CCO50- Digital Speech Processing

Short Test 2

Description: Define two discrete-time signals, i.e., a[·] and b[·], being the former two-sample long and the latter four-sample long. Then, obtain the resulting signal y[·]=a[·] * b[·].

$$a = \begin{pmatrix} 10 & 0 \end{pmatrix}$$
$$b = \begin{pmatrix} -6 & -1 & 0.41 & 2 \end{pmatrix}$$
$$y = a * b$$
$$a * b = \sum_k a_k \cdot b_{n-k}$$

We can use a practical procedure to solve this by hand:

			10	0	a
	-6	-1	0.41	2	b
			20	0	
		4.1	0	+	
	-10	0	+		
-60	0	+			
(-60	-10	4.1	20	0)	$= a * b$

$$y = \begin{pmatrix} -60 & -10 & 4.1 & 20 & 0 \end{pmatrix}$$

Description: Convolve y[·], just obtained, with itself.

				-60	-10	4.1	20	0	y
				-60	-10	4.1	20	0	y
				0	0	0	0	0	
			-1200	-200	82	400	0	+	
		-246	-41	16,81	82	0	+		
	600	100	-41	-200	0	+			
3600	600	-246	-1200	0	+				
(3600	1200	-392	-2482	-383.19	164	400	0	0)	$= y * y$

$$y * y = \begin{pmatrix} 3600 & 1200 & -392 & -2482 & -383.19 & 164 & 400 & 0 & 0 \end{pmatrix}$$

Description: find a general equation to calculate the length of any resulting convolved signal from the lengths of both the input signals.

$$length(a) = 2$$
$$length(b) = 4$$
$$length(y) = 5 = length(a) + length(b) - 1$$
$$length(y * y) = 9 = length(y) + length(y) - 1$$
$$y = h[\cdot] * x[\cdot]$$
$$length(y) = length(h) + length(x) - 1$$

Plus: implementing the convolution in python

In [2]:
```
def discrete_time_convolution(k, x):
    n = len(k)
    m = len(x)
    y = [0] * (m + n - 1)

    for i in range(m):
        for j in range(n):
```

```
            y[i + j] += x[i] * k[j]

    return y
```

In [8]:
```
a = [10, 0]
b = [-6, -1, 0.41, 2]

y = discrete_time_convolution(a, b)

print(y)
```

```
[-60, -10, 4.1, 20.0, 0]
```

In [9]:
```
y_2 = discrete_time_convolution(y, y)
print(y_2)
```

```
[3600, 1200, -391.99999999999994, -2482.0, -383.19, 164.0, 400.0, 0.0, 0]
```

Extra

Generating waves and covolving them

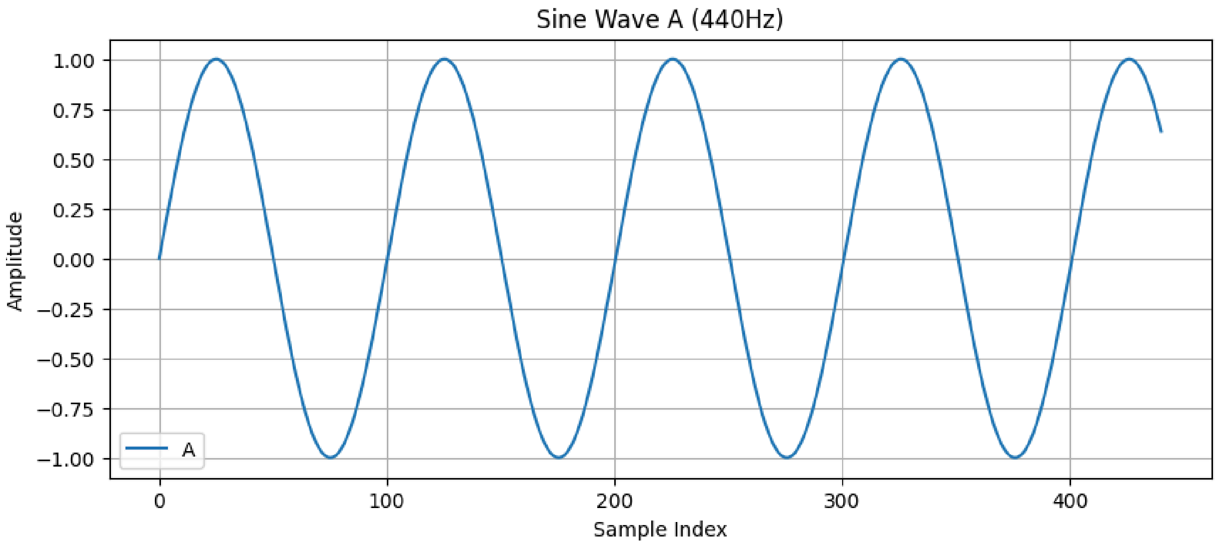
In [14]:
```
import numpy as np
import pyaudio
import matplotlib.pyplot as plt
```

In [40]:
```
# defining a function to play the signal
def play_signal(signal, sample_rate=44100):
    p = pyaudio.PyAudio()

    stream = p.open(format=pyaudio.paFloat32,
                    channels=1,
                    rate=sample_rate,
                    output=True)

    stream.write(signal.tobytes())

    stream.stop_stream()
    stream.close()

    p.terminate()
```

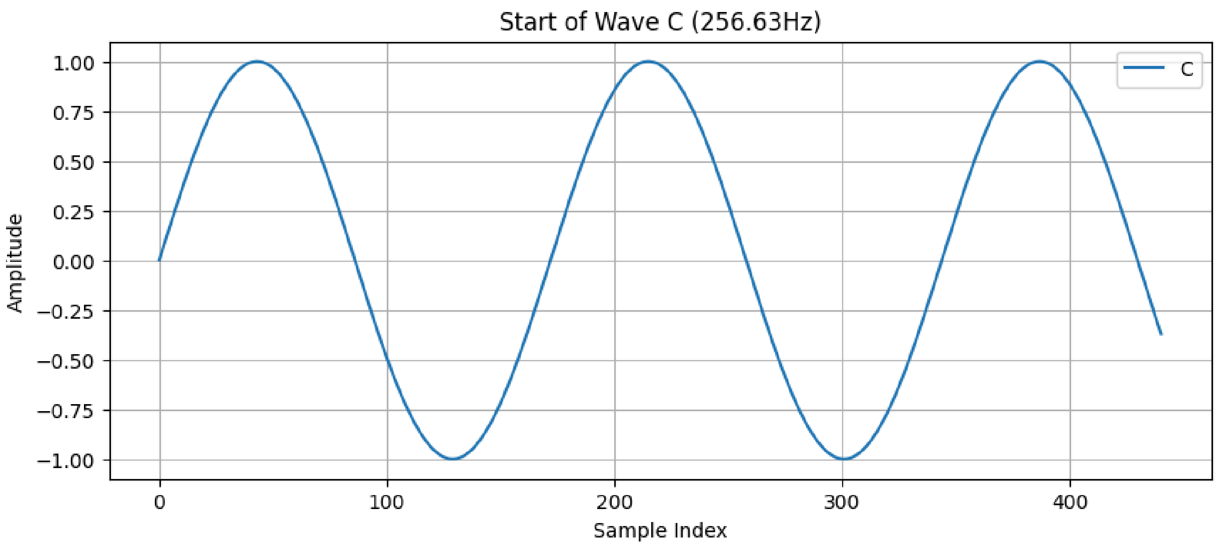
In [ ]:
```
# defining a function to see the signal
def plot_waveform(wave, label, title):
    plt.figure(figsize=(10, 4))
    plt.plot(wave, label=label)
    plt.title(title)
    plt.xlabel("Sample Index")
    plt.ylabel("Amplitude")
    plt.legend()
    plt.grid()
    plt.show()
```

In [12]:
```
# generating a sine wave with a frequency of 440Hz (note A) and a sine wave with frequency of 256.63 Hz (note C)
sample_rate = 44100
s_A = (np.sin(2 * np.pi * np.arange(sample_rate) * 440 / sample_rate)).astype(np.float32)
s_C = (np.sin(2 * np.pi * np.arange(sample_rate) * 256.63 / sample_rate)).astype(np.float32)
```

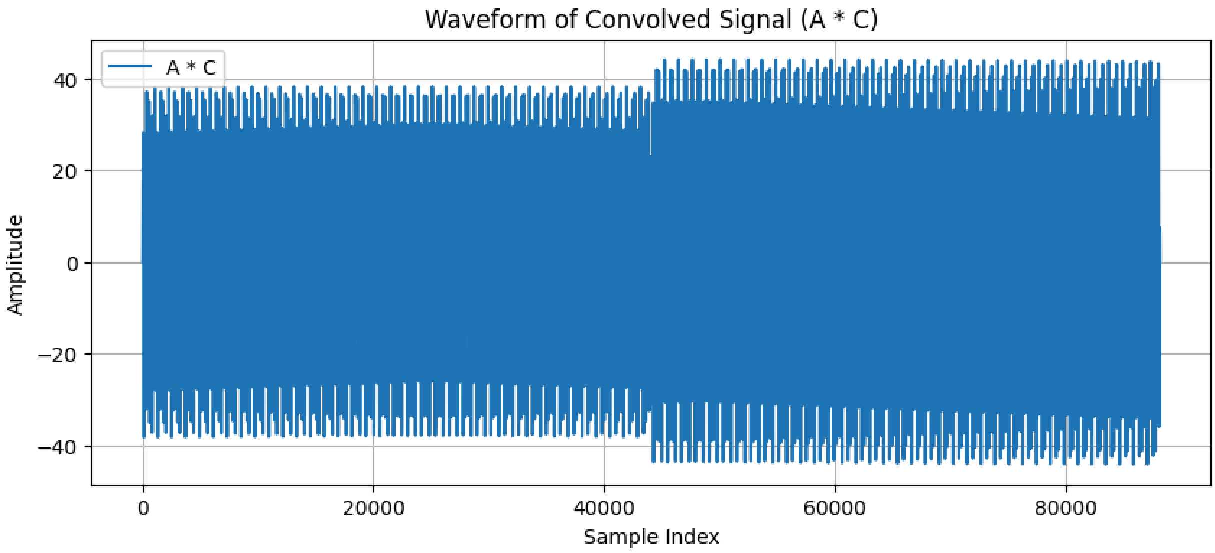
In [ ]:
```
plot_waveform(s_A[:441], "A", "Start of Wave A (440Hz)")
```



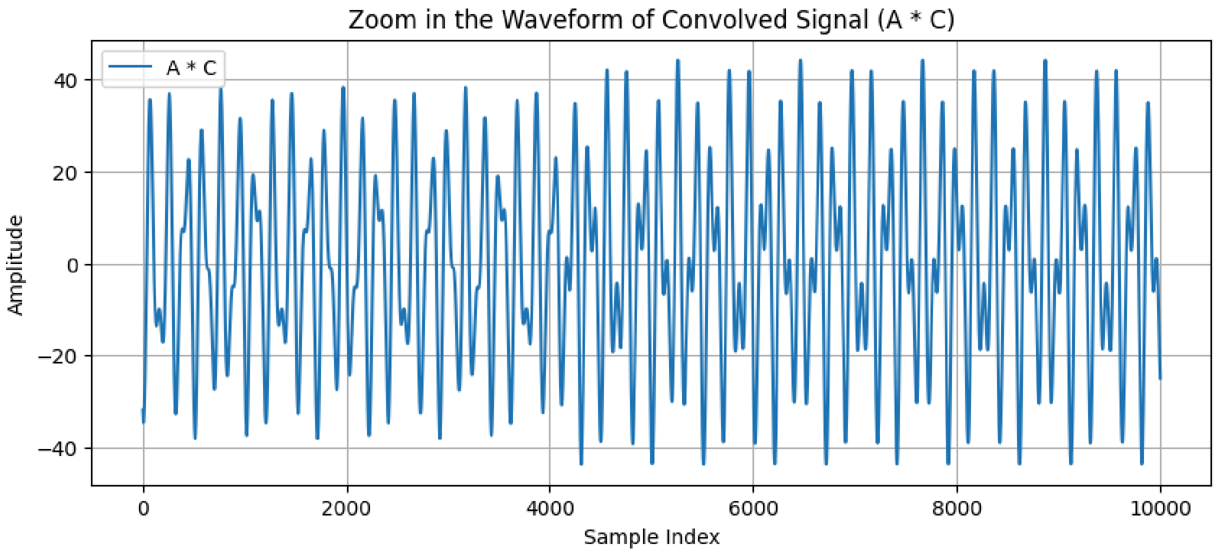
In [21]:
```
plot_waveform(s_C[:441], "C", "Start of Wave C (256.63Hz)")
```



In [ ]:
```
A_conv_C = np.convolve(s_A, s_C) #for perfomances reasons
plot_waveform(A_conv_C, label="A * C", title="Waveform of Convolved Signal (A * C)")
```



In [23]:
```
plot_waveform(A_conv_C[40000:50000], label="A * C", title="Zoom in the Waveform of Convolved Signal (A * C)")
```

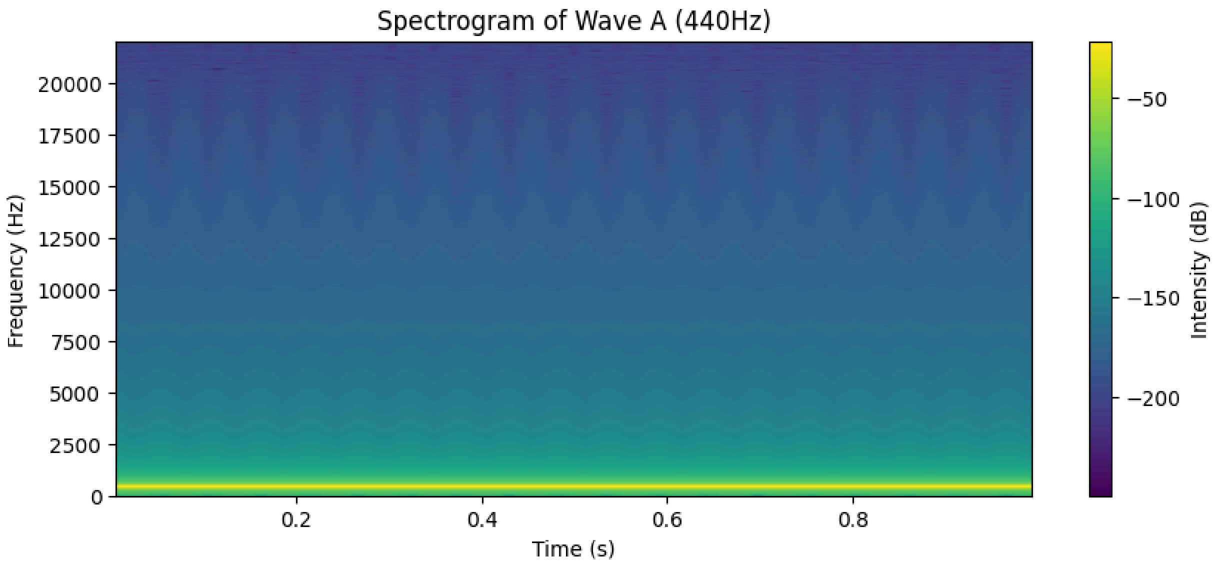


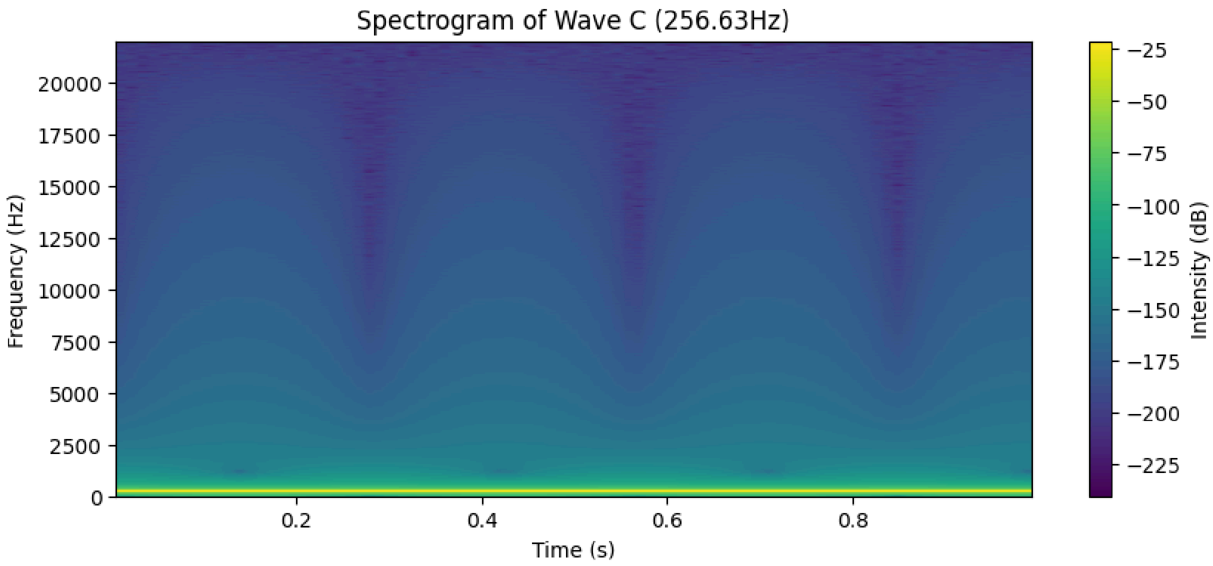

```
In [ ]: play_signal(s_A * 0.2) #play the original signal A with lower amplitude
play_signal(s_C * 0.2) #play the original signal C with lower amplitude
play_signal(A_conv_C * 0.02) #play the convolved signal, with very lower amplitude otherwise the signal clips
```

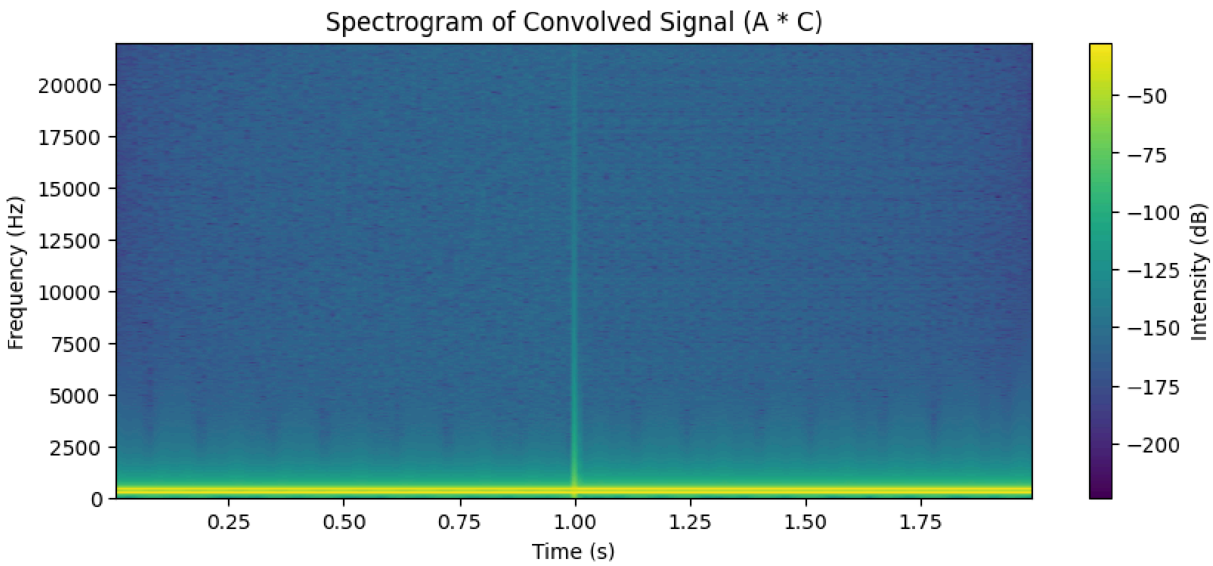
Analyzing the spectogram

```
In [30]: # defining a function to see the spectrogram
def plot_spectrogram(signal, title):
    plt.figure(figsize=(10, 4))
    plt.specgram(signal, Fs=sample_rate, NFFT=1024, noverlap=512, cmap='viridis')
    plt.title(title)
    plt.xlabel("Time (s)")
    plt.ylabel("Frequency (Hz)")
    plt.colorbar(label='Intensity (dB)')
    plt.show()
```

```
In [32]: plot_spectrogram(s_A, "Spectrogram of Wave A (440Hz)")
plot_spectrogram(s_C, "Spectrogram of Wave C (256.63Hz)")
plot_spectrogram(A_conv_C * 0.02, "Spectrogram of Convolved Signal (A * C)")
```







As seen in the Spectogram of the convolved waves, both frequencies [440 and 256.3] have high intensity, right in the middle, where the first lenghth is met there is a peak of intensity for all frequencies. Also, compared to the original waves, the convolved seems to have more presence of all frequencies.

Using it as a Reverb

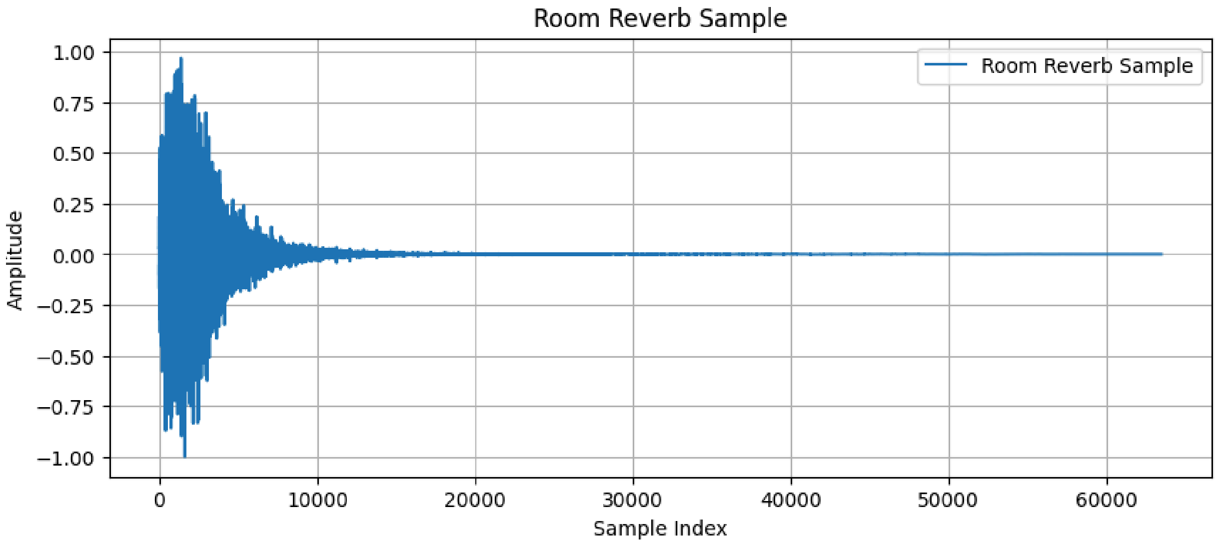
The concept of convolution as described in the class had me thinking about reverb. So we need a reverberation sample, it will be the x[·] vector that the h[·] will reverberate in, and h[·] can be any sound.

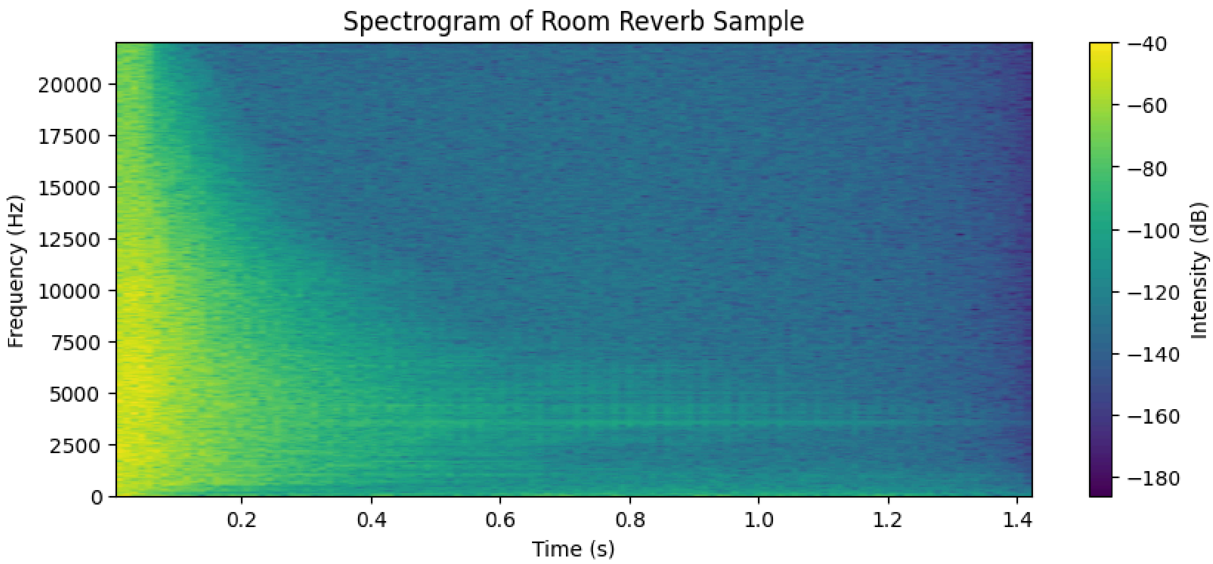
In [ ]:
```
from scipy.io import wavfile
```

In [60]:
```
def load_wav_file(file_path):
    sample_rate, data = wavfile.read(file_path)
    if data.ndim > 1:
        # If stereo, convert to mono by averaging channels
        data = np.mean(data, axis=1)
    # Normalize to float32 range
    data = data.astype(np.float32) / np.max(np.abs(data))
    # Ensure data is in the range [-1, 1]
    data = np.clip(data, -1.0, 1.0)
    return sample_rate, data
```

In [62]:
```
#Loading the reverb sample
sample_rate_reverb, room_reverb_sample = load_wav_file("room_reverb_sample.wav")

play_signal(room_reverb_sample)
plot_waveform(room_reverb_sample, "Room Reverb Sample", "Room Reverb Sample")
plot_spectrogram(room_reverb_sample, "Spectrogram of Room Reverb Sample")
```

```
C:\Users\mathe\AppData\Local\Temp\ipykernel_7400\3892187661.py:2: WavFileWarning: Chunk (non-data) not understoo
d, skipping it.
  sample_rate, data = wavfile.read(file_path)
```

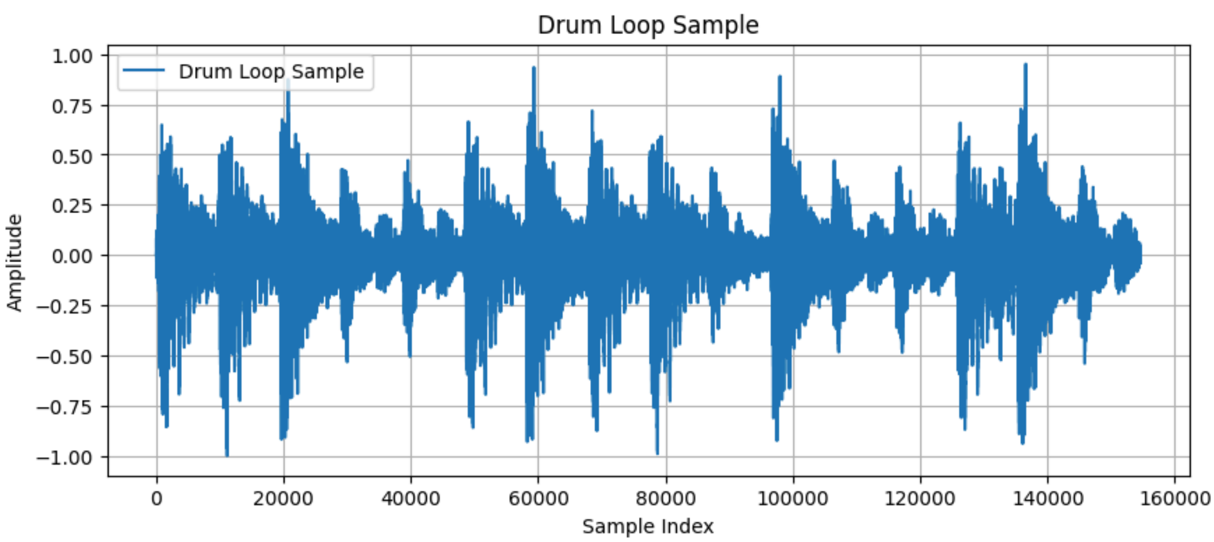


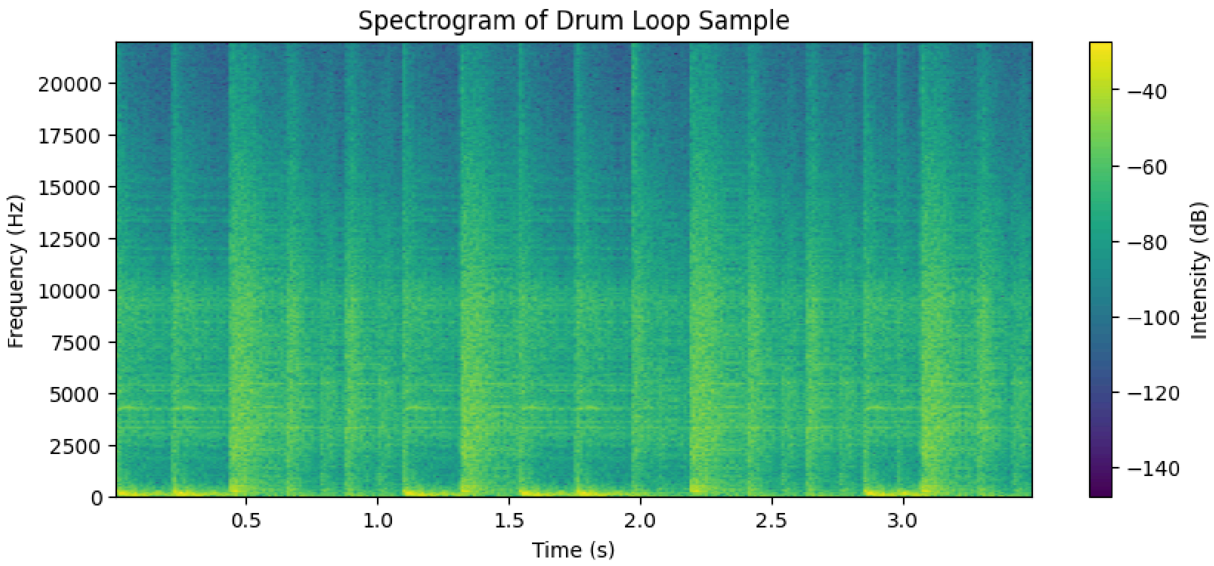


In [64]:
```
# Loading a drum loop sample
sample_rate_drum, drum_loop_sample = load_wav_file("drum_break_137.wav")
```

In [65]:
```
play_signal(drum_loop_sample, sample_rate=sample_rate_drum)
```

In [58]:
```
plot_waveform(drum_loop_sample, "Drum Loop Sample", "Drum Loop Sample") #we have L and R signals
plot_spectrogram(drum_loop_sample, "Spectrogram of Drum Loop Sample")
```





In [51]:
```
drum_reverb = np.convolve(drum_loop_sample, room_reverb_sample)
```

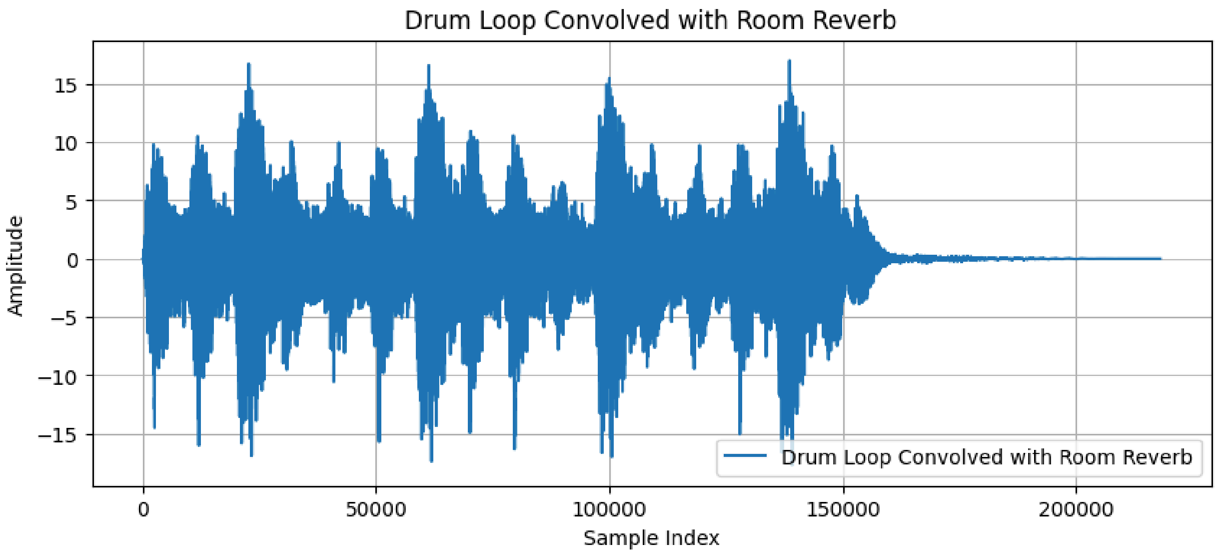
Playing the original and the convolved signals

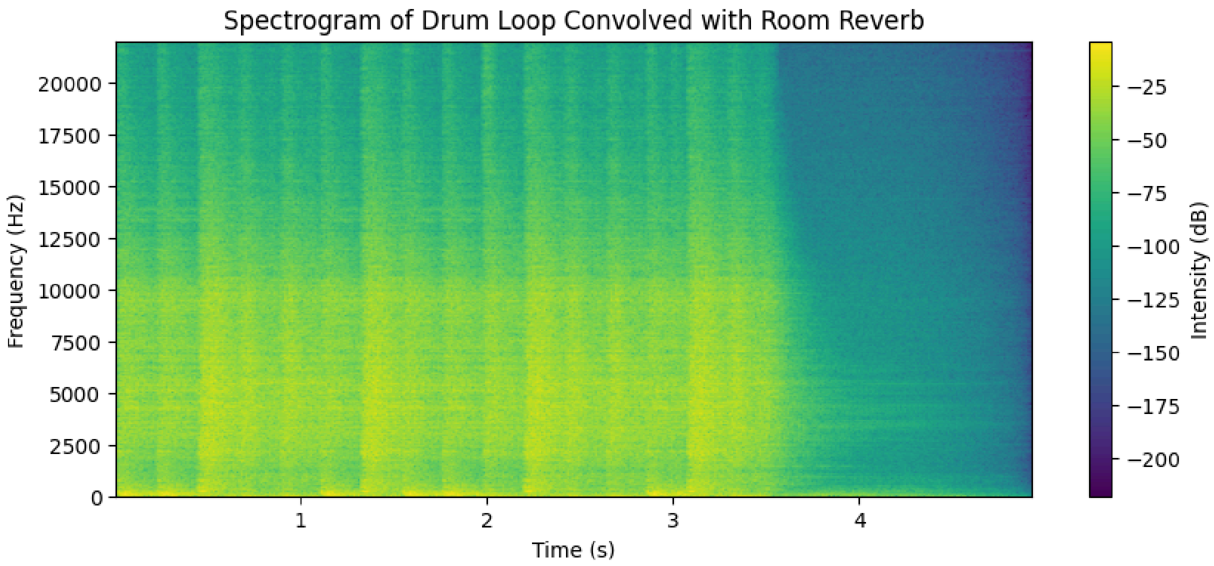
We can see the reverb!

In [66]:
```
play_signal(drum_loop_sample, sample_rate=sample_rate_drum) #play the original signal
```

In [67]:
```
play_signal(drum_reverb * 0.05, sample_rate=sample_rate_drum) #play the convolved signal with very lower amplituc
```

In [ ]:
```
plot_waveform(drum_reverb, "Drum Loop Convolved with Room Reverb", "Drum Loop Convolved with Room Reverb")
plot_spectrogram(drum_reverb, "Spectrogram of Drum Loop Convolved with Room Reverb")
```





Other reverb sounds

Lets try reverberating the drum through an open hihat and a very low kick

In [69]:
```
sample_rate_hh, openhh_sample = load_wav_file("open_hihat.wav")
sample_rate_kick, kick_sample = load_wav_file("low_kick.wav")
```

```
C:\Users\mathe\AppData\Local\Temp\ipykernel_7400\3892187661.py:2: WavFileWarning: Chunk (non-data) not understoo
d, skipping it.
  sample_rate, data = wavfile.read(file_path)
```

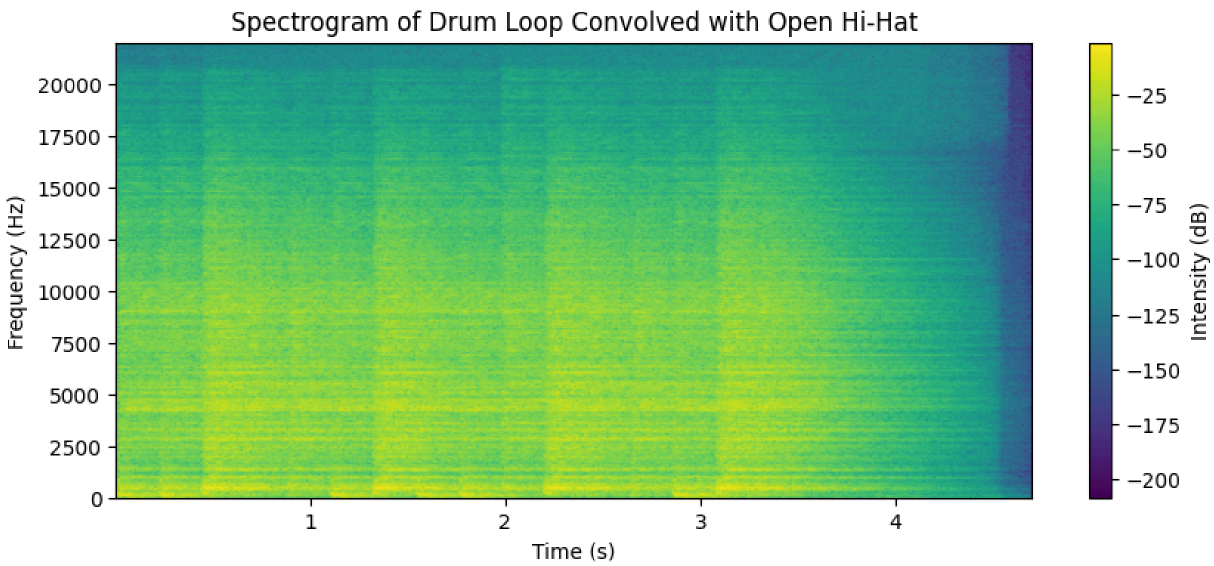
In [ ]:
```
play_signal(openhh_sample, sample_rate=sample_rate_hh)
play_signal(kick_sample, sample_rate=sample_rate_kick)
```

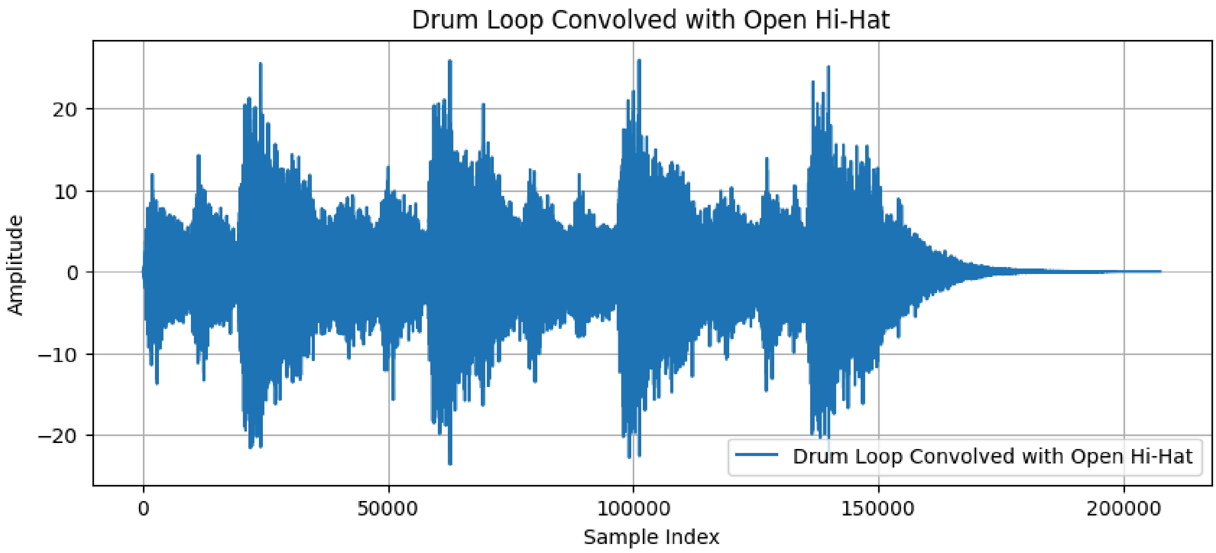
In [71]:
```
drum_loop_hh_reverb = np.convolve(drum_loop_sample, openhh_sample) #convolve the drum loop with the open hi-hat
```

In [85]:
```
play_signal(drum_loop_hh_reverb * 0.03, sample_rate=sample_rate_drum)
```

very nice highs!

In [92]:
```
plot_spectrogram(drum_loop_hh_reverb, "Spectrogram of Drum Loop Convolved with Open Hi-Hat")
plot_waveform(drum_loop_hh_reverb, "Drum Loop Convolved with Open Hi-Hat", "Drum Loop Convolved with Open Hi-Hat'
```



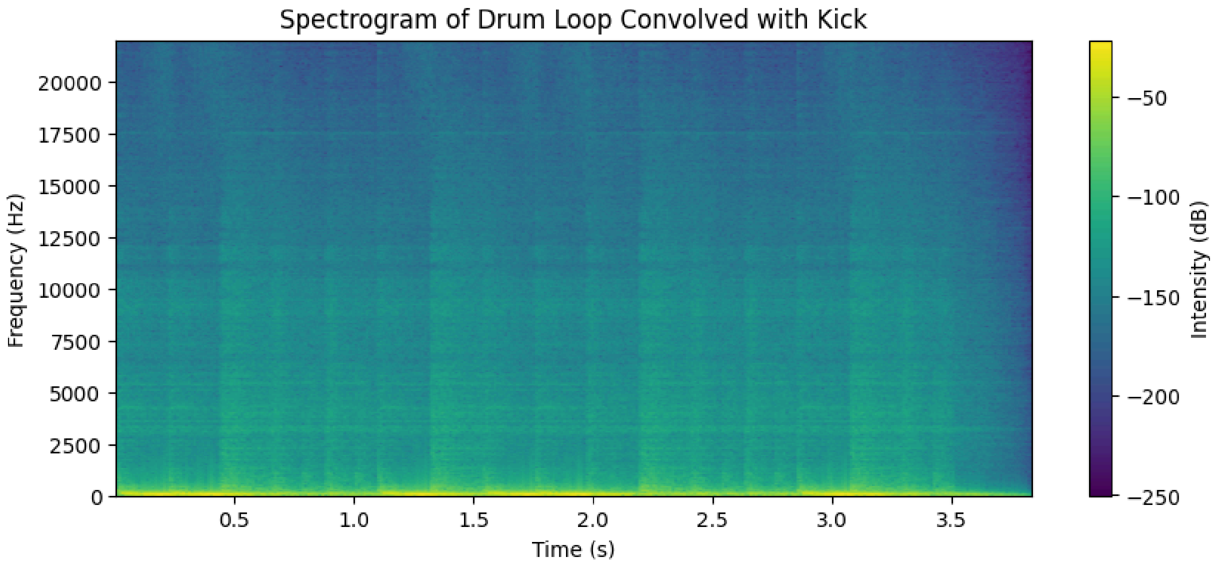


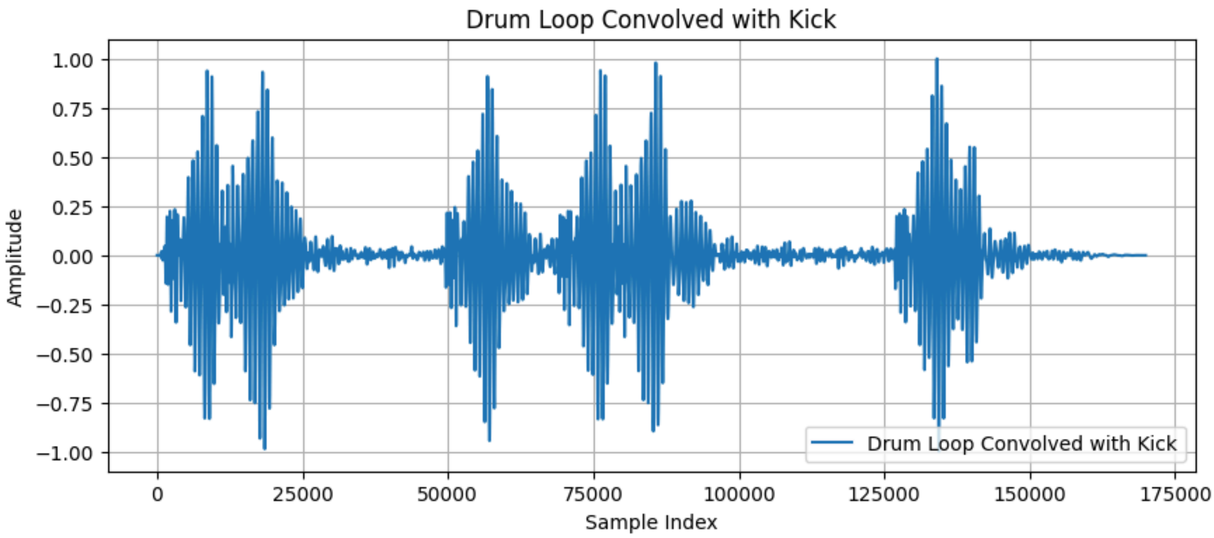
In [94]:
```
drum_loop_kick_reverb = np.convolve(drum_loop_sample, kick_sample) #convolve the drum loop with the kick
drum_loop_kick_reverb = drum_loop_kick_reverb / np.max(np.abs(drum_loop_kick_reverb)) #normalize the audio
```

In [98]:
```
play_signal(drum_loop_kick_reverb, sample_rate=sample_rate_drum)
```

wow, thats deep! All the high frequencies are cut!

In [96]:
```
plot_spectrogram(drum_loop_kick_reverb, "Spectrogram of Drum Loop Convolved with Kick")
plot_waveform(drum_loop_kick_reverb, "Drum Loop Convolved with Kick", "Drum Loop Convolved with Kick")
```





wow, thats deep!

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Date: March 17 2025