

Electrical and Computer Engineering Spoken Language Processing - Spring 2024

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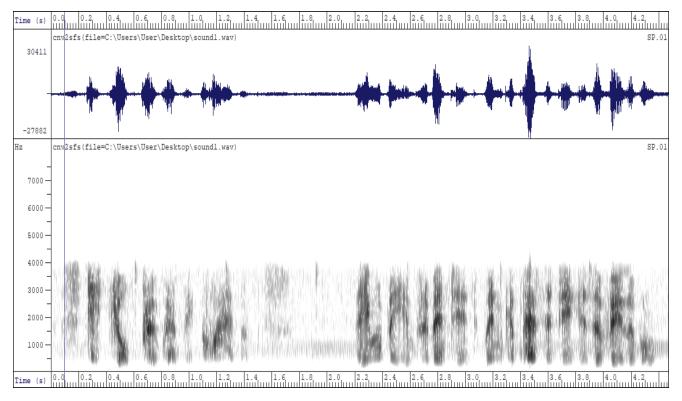
Section: 2

Date: 29/4/2024

Part 1 – Basic sound analysis, Spectrograms

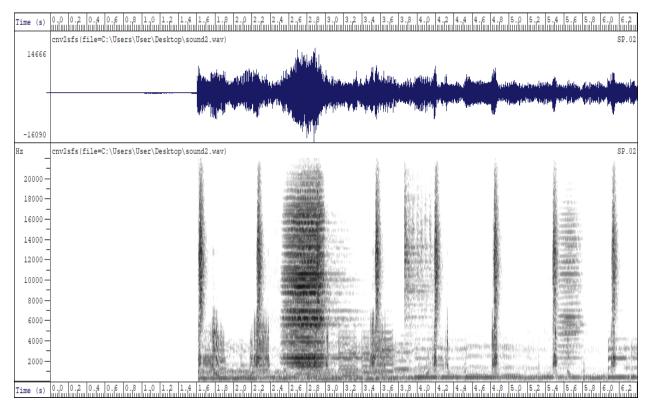
a) Bandwidth

sound1.wav:



this is the waveform and wide band spectrogram of sound1.wav, we find from the spectrogram that the bandwidth equal 4000 Hz, and the lower frequency equal 500 Hz, and upper frequency equal 4000 Hz.

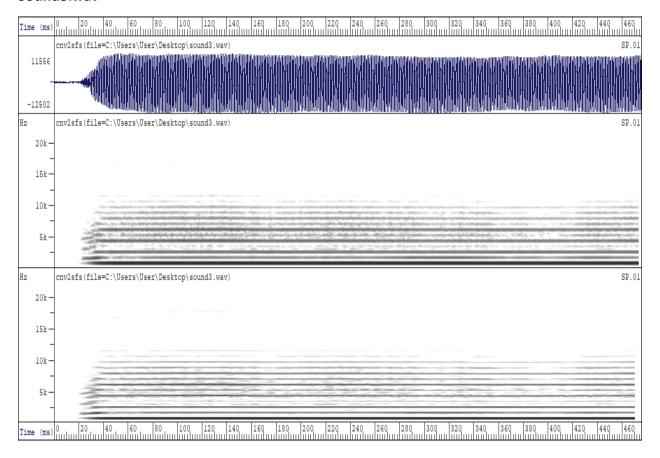
Sound2.wav:



this is the waveform and wide band spectrogram of sound2.wav, we find from the spectrogram that the bandwidth equal 22000 Hz, and the lower frequency equal 100 Hz, and upper frequency equal 22000 Hz.

b) Spectrogram

Sound3.wav

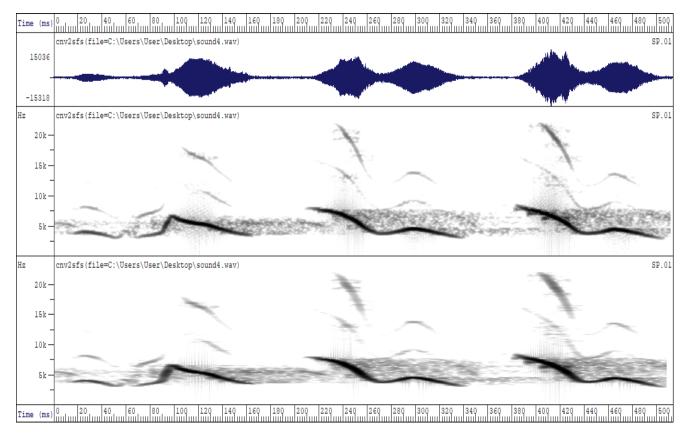


This is the waveform and wide band and narrow band spectrogram of sound3.wav.

We notice that the narrowband spectrogram exhibits the harmonic structure in the form horizontal striations and in this case the frequency resolution will be the best.

The wideband spectrogram exhibits periodic temporal structure in the form of vertical striations and in this case the frequency resolution will be the worst, and it is good for temporal resolution.

Sound4.wav

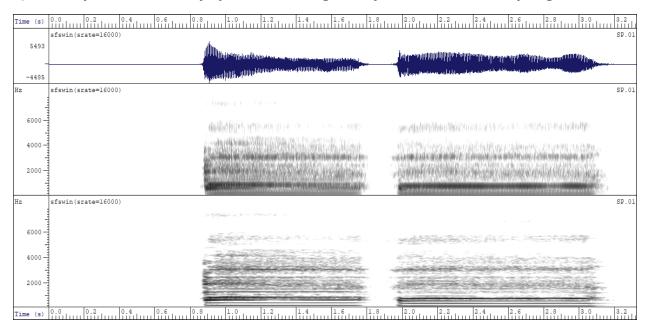


This is the waveform and wide band and narrow band spectrogram of sound4.wav.

We notice that the narrowband spectrogram exhibits the harmonic structure in the form horizontal striations and in this case the frequency resolution will be the best.

The wideband spectrogram exhibits periodic temporal structure in the form of vertical striations and in this case the frequency resolution will be the worst, and it is good for temporal resolution.

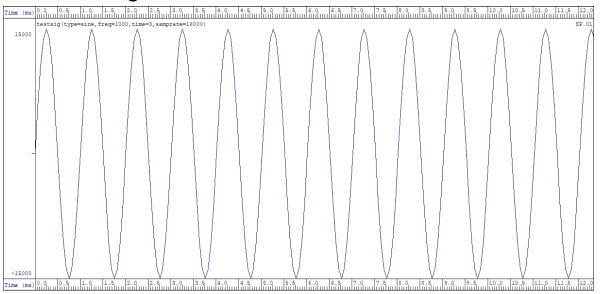
c) In this part I record my speech uttering the syllable 'afa' at sampling rate 16KH



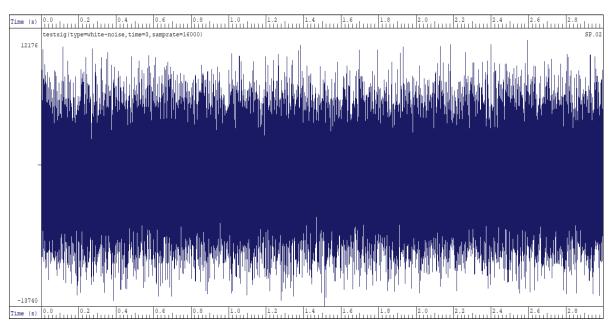
This is the waveform and wide band and narrow band spectrogram of 'afa' we notice that 'afa' contain 2 voiced (periodic) part (vowel 'a') and one unvoiced part 'f'. we notice from spectrogram that in voiced parts the power positioned in low frequencies , and in unvoiced parts the power positioned in high frequencies.

Part 2 – Filters

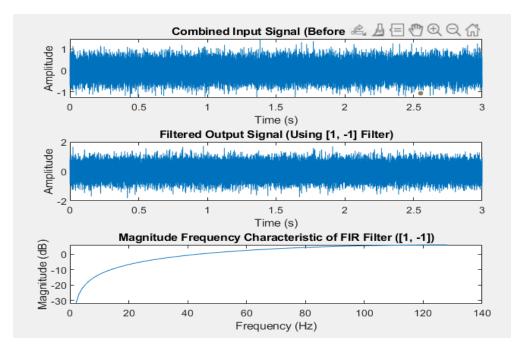
Sinousoidal signal:



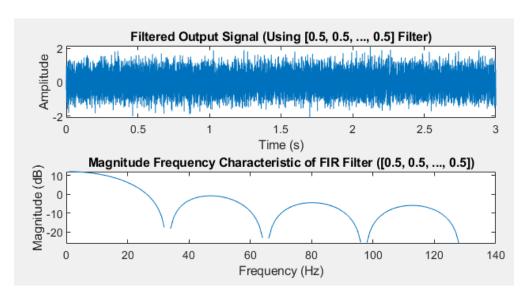
White noise signal:



Combined signal and result of first filter:



Result of second filter:



FIR Filter with Impulse Response {1, -1}:

Sine-wave Strength: This filter is designed to act as a simple differentiator. It emphasizes high-frequency components in the input signal. Therefore, the sine-wave component, being a 1 kHz signal, might be preserved in the output with minimal attenuation.

Noise Attenuation: As white noise contains a broad spectrum of frequencies, this filter may attenuate high-frequency noise components more effectively than low-frequency ones. Thus, the noise may be somewhat reduced in the output.

FIR Filter with Difference Equation Coefficients:

Sine-wave Strength: The coefficients in the difference equation suggest that this filter is a type of low-pass filter. It emphasizes lower frequencies and attenuates higher frequencies. Therefore, the output may have a weaker representation of the 1 kHz sine-wave compared to the input.

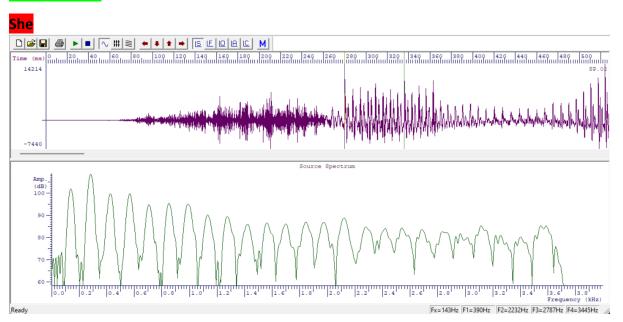
Noise Attenuation: As a low-pass filter, it may attenuate higher-frequency noise components more effectively, resulting in a reduction of noise in the output signal. However, it may also introduce some phase distortion due to the recursive nature of the filter.

Part 3 – Speech Analysis

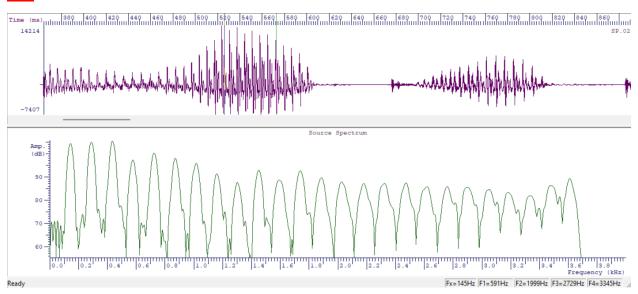
a) Formants for vowels

In this part I estimate the formant frequencies F1,F2,F3, of four vowels 'e' in she, 'a' in had, 'u' in suit 'a' in dark from Sample1.wav, my recording, and my partner recording.

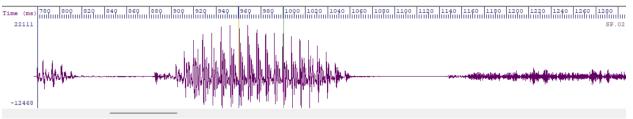
Sample1.wav:

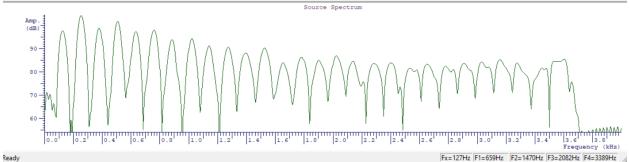




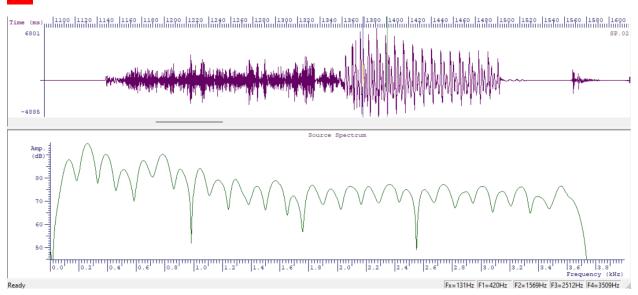








Suit



Sample1.wav formants table:

Formants	F1 (HZ)	F2 (HZ)	F3 (HZ)
sh <mark>e</mark>	390	2232	2787
h <mark>a</mark> d	591	1999	2729
d <mark>a</mark> rk	659	1470	2082
s <mark>u</mark> it	420	1569	2512

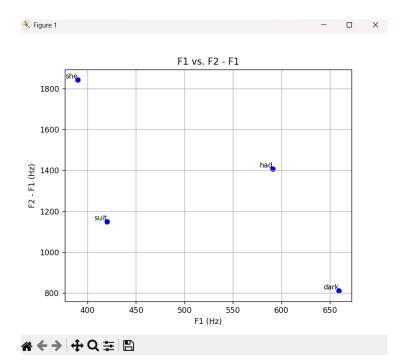
Python code to plot F1 versus F2-F1:

```
import matplotlib.pyplot as plt

# Formant data
words = ['she', 'had', 'dark', 'suit']
f1_values = [390, 591, 659, 420]
f2_values = [2232, 1999, 1470, 1569]

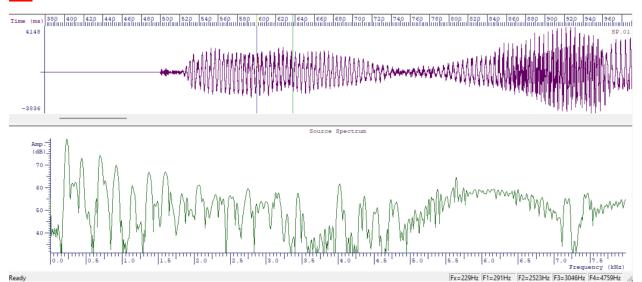
# Calculate F2 - F1
f2_minus_f1 = [f2 - f1 for f1, f2 in zip(f1_values, f2_values)]

# Plot
plt.figure(figsize=(8, 6))
plt.scatter(f1_values, f2_minus_f1, color='blue')
plt.title('F1 vs. F2 - F1')
plt.xlabel('F1 (Hz)')
plt.ylabel('F2 - F1 (Hz)')
for i, word in enumerate(words):
    plt.text(f1_values[i], f2_minus_f1[i], word, fontsize=9, ha='right', va='bottom')
plt.grid(True)
plt.show()
```

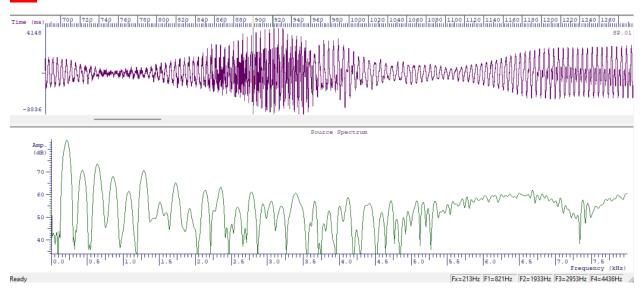


Redording1.wav

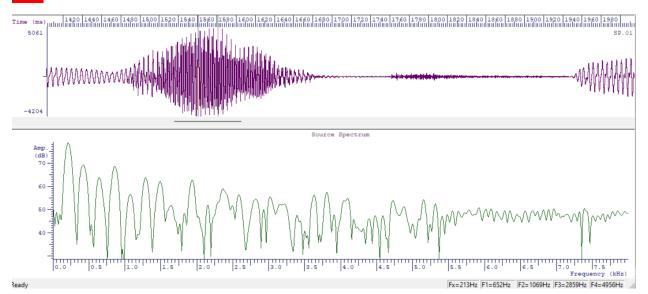




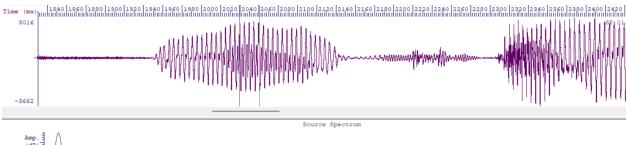
Had

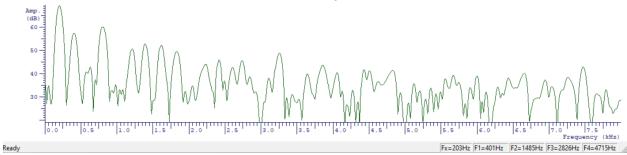






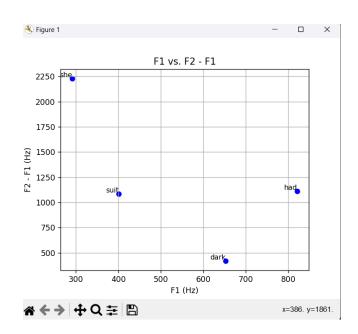
Suit





Recording 1. wav formants table:

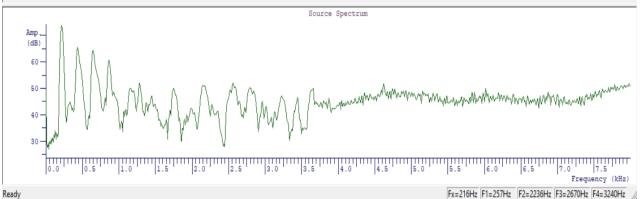
Formants	F1 (HZ)	F2 (HZ)	F3 (HZ)
sh <mark>e</mark>	291	2523	3046
h <mark>a</mark> d	821	1933	2953
d <mark>a</mark> rk	652	1069	2859
s <mark>u</mark> it	401	1485	2826



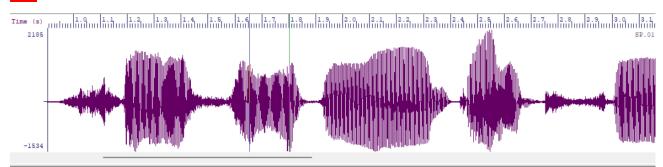
Recording2.wav

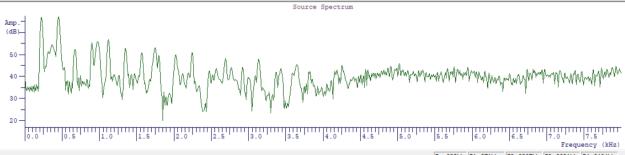
She





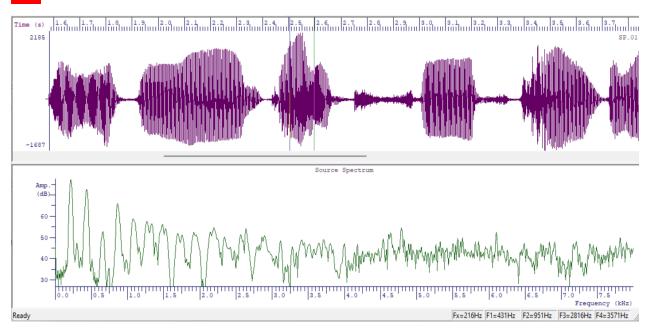
Had



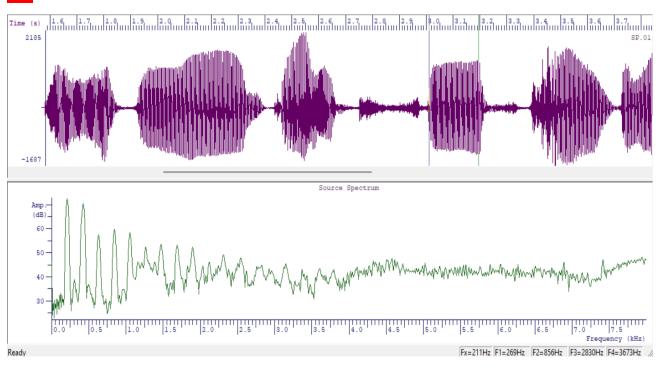


Fx=225Hz F1=271Hz F2=2207Hz F3=2821Hz F4=3184Hz

dark

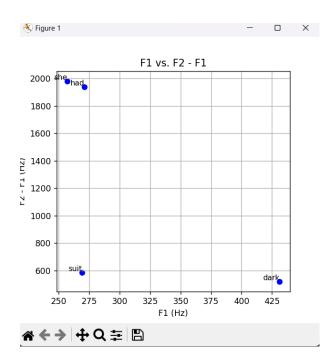


Suit



Recording2.wav formants table:

Formants	F1 (HZ)	F2 (HZ)	F3 (HZ)
sh <mark>e</mark>	257	2236	2670
h <mark>a</mark> d	271	2207	2821
d <mark>a</mark> rk	431	951	2816
s <mark>u</mark> it	269	856	2830



From this part after measuring formants frequencies for vowels in 3 files different speakers, we find that the formants frequency vary for different vowels and for the same vowel across different speakers.

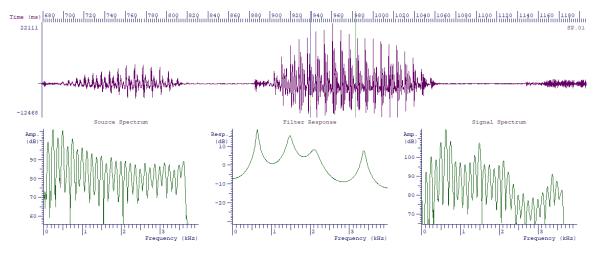
For voiced speech the magnitude of the lower formants frequencies is larger than magnitude of higher formants frequencies .

For unvoiced speech the magnitude of the higher formants frequencies is larger than magnitude of lower formants frequencies .

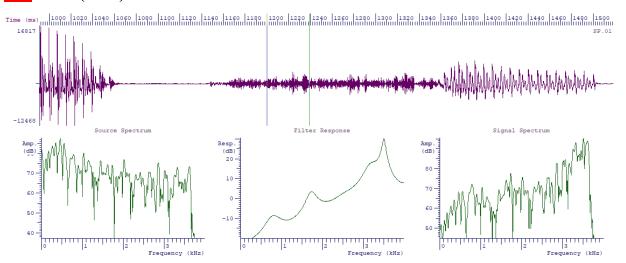
b) Source and Filter analysis

In this part I perform the source filter analysis for phoneme $\slash\!\!\!/ s/$ in 'suit' and $\slash\!\!\!/ A/$ in 'dark' .

Dark 'a' (vowel periodic)



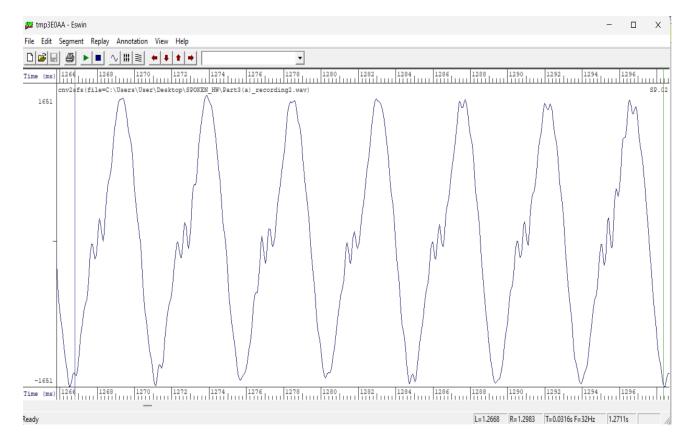
Suit 's' (noise)



There is difference between filter response for each phoneme.

c) Fundamental Frequency

RECORDING1



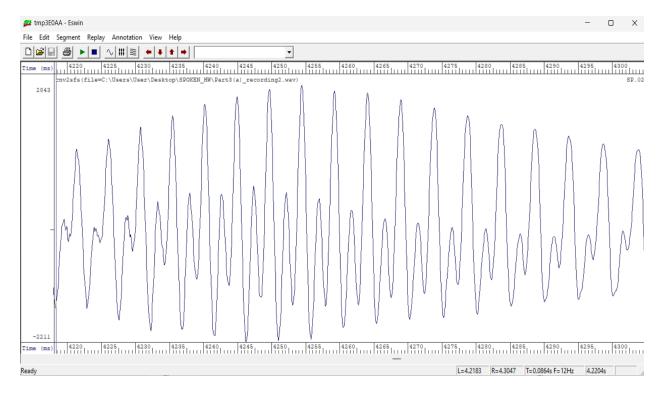
Fundamental period= duration/ number of periods

= 0.0316/7=4.514 * 10^-3

Fundamental frequency=1/ fundamental period= 1/(4.514 * 10^-3)

= 221.5 HZ

RECORDING2



Fundamental period= duration/ number of periods

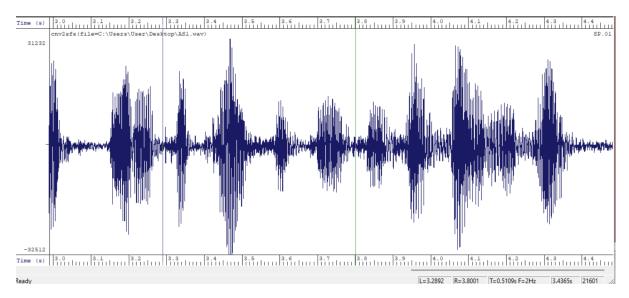
= 0.0864/18=4.8 * 10^-3

Fundamental frequency=1/ fundamental period= 1/(4.8 * 10^-3)

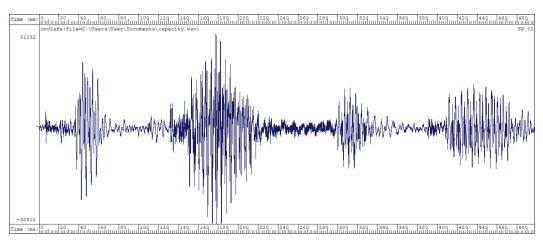
= 208.33 HZ

Part 4 – Speech analysis

All waveform in AS1.wav



'Capacity' waveform:

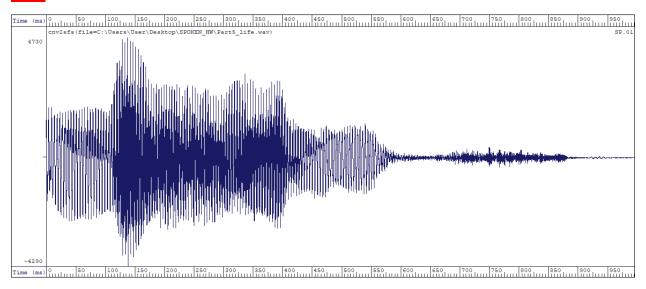


Phoneme	Start time (ms)	End time (ms)
/k/ C	0	17
/@/ a	18	62
/p/ p	63	130
/{/ a	131	222
/s/ c	223	298
/I/ i	299	232
/t/ t	233	292
/i/ y	293	508

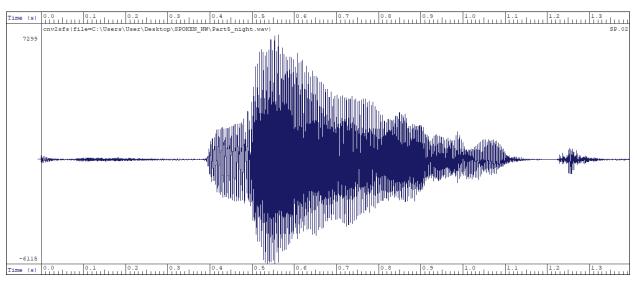
Part 5 – Sub-word level concatenation

In this part I record 'life' and night' words to make a new word 'light' by make concatenation between 'li' from 'life' and 'ght' from 'night' and I used python code to make concatenation between the two sub word level .wav files.

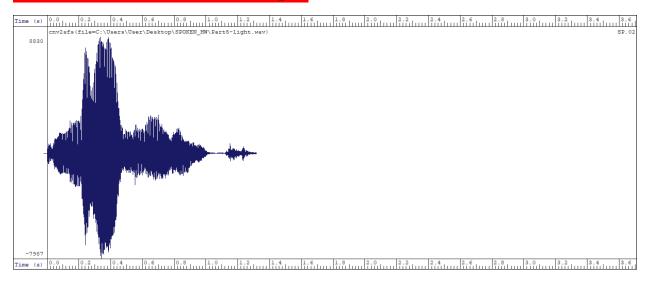
Life waveform>



Night waveform>



Result of concatenation: light



Python code for concatenation:

```
ifrom pydub import AudioSegment
ifrom tkinter import Tk, filedialog

lusage

idef get_file_paths():
    Tk().withdraw()_# Hide the Tkinter root window
    print("Please select the first .wav file:")
    first_file_path = filedialog.askopenfilename(filetypes=[("WAV files", "*.wav")])
    print("Please select the second .wav file:")
    second_file_path = filedialog.askopenfilename(filetypes=[("WAV files", "*.wav")])

return first_file_path, second_file_path
# Function to concatenate two audio files

lusage

ilusage

ilusage
```

We produce a high quality speech from using two sub word level concatenation process.

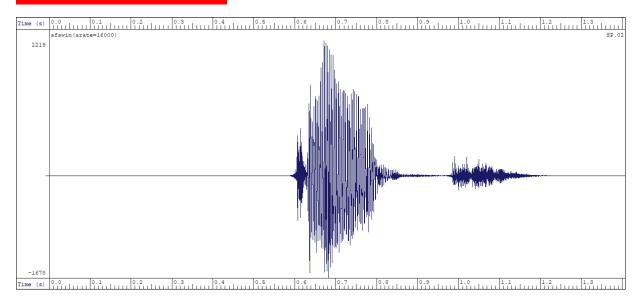
Part 6 – Phone-level concatenation

In this part I make a different word from capacity word phonemes {c,a,p,a,c,i,t,y} And the new word is 'cat' by making concatenation between 'c' 'a' 't' parts.

Python code to make concatenation between phonemes:

```
from pydub import AudioSegment
from tkinter import Tk, filedialog
def get_file_paths():
    Tk().withdraw() # Hide the Tkinter root window
    first_file_path = filedialog.askopenfilename(filetypes=[("WAV files", "*.wav")])
    second_file_path = filedialog.askopenfilename(filetypes=[("WAV files", "*.wav")])
    third_file_path = filedialog.askopenfilename(filetypes=[("WAV files", "*.wav")])
    return first_file_path, second_file_path, third_file_path
def concatenate_audio(first_file_path, second_file_path):
    first_audio = AudioSegment.from_wav(first_file_path)
    second_audio = AudioSegment.from_wav(second_file_path)
    concatenated_audio = first_audio + second_audio
    return concatenated_audio
def save_audio(output_audio, output_file_path):
    output_audio.export(output_file_path, format="wav")
    print(f"Concatenated audio saved as '{output_file_path}'.")
if __name__ == "__main__":
    first_file_path, second_file_path, third_file_path = get_file_paths()
    output_file_path = filedialog.asksaveasfilename(defaultextension=".wav", filetypes=[("WAV files", "*.wav")])
    concatenated_audio = concatenate_audio(first_file_path, second_file_path, third_file_path)
    save_audio(concatenated_audio, output_file_path)
```

C+a+t: cat waveform



We produce a low quality speech from using phone level concatenation process. We can improve quality by select high quality speech units, and implement algorithms that select a most appropriate speech units for concatenations.

Part 7 – Generation of vowel sounds using the source-filter model (parallel formant synthesiser)

In this exercise, the source-filter model was employed to generate three vowel sounds (/A:/, /u:/, and /ə/) using a pulse-train source signal passed through vocal-tract filters.

Generation Procedure:

A periodic pulse-train source signal at 110 Hz frequency was generated to mimic the vibrating vocal cords during voiced sounds.

Two band-pass filters were applied in parallel to the source signal, each corresponding to one of the first two formant frequencies of the respective vowel.

The first two formant frequencies used were: A:/(F1=700Hz, F2=1100Hz), u:/(F1=300Hz, F2=800Hz), and a/a/(F1=500Hz, F2=1500Hz).

The filtered signals were exported as separate way files.

Signal Combination:

The filtered signals were then loaded into Matlab using the audioread function.

Each signal was summed element-wise to obtain the final speech signal representing the vowel sound.

The final signal was stored in the way file format using the audiowrite function.

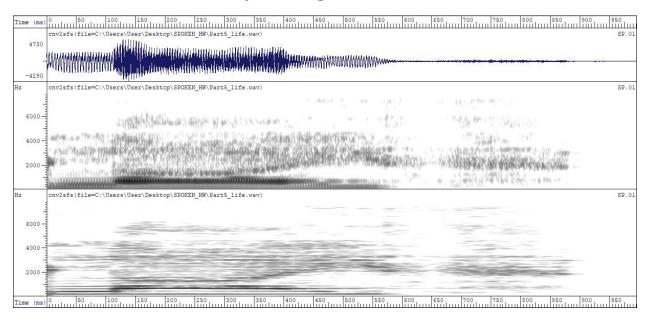
Output:

The generated way files for the three vowels were submitted as the deliverable.

In conclusion, This exercise demonstrates the concept of vowel generation using the source-filter model, where the source signal (vibrating vocal cords) is filtered through the vocal tract (formant frequencies) to produce distinct vowel sounds. By adjusting the formant frequencies, different vowels can be synthesized, showcasing the importance of formant frequencies in speech production and perception. Additionally, the option to use white noise as the source signal provides insight into the generation of whispered vowels, where vocal fold vibration is absent. Overall, this exercise enhances understanding of the acoustic characteristics of vowels and their synthesis using the source-filter model.

Part 8 – Formant synthesis in SFS

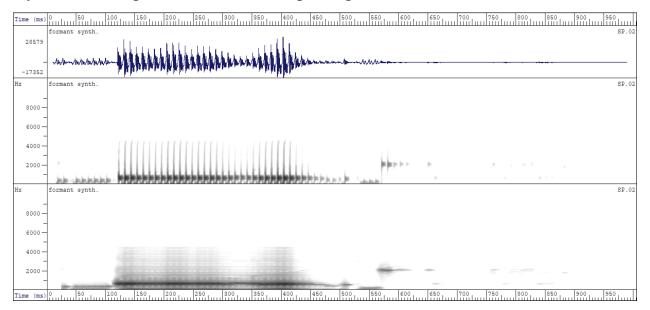
'life' waveform and spectrogram:



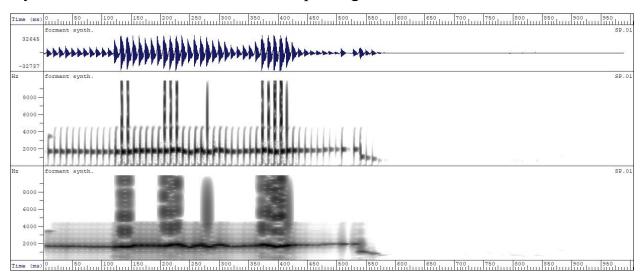
Formants.txt of original signal 'life':

oosn size v	gain;	F1	B1	A1;	F2	B2	A2;	F3	В3	A3;	F4	B4	A4;	F5	B5	A5;
10.0 20.0 1	85.0;	294	62	82:	2224	24	67:	4239	27	49:	5645	321	24:	7220	242	25;
20.0 20.0 1	85.8;	293			2266		,	3171			4321			6766		23;
30.0 20.0 1	86.8;	292			685			2232			3131			4286		51;
10.0 20.0 1	86.9;	393			2462			3295			4209			5282		25;
50.0 20.0 1	86.7;	391			2530			3146			4230			5304		21;
50.0 20.0 1	86.9;	386			3084			4498			6129			7349		18;
70.0 20.0 1	86.9;	384			2803			3280			4390			7298		23;
30.0 20.0 1	86.9;	383			2453			3237			4377			7168		21;
90.0 20.0 1	86.4;	418			2477		-	3214			4326			7074		21;
00.0 20.0 1	86.2;	378	78	82;	761			2508		-	3161		_	4315		34;
10.0 20.0 1	86.4;	481	195	79;	677	190	75;	1929	202	50;	3171	77	55;	4190	255	40;
20.0 20.0 1	89.4;	692	28	91;	3132	92	51;	4220	115	47;	5513	137	35;	7295	229	23;
30.0 20.0 1	91.6;	687	19	95;	979	331	70;	3238	214	47;	4243	133	47;	5520	154	36;
10.0 20.0 1	92.2;	680	22	94;	972	176	74;	3069	235	46;	4064	55	53;	5699	265	38;
50.0 20.0 1	91.4;	691	28	92;	4101	72	51;	5337	111	41;	5833	58	42;	7260	89	24;
50.0 20.0 1	90.8;	693	27	92;	3368	164	50;	4229	111	44;	5539	162	37;	5969	181	34;
70.0 20.0 1	89.8;	699	22	91;	1363	143	65;	2821	188	48;	3838	182	48;	6085	145	36;
30.0 20.0 1	88.4;	694	13	92;	1341	76	66;	2640	185	49;	4147	170	41;	5529	185	39;
90.0 20.0 1	88.0;	690	12	92;	1348	94	64;	3270	80	50;	5501	132	37;	6043	161	33;
00.0 20.0 1	88.1;	686	8	94;	1425	105	61;	2588	247	44;	3370	74	50;	5529	79	40;
10.0 20.0 1	88.3;	684	7	94;	1392	79	63;	2705	170	48;	3384	138	49;	5449	65	41;
20.0 20.0 1	88.0;	689	8	94;	1448	205	56;	2565	182	48;	3338	147	46;	5515	51	43;
30.0 20.0 1	87.6;	681	11	91;	1481	204	55;	2626	259	45;	3384	116	49;	5501	378	32;
10.0 20.0 1	87.2;	686	15	90;	1380	91	63;	2758	172	49;	4099	228	38;	5513	105	39;
50.0 20.0 1	87.3;	688	7	94;	1457	170	55;	2677			3339	176	45;	5510	149	38;
50.0 20.0 1	87.4;	679	11	91;	1379	81	63;	2723	197	47;	3361	137	49;	5410	267	35;
70.0 20.0 1	87.3;	682	8	93;	1374	64	64;	2824	175	45;	3720	176	44;	5694	128	35;
30.0 20.0 1	87.3;	676	13	90;	1381	80	63;	2709	170	47;	3620	153	49;	5467	213	37;
90.0 20.0 1	86.7;	673	14	90;	1360	72	64;	3349			3939	172	43;	5710	208	32;
00.0 20.0 1	86.7;	678	22	88;	1357	81	66;	3217	141	50;	4352	198	37;	5509	86	40;
10.0 20.0 1	86.8;	713	42	85;	1335	172	65;	3026	89	56;	3854	245	41;	5344	234	32;
20.0 20.0 1	87.1;	730	76	84;	945	280	73;	2205	348	49;	3049	108	53;	4018	195	40;
30.0 20.0 1	87.2;	736			1730			3025			5604			7368		19;
10.0 20.0 1	87.2;	700			3114			3888			5646			7317		22;
50.0 20.0 1	87.3;	693			1567			3145			4269			5614		33;
50.0 20.0 1 70.0 20.0 1	87.2; 87.4;	673 667			1589 1762			4141 3101			5578 4290			7290 5570		20; 39;

'synthesized original' waveform and spectrogram:



'synthesized modified' waveform and spectrogram:



To change the speaker's voice in a formant-based speech synthesizer, we use speech analysis tools to extract formant data from the reference recordings. Formants represent the resonant frequencies of the vocal tract and play a crucial role in determining the characteristics of a speaker's voice.