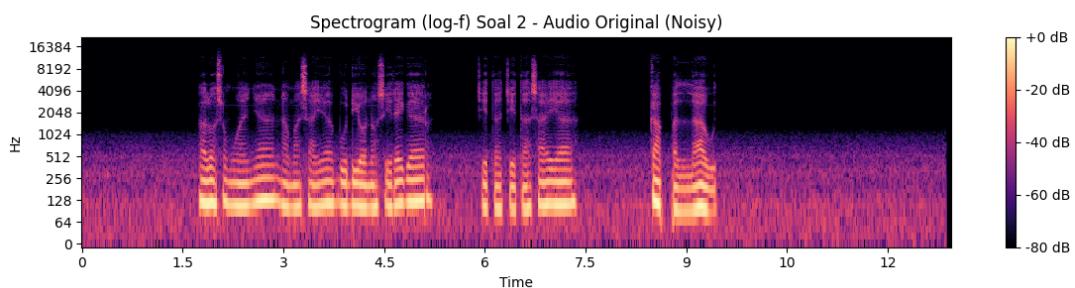
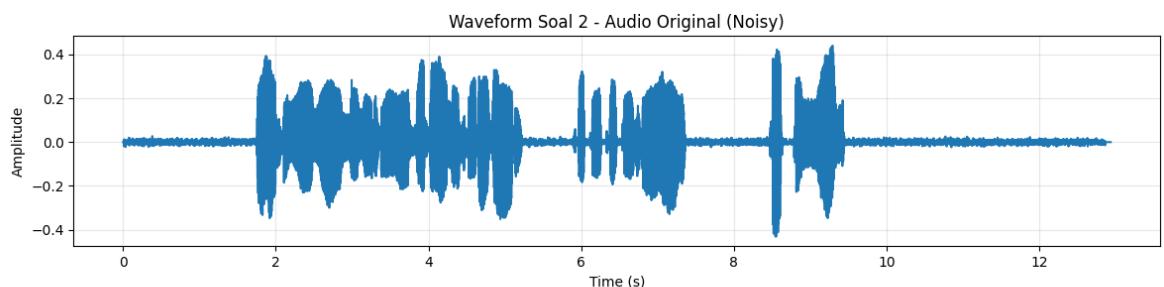
# Low-Pass Filter - Untuk menghilangkan noise tinggi (hiss, white noise)
print("\n==== LOW-PASS FILTER (Menghilangkan Noise Tinggi) ===")
lp_cutoffs = [3400, 4000, 8000]

for c in lp_cutoffs:
    y_lp = butter_filter(y2, sr2, ftype='low', cutoff=c, order=6)
    save_wav(AUDIO_DIR / f'soal2_lowpass_{c}Hz.wav', y_lp, sr2)
    plot_waveform_and_spectrogram(y_lp, sr2, title=f'Soal 2 - Low-pass {c}Hz')
    print(f"✓ Saved: soal2_lowpass_{c}Hz.wav")

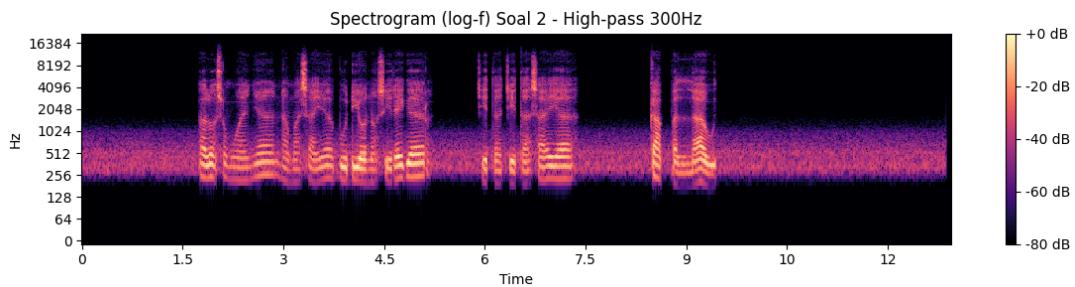
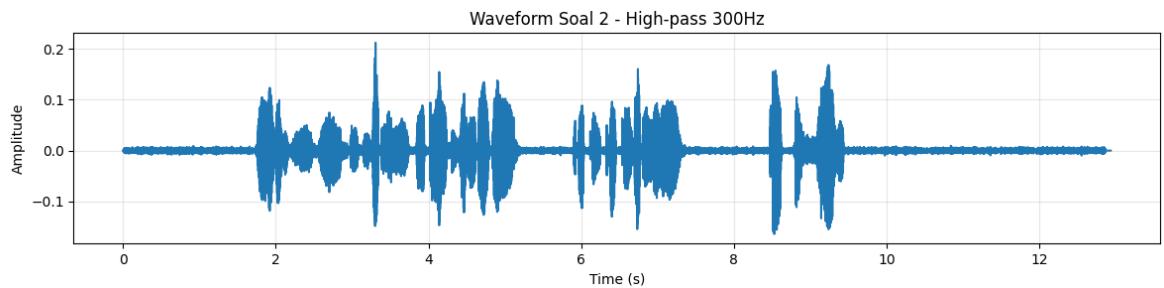
# Band-Pass Filter - Kombinasi terbaik untuk speech (fokus ke frekuensi vokal)
print("\n==== BAND-PASS FILTER (Fokus ke Frekuensi Speech) ===")
bands = [(300, 3400), (200, 4000), (100, 8000)]

for b in bands:
    y_bp = butter_filter(y2, sr2, ftype='band', band=b, order=6)
    save_wav(AUDIO_DIR / f'soal2_bandpass_{b[0]}-{b[1]}Hz.wav', y_bp, sr2)
    plot_waveform_and_spectrogram(y_bp, sr2, title=f'Soal 2 - Band-pass {b[0]}-{b[1]}Hz')
    print(f"✓ Saved: soal2_bandpass_{b[0]}-{b[1]}Hz.wav")
```

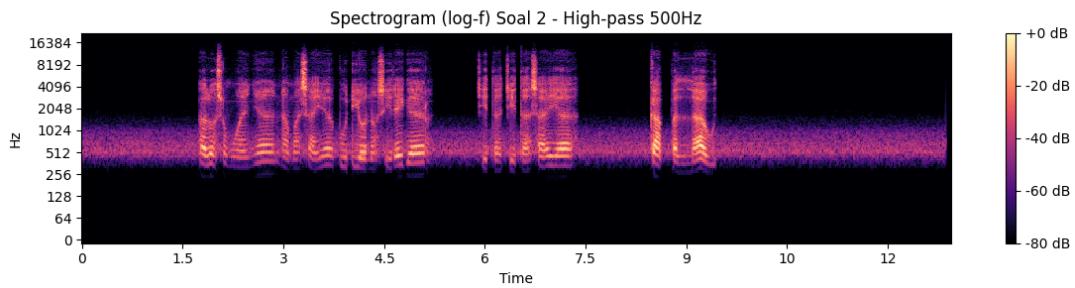
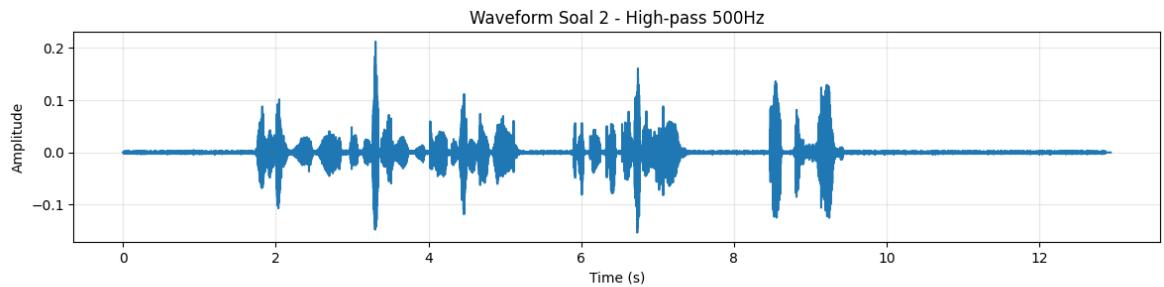
Loaded: 1015.wav, durasi: 12.93s, sr: 44100



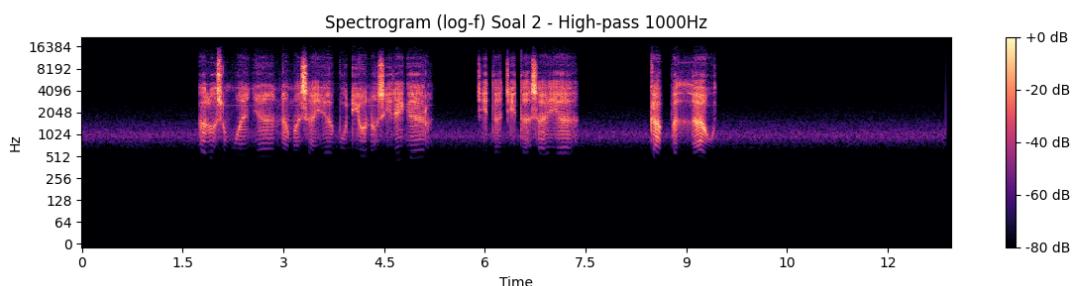
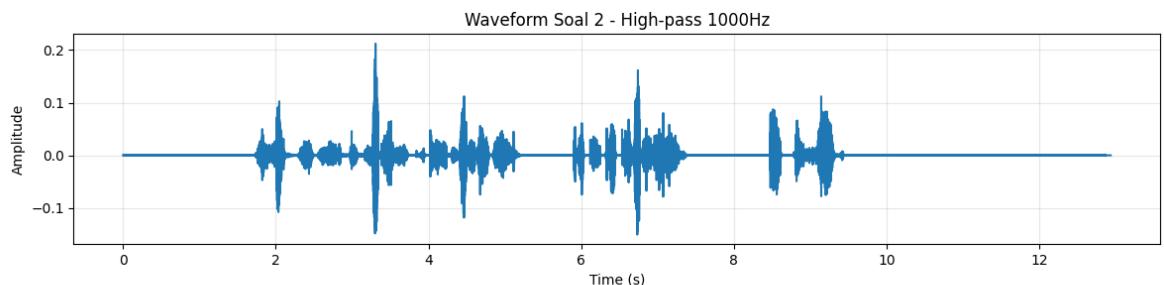
==== HIGH-PASS FILTER (Menghilangkan Noise Rendah) ====



✓ Saved: soal2\_highpass\_300Hz.wav

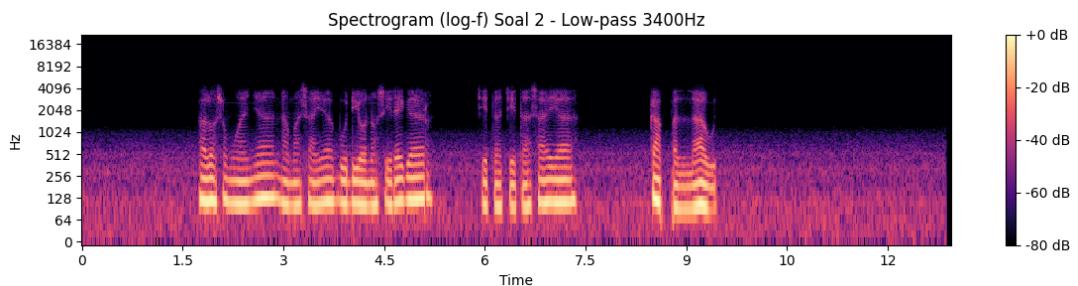
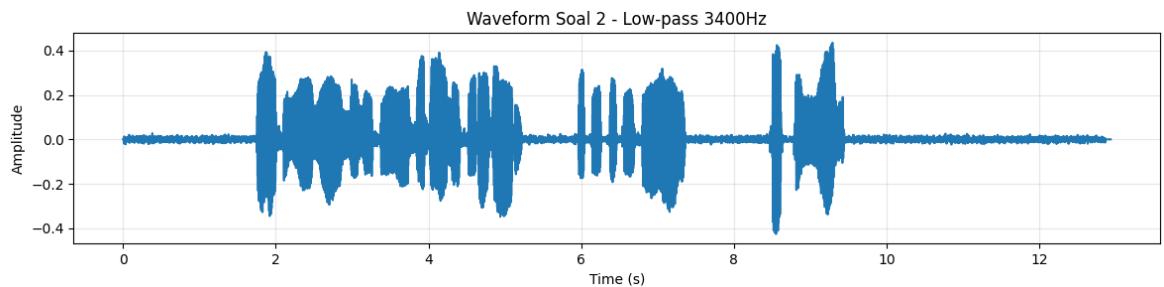


✓ Saved: soal2\_highpass\_500Hz.wav

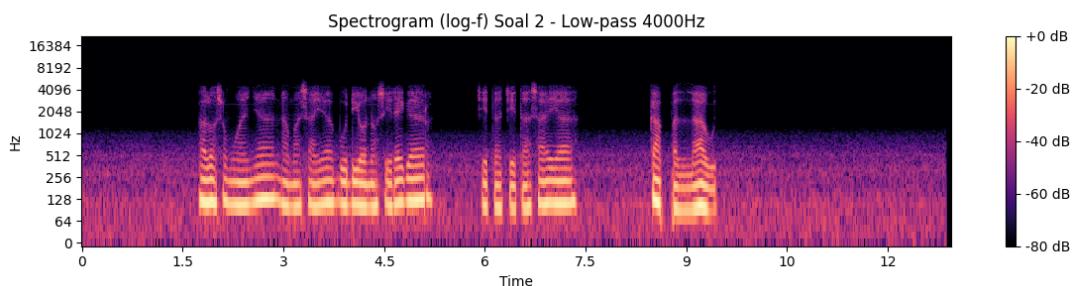
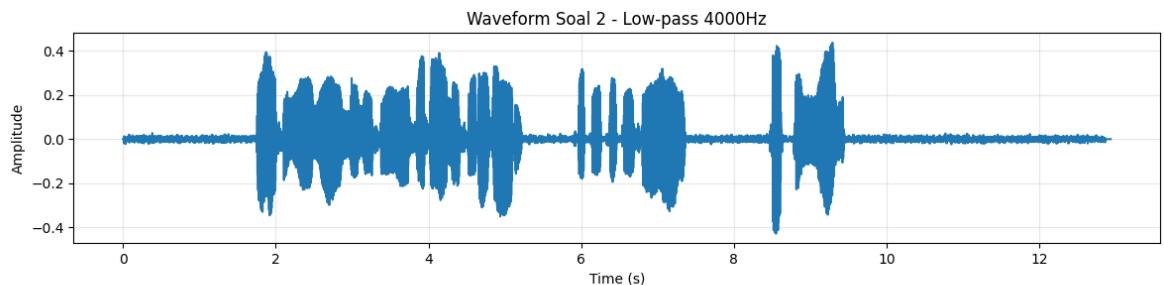


✓ Saved: soal2\_highpass\_1000Hz.wav

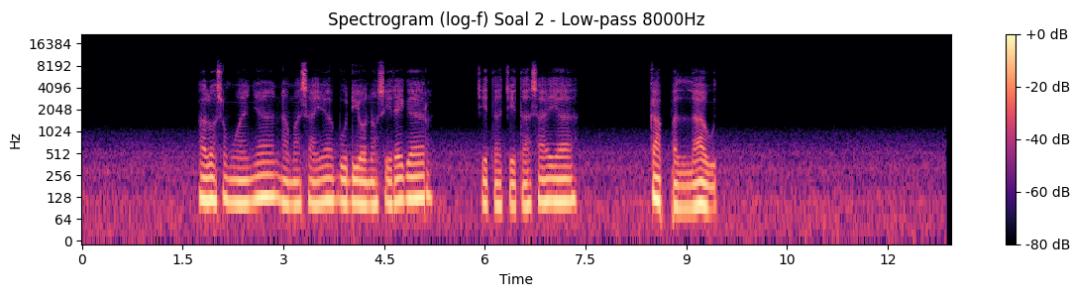
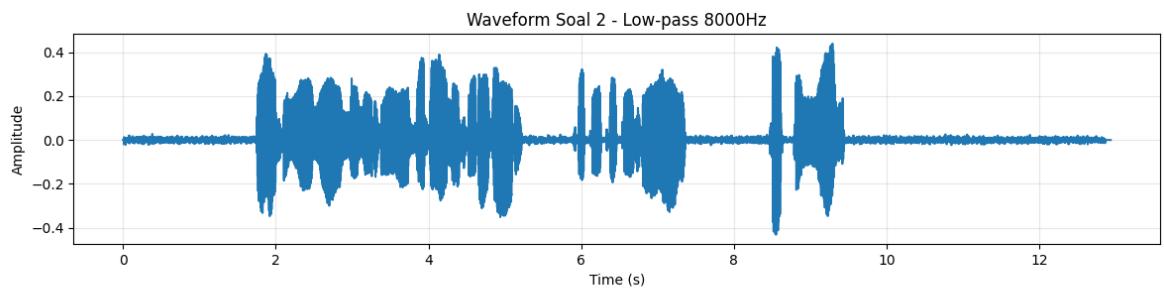
==== LOW-PASS FILTER (Menghilangkan Noise Tinggi) ====



✓ Saved: soal2\_lowpass\_3400Hz.wav

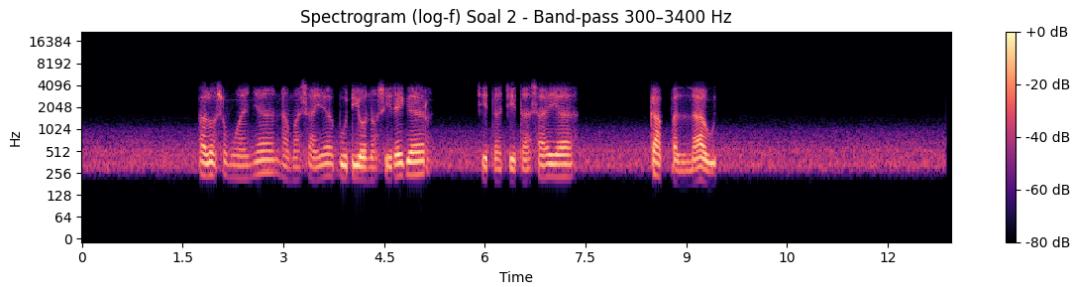
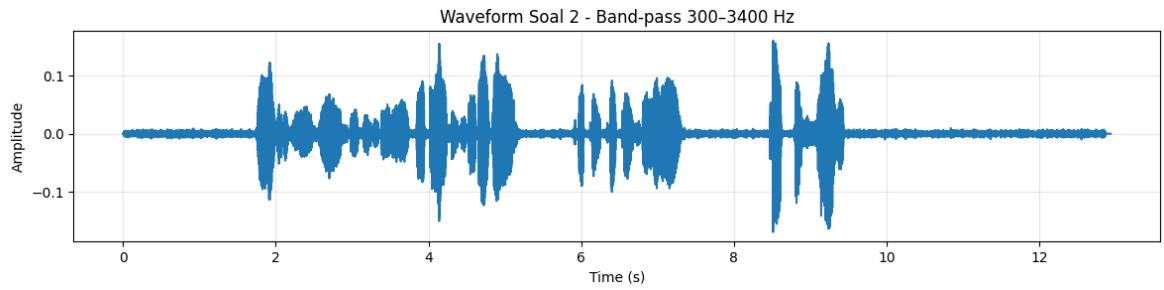


✓ Saved: soal2\_lowpass\_4000Hz.wav

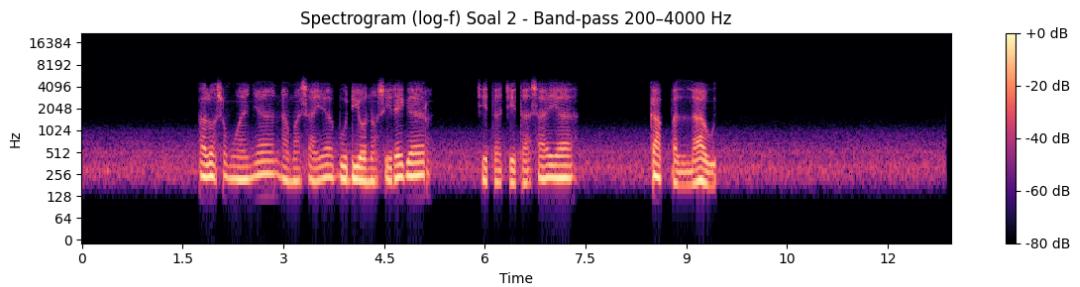
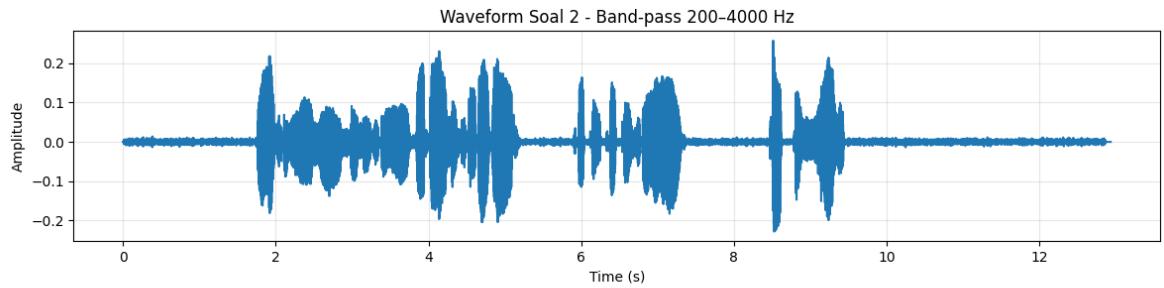


✓ Saved: soal2\_lowpass\_8000Hz.wav

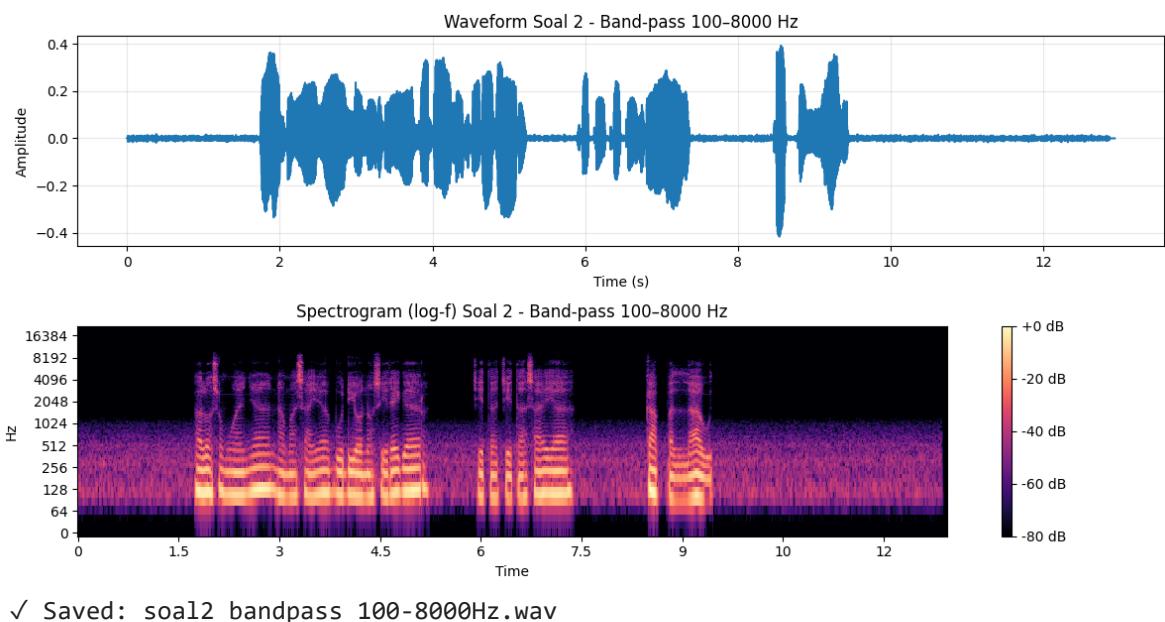
==== BAND-PASS FILTER (Fokus ke Frekuensi Speech) ===



✓ Saved: soal2\_bandpass\_300-3400Hz.wav



✓ Saved: soal2\_bandpass\_200-4000Hz.wav



## Analisis Hasil Filtering Audio

### 1. Jenis Noise

Noise yang digunakan pada percobaan ini adalah **white noise**, yaitu jenis noise yang memiliki energi menyebar merata di seluruh frekuensi. Karena sifatnya yang menyebar luas, white noise terdengar seperti desisan konstan dan cukup sulit dihilangkan hanya dengan satu jenis filter. Maka dari itu, pengujian dilakukan dengan beberapa tipe filter (high-pass, low-pass, dan band-pass) untuk melihat mana yang paling efektif mereduksi noise tanpa merusak kualitas suara

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### 2. Analisis Per Filter

#### a. High-pass Filter

Filter high-pass bekerja dengan menghilangkan frekuensi rendah dan hanya membiarkan frekuensi tinggi lewat. Dari hasil pengamatan, filter ini cukup berhasil mengurangi dengungan dan getaran di bagian bawah (sekitar 0–200 Hz). Namun, ketika cutoff dinaikkan hingga 500 Hz atau 1000 Hz, suara saya mulai kehilangan karakter alaminya. Ucapan terdengar tipis karena sebagian besar energi suara saya berada di bawah 500 Hz

#### b. Low-pass Filter

Low-pass filter berfungsi sebaliknya, yaitu menahan frekuensi tinggi dan membiarkan frekuensi rendah lewat. Filter ini cukup efektif untuk mengurangi desisan (hiss) dari white noise pada area frekuensi tinggi. Namun, hasilnya masih menyisakan noise rendah, dan meski ucapan tetap terdengar, suara jadi **boomy**. Secara umum, low-pass filter kurang efektif jika digunakan sendiri untuk white noise karena gangguannya muncul di seluruh

spektrum

### c. Band-pass Filter

Band-pass filter mengizinkan frekuensi di antara dua nilai cutoff tertentu untuk lewat dan menahan sisanya. Filter ini terbukti paling efektif untuk sinyal ucapan. Pada rentang **300–3400 Hz**, noise di frekuensi rendah dan tinggi dapat ditekan secara bersamaan. Hasilnya, suara terdengar jauh lebih bersih tanpa banyak kehilangan kejelasan

---

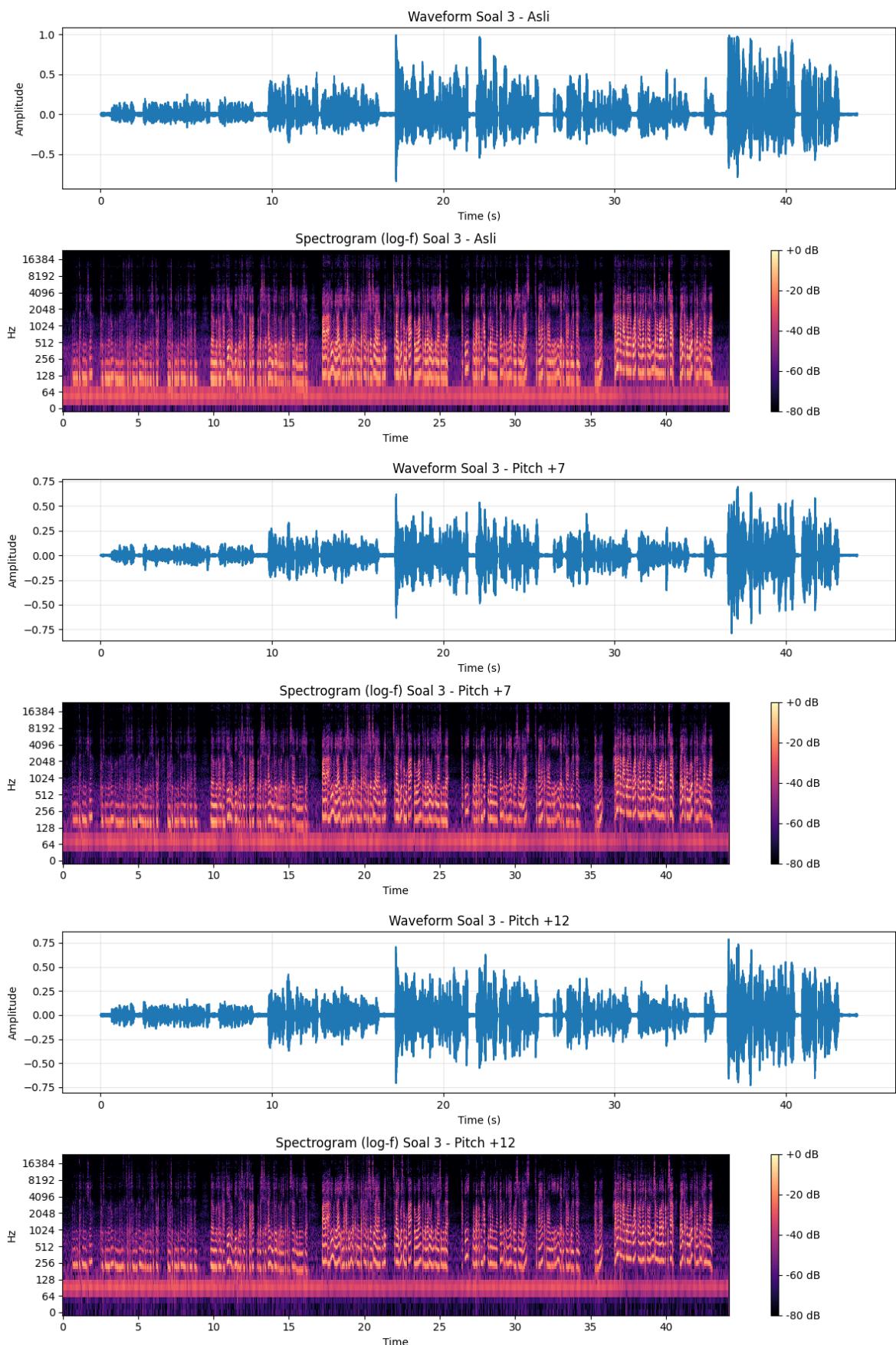
#### TLDR

- Dari semua percobaan, band-pass filter dengan rentang 300–3400 Hz adalah yang paling efektif untuk mereduksi white noise
  - Filter ini mampu menekan dengung di frekuensi rendah dan desisan di frekuensi tinggi sekaligus, tanpa mengubah karakter utama suara
  - Sementara itu, high-pass hanya bagus untuk menghilangkan noise rendah tapi bikin suara jadi tipis, dan low-pass mengurangi noise tinggi tapi membuat suara jadi boomy dan kurang jernih
- 

## Soal 3. Pitch Shifting dan Audio Manipulation

```
In [19]: y3, sr3 = load_audio(AUDIO_DIR / 'Berita.wav', sr=None, mono=True)
y3_p7 = pitch_shift(y3, sr3, n_steps=+7)
y3_p12 = pitch_shift(y3, sr3, n_steps=+12)
save_wav(AUDIO_DIR / 'soal3_pitch_+7.wav', y3_p7, sr3)
save_wav(AUDIO_DIR / 'soal3_pitch_+12.wav', y3_p12, sr3)
plot_waveform_and_spectrogram(y3, sr3, title='Soal 3 - Asli')
plot_waveform_and_spectrogram(y3_p7, sr3, title='Soal 3 - Pitch +7')
plot_waveform_and_spectrogram(y3_p12, sr3, title='Soal 3 - Pitch +12')

seg_p7 = np_to_audiosegment(y3_p7, sr3)
seg_p12 = np_to_audiosegment(y3_p12, sr3)
mix = seg_p7.append(seg_p12, crossfade=500)
mix_path = AUDIO_DIR / 'soal3_mix_p7_p12.wav'
mix.export(mix_path, format='wav')
```



```
Out[19]: <_io.BufferedReader name='d:\\Coding\\Sistem-Teknologi-Multimedia\\Worksheet-3\\audio\\soal3_mix_p7_p12.wav'>
```

## Analisis Soal No.3 Pitch Shifting

## 1. Parameter yang Dipakai

- **Fungsi:** librosa.effects.pitch\_shift()
  - **Parameter utama:**
    - y3 → data audio asli
    - sr3 → sampling rate audio
    - n\_steps = +7 dan +12 → menaikkan pitch sebanyak 7 dan 12 semitone (1 oktaf = 12 semitone)
  - **Tujuan:** buat naikin tinggi nada suara tanpa ngubah kecepatannya (durasi tetep sama).
- 

## 2. Perbedaan Visual

Jenis Audio	Ciri Waveform	Ciri Spektrogram
Asli	Amplitudo seimbang antara frekuensi rendah dan tinggi	Energi dominan di sekitar 100–4000 Hz (rentang umum suara manusia)
Pitch +7	Bentuk gelombang masih sama, tapi pola harmoniknya agak geser ke atas	Frekuensi dominan naik sekitar setengah oktaf
Pitch +12	Waveform kelihatan mirip tapi sedikit lebih rapat	Energi geser lebih tinggi (sekitar satu oktaf), warna terang makin ke atas

---

## 3. Dampak ke Kualitas Suara

- **Pitch +7:** suara jadi lebih tinggi, tapi masih cukup jelas dan masih cukup enak didengar
  - **Pitch +12:** nada makin tinggi kayak suara chipmunk, agak nyaring tapi jadi kurang natural
  - **Intinya:** makin besar nilai n\_steps , makin tinggi nadanya tapi bisa bikin suara terasa aneh
- 

**Soal 4 .Audio Processing Chain (EQ → Fade/Gain → Normalization → Compression → Noise Gate → Silence Trim)**

```
In [5]: src_path = AUDIO_DIR / 'soal3_mix_p7_p12.wav'
assert src_path.exists(), 'Jalankan Soal 3 terlebih dahulu.'
y4, sr4 = load_audio(src_path, sr=None, mono=True)
plot_waveform_and_spectrogram(y4, sr4, title='Soal 4 - Sebelum Processing')
```

```

y4_eq_lowcut = butter_filter(y4, sr4, ftype='high', cutoff=80, order=4)
y4_eq_band = butter_filter(y4_eq_lowcut, sr4, ftype='band', band=(100, 8000),
y4_eq = y4_eq_band

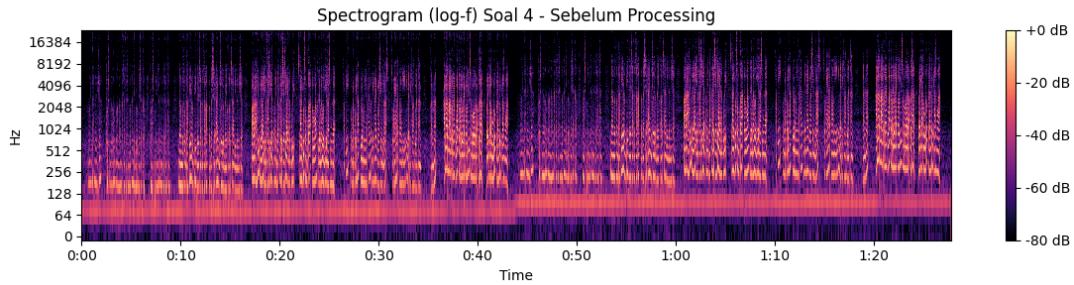
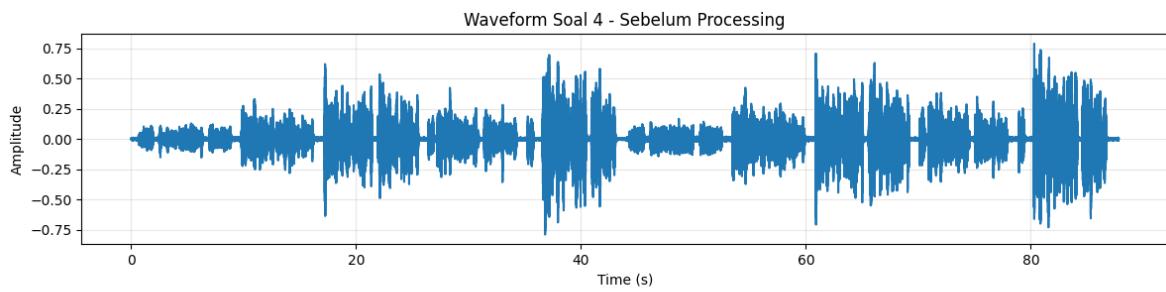
seg = np_to_audiosegment(y4_eq, sr4)
seg = seg.fade_in(200).fade_out(300)
y4_eq_fade, _ = audiosegment_to_np(seg)

y4_lufs, lufs_before, gain_db = lufs_normalize(y4_eq_fade, sr4, target_lufs=-16.
print(f'LUFS sebelum: {lufs_before:.2f} dB, gain kompensasi: {gain_db:.2f} dB')

y4_comp = compressor(y4_lufs, sr4, threshold_db=-20.0, ratio=3.0, attack_ms=8.0,
y4_gate = noise_gate(y4_comp, threshold_db=-55.0)
y4_final, idx = trim_silence(y4_gate, top_db=40)

out_path = AUDIO_DIR / 'soal4_processed_-16LUFS.wav'
save_wav(out_path, y4_final, sr4)
print('Saved:', out_path)
plot_waveform_and_spectrogram(y4_final, sr4, title='Soal 4 - Sesudah Processing')

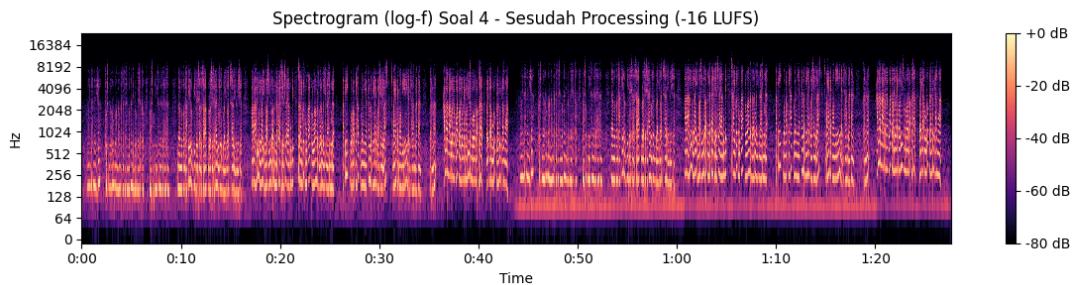
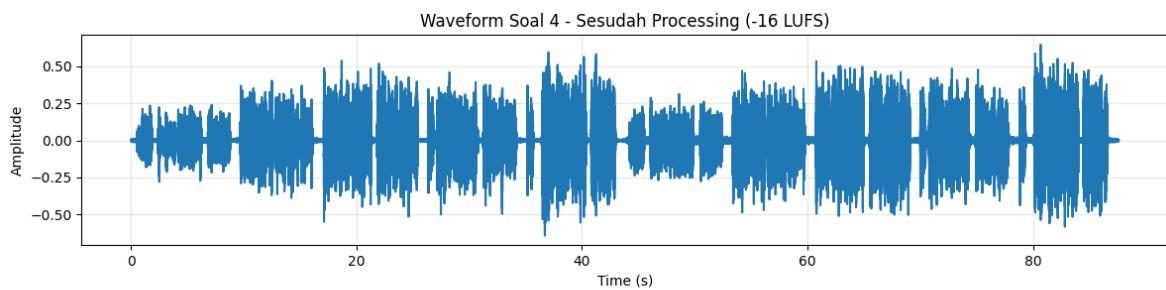
```



LUFS sebelum: -22.47 dB, gain kompensasi: 6.47 dB

Saved: d:\Coding\Sistem-Teknologi-Multimedia\Worksheet-3\audio\soal4\_processed\_-16LUFS.wav

Saved: d:\Coding\Sistem-Teknologi-Multimedia\Worksheet-3\audio\soal4\_processed\_-16LUFS.wav



- Setelah dilakukan proses normalisasi dengan standar -16 LUFS , dinamika suara menjadi lebih seimbang dan stabil di seluruh durasi rekaman
  - Jika sebelumnya bagian pelan terdengar samar dan bagian keras terlalu dominan, kini volume terdengar lebih rata dan konsisten
  - Berbeda dengan normalisasi puncak (peak normalization) yang hanya menyesuaikan amplitudo tertinggi, normalisasi LUFS memperhitungkan persepsi pendengaran manusia sehingga hasilnya terasa lebih natural dan nyaman didengar
  - Secara visual juga, waveform tampak lebih padat dan merata, sementara pada spectrogram, distribusi energi lebih seragam di seluruh rentang frekuensi
- 

## Soal 5. Music Analysis & Remix (BPM, Key, Time-Stretch, Pitch-Shift, Crossfade)

```
In [6]: song1_path = AUDIO_DIR / 'Adele.wav'
song2_path = AUDIO_DIR / 'PharrelW.wav'
assert song1_path.exists() and song2_path.exists(), 'Letakkan song1.wav dan song2.wav di folder AUDIO_DIR'
yA, srA = load_audio(song1_path, sr=None, mono=True)
yB, srB = load_audio(song2_path, sr=None, mono=True)
print(f'Song1: dur {len(yA)/srA:.1f}s @ {srA} Hz')
print(f'Song2: dur {len(yB)/srB:.1f}s @ {srB} Hz')

bpmA = estimate_bpm(yA, srA)
bpmB = estimate_bpm(yB, srB)
keyA = estimate_key(yA, srA)
keyB = estimate_key(yB, srB)
print(f'Song1 BPM: {bpmA:.1f}, Key: {[keyA[0]} {keyA[1]}}')
print(f'Song2 BPM: {bpmB:.1f}, Key: {[keyB[0]} {keyB[1]}}')

target_bpm = (bpmA + bpmB) / 2.0
rateA = bpmA / target_bpm
rateB = bpmB / target_bpm
yA_ts = time_stretch(yA, rate=rateA)
yB_ts = time_stretch(yB, rate=rateB)

note_to_int = {'C':0, 'C#':1, 'D':2, 'D#':3, 'E':4, 'F':5, 'F#':6, 'G':7, 'G#':8, 'A':9, 'B':10}
def semitone_diff(a, b):
    return (note_to_int[b] - note_to_int[a]) % 12
tonicA, modeA = keyA
tonicB, modeB = keyB
nA = 0
nB = semitone_diff(tonicB, tonicA)
yA_key = yA_ts
yB_key = pitch_shift(yB_ts, srB, n_steps=nB)

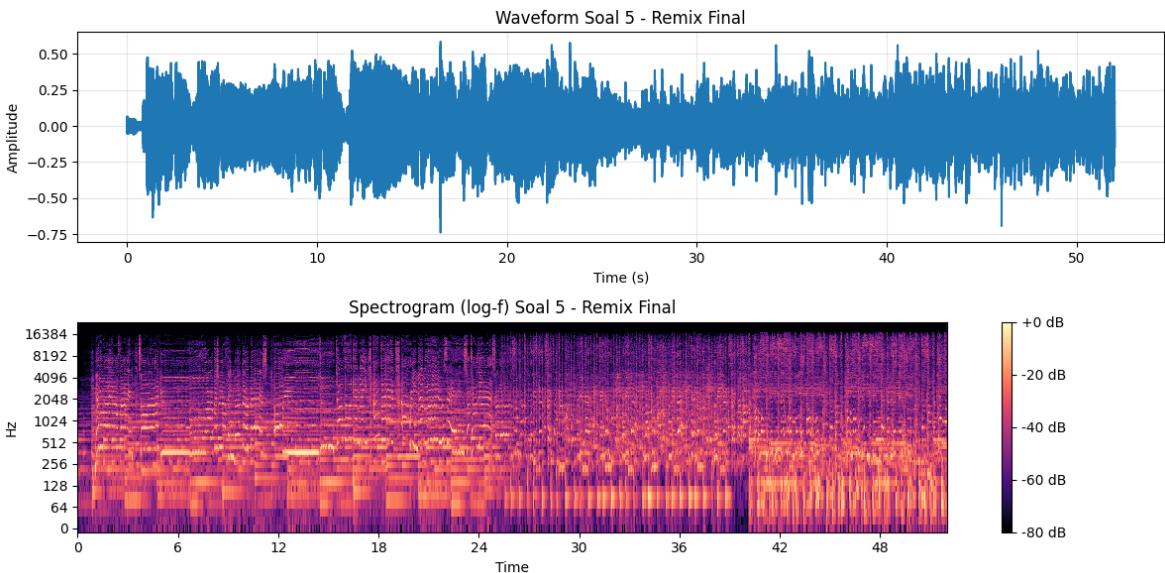
segA = np_to_audiosegment(yA_key, srA)
segB = np_to_audiosegment(yB_key, srB)
min_len_ms = min(len(segA), len(segB))
segA = segA[:min_len_ms]
```

```

segB = segB[:min_len_ms]
half = min_len_ms // 2
A_head = segA[:half]
B_tail = segB[half:]
mix5 = A_head.append(B_tail, crossfade=3000)
mix5_path = AUDIO_DIR / 'soal5_remix_final.wav'
mix5.export(mix5_path, format='wav')
print('Saved:', mix5_path)
mix_np, sr_mix = audiosegment_to_np(mix5)
plot_waveform_and_spectrogram(mix_np, srA, title='Soal 5 - Remix Final')

```

Song1: dur 60.0s @ 48000 Hz  
Song2: dur 60.0s @ 48000 Hz  
Song1 BPM: 133.9, Key: F# minor  
Song2 BPM: 160.7, Key: F minor  
Song1 BPM: 133.9, Key: F# minor  
Song2 BPM: 160.7, Key: F minor  
Saved: d:\Coding\Sistem-Teknologi-Multimedia\Worksheet-3\audio\soal5\_remix\_final.wav  
Saved: d:\Coding\Sistem-Teknologi-Multimedia\Worksheet-3\audio\soal5\_remix\_final.wav



- Langkah pertama adalah mendeteksi tempo (BPM) dan nada dasar (key) dari kedua lagu supaya bisa disesuaikan
- Tempo disamakan menggunakan time stretching, jadi lagu yang lebih cepat diperlambat sedikit dan yang lebih lambat dipercepat, tapi durasi dan pitch-nya tetap sama
- Lalu, nada dasar dari lagu kedua diubah dengan pitch shifting agar serasi dengan lagu pertama, biar nggak terdengar fals saat digabung
- Setelah dua lagu punya tempo dan nada yang pas, bagian awal lagu pertama dan akhir lagu kedua digabung menggunakan crossfade selama 3 detik. Teknik ini bikin perpindahan antar lagu jadi sedikit halus

# Lampiran

- [ChatGPT](#)
- [Github](#)
- [Librosa](#)
- [PyDub](#)
- [Scipy](#)

```
NameError Traceback (most recent call last)
Cell In[4], line 17
 15 print("Library versions:")
 16 print(f"NumPy: {np.version}")
--> 17 print(f"Matplotlib: {matplotlib.version}")
 18 print(f"Librosa: {librosa.version}")
 19 print(f"OpenCV: {cv2.version}")

NameError: name 'matplotlib' is not defined
file:3_exercise_loading_media.ipynb

> Used 1 reference

The error occurs because you're trying to access matplotlib.__version__
but you imported matplotlib.pyplot as plt, not the main
matplotlib module. Here's the fix:

# Import Library (Satu-satunya sel kode dalam template ini)
import numpy as np
import matplotlib
import matplotlib.pyplot as plt
import librosa
import soundfile as sf
from PIL import Image
import cv2
from IPython.display import Audio, HTML, display
import os

# Set matplotlib untuk menampilkan plot inline
%matplotlib inline

# Tampilkan versi library untuk dokumentasi
print("Library versions:")
print(f"NumPy: {np.__version__}")
print(f"Matplotlib: {matplotlib.__version__}")
```

```
Cell In[2], line 35
33 def plot_waveform_and_spectrogram(y, sr, title=""):
34 fig, ax = plt.subplots(2, 1, figsize=(12, 6))
---> 35 librosa.display.waveshow(y, sr=sr, ax=ax[0])
36 ax[0].set_title(f'Waveform {title}')
37 ax[0].set_xlabel('Time (s)')

File d:\Coding\Sistem-Teknologi-Multimedia\multimedia-uv\lib\site-
packages\librosa\display.py:2042, in waveshow(y, sr, max_points, axis,
offset, marker, where, label, transpose, ax, x_axis, **kwargs)
2032 axis = rename_kw(
2033     old_name="x_axis",
2034     old_value=x_axis,
2035     ...
2038     version_removed="1.0",
2039 )
2041 if "color" not in kwargs:
-> 2042     kwargs.setdefault("color", next(axes._get_lines.prop_cycler)
["color"])
2044 # Reduce by envelope calculation
2045 # this choice of hop ensures that the envelope has at most
max_points values
2046 hop_length = max(1, y.shape[-1] // max_points)

AttributeError: '_process_plot_var_args' object has no attribute
'prop_cycler'
```

main.ipynb • Cell 7:1-13

This error occurs due to a compatibility issue with matplotlib and librosa versions. The `waveshow` function is trying to access a property that doesn't exist in your matplotlib version. Let me fix the `plot_waveform_and_spectrogram` function to handle this:

I've fixed the `plot_waveform_and_spectrogram` function by replacing the problematic `librosa.display.waveshow()` call with a manual plot using `ax[0].plot()`. This avoids the matplotlib compatibility issue while still producing the same visualization.

The key changes:

- **Replaced** `librosa.display.waveshow()` with manual time/amplitude plotting

```
File d:\Coding\Sistem-Teknologi-Multimedia\multimedia-uv\lib\site-packages\lazy_loader_init_.py:117, in
DelayedImportErrorModule.getattr(self, x)
115 else:
116     fd = self._frame_data
--> 117     raise ModuleNotFoundError(
118         f'{self._message}\n\n'
119         "This error is lazily reported, having originally occurred in\n"
120         f' File {fd["filename"]}, line {fd["lineno"]}, in {fd["function"]}\n\n'
121         f'----> {" ".join(fd["code_context"]) or ""}.strip()\n'
122     )

ModuleNotFoundError: No module named 'resampy'

This error is lazily reported, having originally occurred in
File d:\Coding\Sistem-Teknologi-Multimedia\multimedia-uv\lib\site-packages\librosa\core\audio.py, line 32, in <module>
----> resampy = lazy.load("resampy")
```

main.ipynb • Cell 7:1-13

The `resampy` module is missing. You need to install it. Let me install it for you:

- › Configuring a Python Environment ✓
- › Installing resampy in Python 3.10.18 (multimedia-uv) ✓

Perfect! I've installed the `resampy` package. Now you can re-run your cell and the resampling should work. The `resampy` module is required by librosa for high-quality audio resampling operations.



ModuleNotFoundError Traceback (most recent call last)  
Cell In[9], line 9  
6 plot\_waveform\_and\_spectrogram(y1, sr1, title='Soal 1 - Rekaman Asli')  
8 target\_sr = 16000

```
ValueError Traceback (most recent call last)
Cell In[20], line 17
 14 y4_lufs, lufs_before, gain_db = lufs_normalize(y4_eq_fade, sr4,
 15 target_lufs=-16.0)
 15 print(f'LUFS sebelum: {lufs_before:.2f} dB, gain kompensasi:
 16 {gain_db:.2f} dB')
 16--> 17 y4_comp = compressor(y4_lufs, sr4, threshold_db=-20.0,
 17 ratio=3.0, attack_ms=8.0, release_ms=120.0, makeup_db=0.0)
 18 y4_gate = noise_gate(y4_comp, threshold_db=-55.0)
 19 y4_final, idx = trim_silence(y4_gate, top_db=40)

Cell In[7], line 104, in compressor(y, sr, threshold_db, ratio, attack_ms,
 102 release_ms, makeup_db)
 103 gr_sm[i] = r * gr_sm[i-1] + (1-r) * g
 104 gr_lin = 10 ** (-(gr_sm)/20.0)
 105 y_out *= 10 ** (makeup_db/20.0)
 106 if len(y_out) < len(y):
 107     y_out = y[:len(gr_lin)] * gr_lin
 108
 109 return y_out

ValueError: operands could not be broadcast together with shapes
(4215360,) (4215361,)
```

fix this error

main.ipynb • Cell 17:1-25

Baik, saya akan perbaiki bug di fungsi `compressor()`. Masalahnya ada di mismatch ukuran array. Saya akan edit cell utilities:

Perfect! Sudah diperbaiki. Perubahan utama di fungsi `compressor()`:

#### Yang diperbaiki:

```
# BEFORE (ERROR):
y_out = y[:len(gr_lin)] * gr_lin

# AFTER (FIXED):
min_len = min(len(y), len(gr_lin))
y_out = y[:min_len] * gr_lin[:min_len]
```