CS448 - Lab 3: Artificial Reverberation and Room Simulation

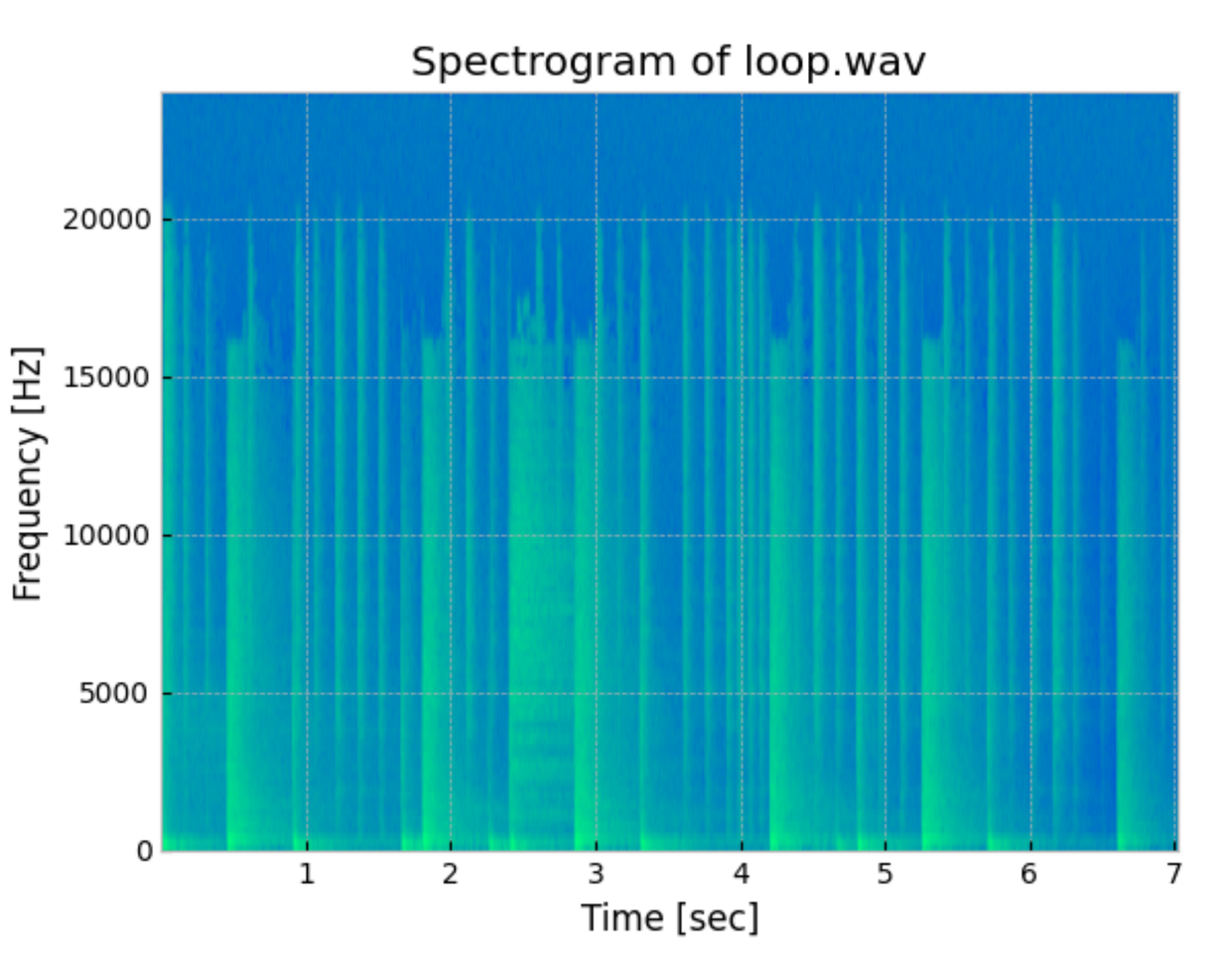
In this lab we will get some experience in simulating rooms digitally. We will design some simple reverbs using small filters and then work towards a more principled room simulator. You can use whichever sound you like for the following examples, usually a sound with sharp onsets is good for such experiments. One good example is the drum loop [https://drive.google.com/uc?export=download&id=16mql3xFN00AK5bP5k1VYJiEfH7ghHHSf]

In [ ]:
```python
### IMPORTS & SETUP ###

import matplotlib.pyplot as plt

import numpy as np
from scipy.io import wavfile
from scipy.signal import lfilter, convolve, fftconvolve

plt.style.use('bmh')
```

In [ ]:
```python
### UTILITIES ###


# Sound player function that plays array "x" with a sample rate "rate", and labels it with "label"
def sound(x, rate=8000, label=''):
    from IPython.display import display, Audio, HTML
    display(
        HTML('<style> table, th, td {border: 0px; }</style> <table><tr><td>' +
             label + '</td><td>' + Audio(x, rate=rate)._repr_html_()[3:] +
             '</td></tr></table>'))


# Function that normalizes a signal
def normalize_signal(x):
    return x / np.max(np.abs(x))


# Function that plots the spectrogram of a sound
def plot_spectrogram(input_sound, fs, title="Spectrogram"):
    plt.title(title)
    plt.specgram(input_sound, Fs=fs, cmap="winter")
    plt.xlabel('Time [sec]')
    plt.ylabel('Frequency [Hz]')
    plt.show()


def plot_filter_spectrogram(filter,
                            input_sound,
                            fs,
                            filter_title="Impulse response of the filter",
                            spec_title="Spectrogram"):
    plt.subplots_adjust(hspace=0.5)

    # Plot the filter's response
    plt.subplot(211)
    plt.plot(filter)
    plt.title(filter_title)
    plt.xlabel("samples")
    plt.ylabel("amplitude")

    # Plot the spectrogram
    plt.subplot(212)
    plt.title(spec_title)
    plt.specgram(input_sound, Fs=fs, cmap="winter")
    plt.xlabel('Time [sec]')
    plt.ylabel('Frequency [Hz]')

    plt.show()
```

In [ ]:
```python
# Load the input sound
fname = 'loop.wav'
fs, input_sound = wavfile.read(f'./data/{fname}')
input_sound = normalize_signal(input_sound)

sound(input_sound, rate=fs, label=fname)
plot_spectrogram(input_sound, fs, title=f"Spectrogram of {fname}")
```

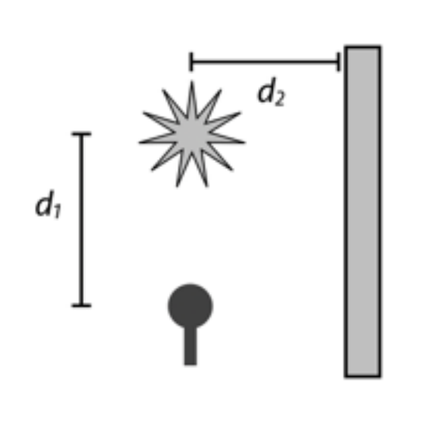
loop.wav -0:07



Part 1. Designing simple reverb using filters

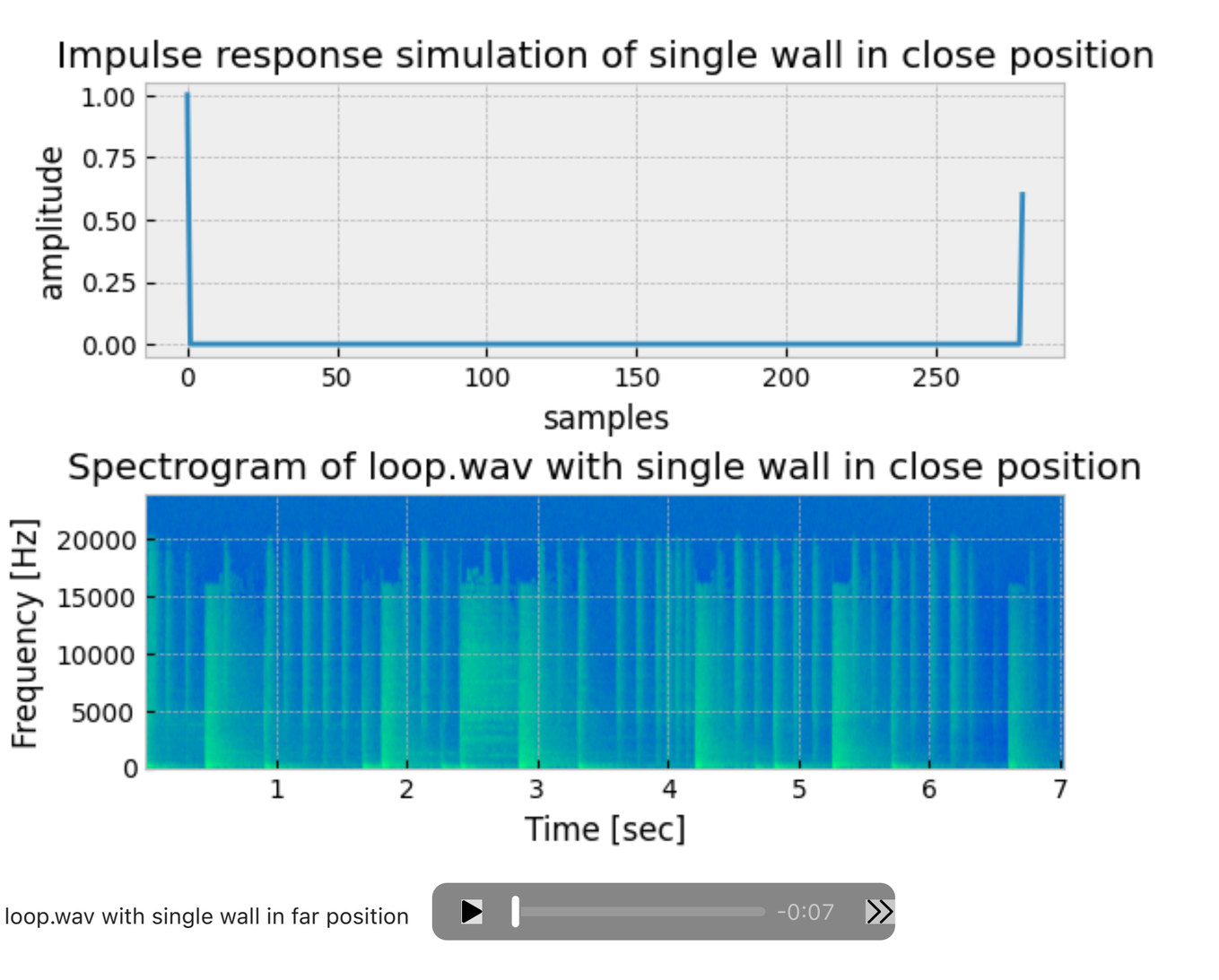
The simplest forms of reverbs can be designed using simple delays, comb and allpass filters. We will design one of each to get started.

a) A room with a single wall. For this reverb we will assume that we have a room that only has one reflective wall as shown below. Design a filter that simulates what the microphone will record. figure out the right filter in the case where $d_i = [0.1, 1]$ meters, and in the case where $d_i = [1, 10]$ meters. Let me know what the differences between these sounds are. To keep things simple, round the delays to an integer number and make up an approximate gain loss due to propagation and the wall bounce.

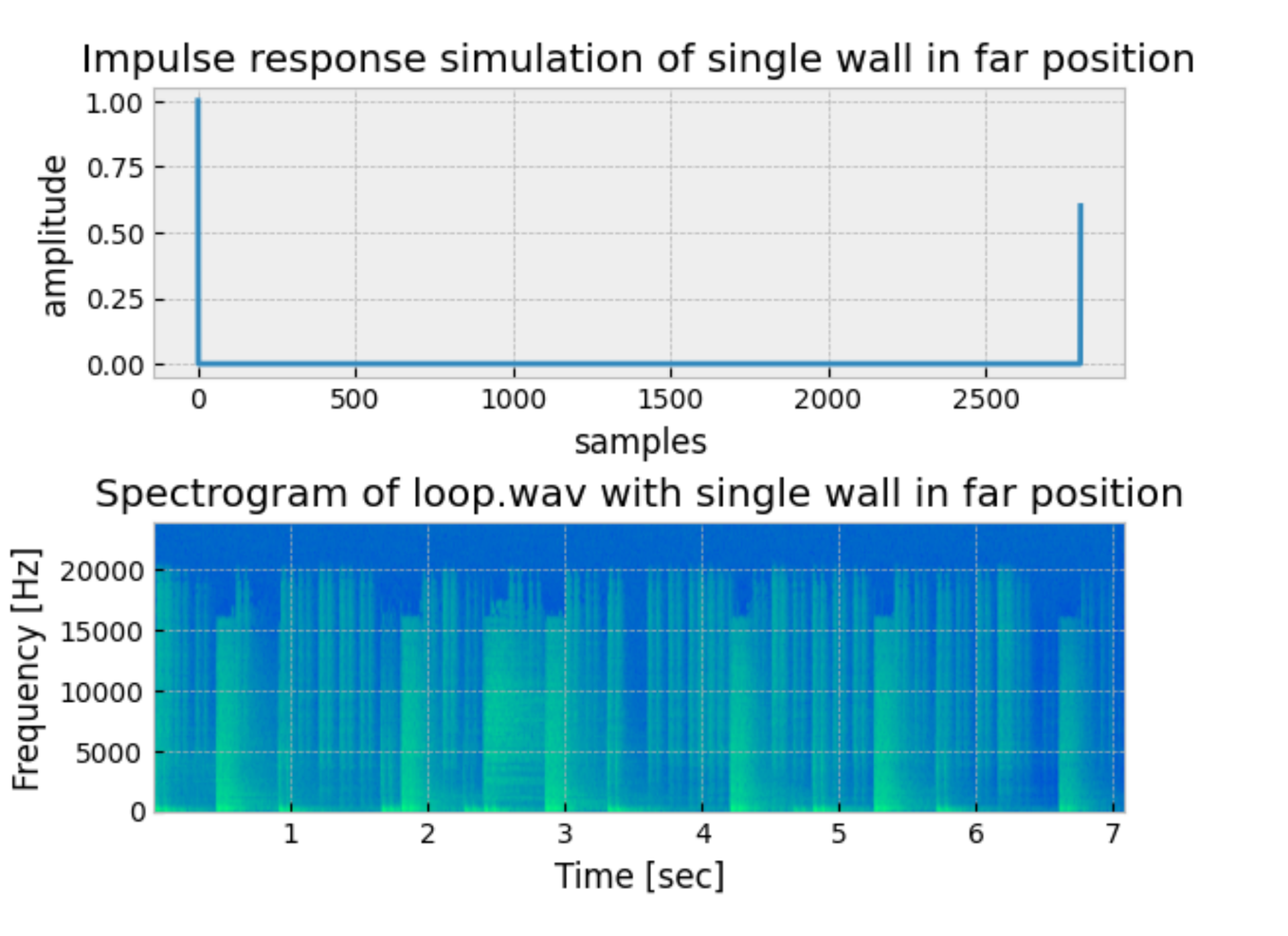


In [ ]:
```python
### PART 1A ###

reflect_gain = 0.6


# Returns a filter that simulates a single wall's reflections
def single_wall(dist_vec, reflect_gain, fs):
    total_dist = np.sqrt((dist_vec[0] / 2)**2 +
                         dist_vec[1]**2) * 2  # Pythagoras * 2
    samples = int(total_dist / 343 * fs)  # Convert to number of samples

    filter = np.zeros(int(samples))
    filter[0] = 1
    filter[-1] = reflect_gain

    return filter


# Simulate a single wall in close position
filter = single_wall([0.1, 1], reflect_gain, fs)
single_wall_close = convolve(input_sound, filter)

sound(single_wall_close,
      rate=fs,
      label=f'{fname} with single wall in close position')
plot_filter_spectrogram(
    filter,
    single_wall_close,
    fs,
    filter_title="Impulse response simulation of single wall in close position",
    spec_title=f"Spectrogram of {fname} with single wall in close position")

# Simulate a single wall in far position
filter = single_wall([1, 10], reflect_gain, fs)
single_wall_far = convolve(input_sound, filter)

sound(single_wall_far,
      rate=fs,
      label=f'{fname} with single wall in far position')
plot_filter_spectrogram(
    filter,
    single_wall_far,
    fs,
    filter_title="Impulse response simulation of single wall in far position",
    spec_title=f"Spectrogram of {fname} with single wall in far position")
```

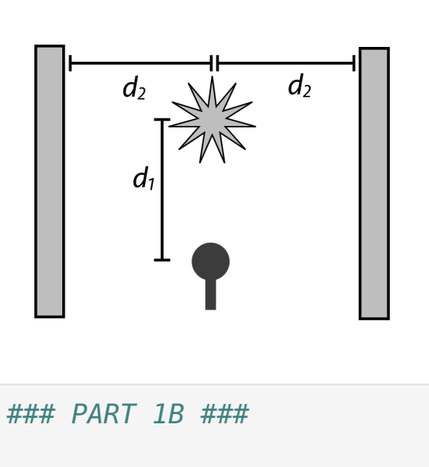
loop.wav with single wall in close position -0:07



loop.wav with single wall in far position -0:07

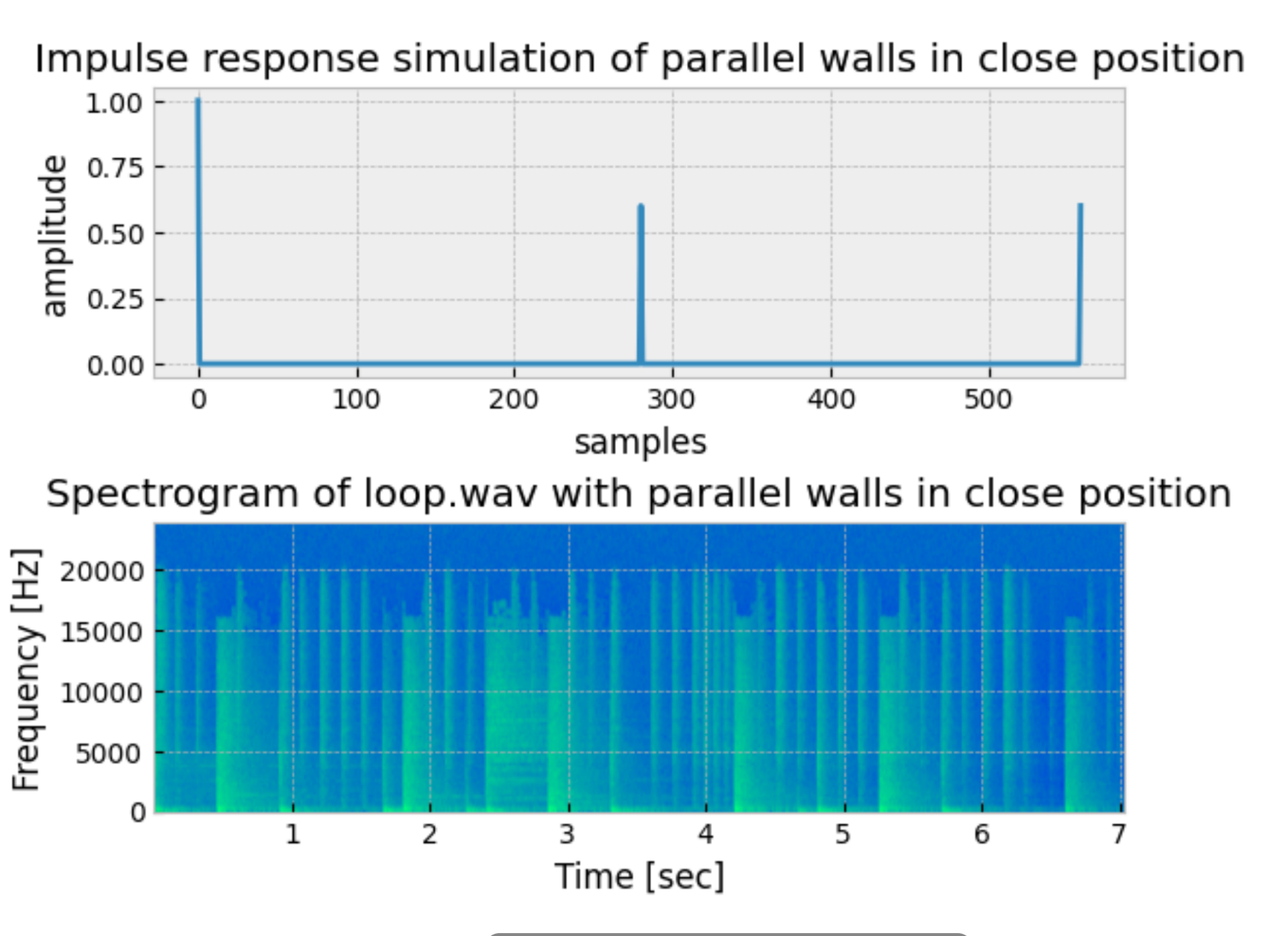


b) Now we will use a room geometry as shown below. Design the proper filter again, for $d_i = [0.1, 1]$ meters and then for $d_i = [1, 10]$ meters. Comment on the audible differences between these two settings.

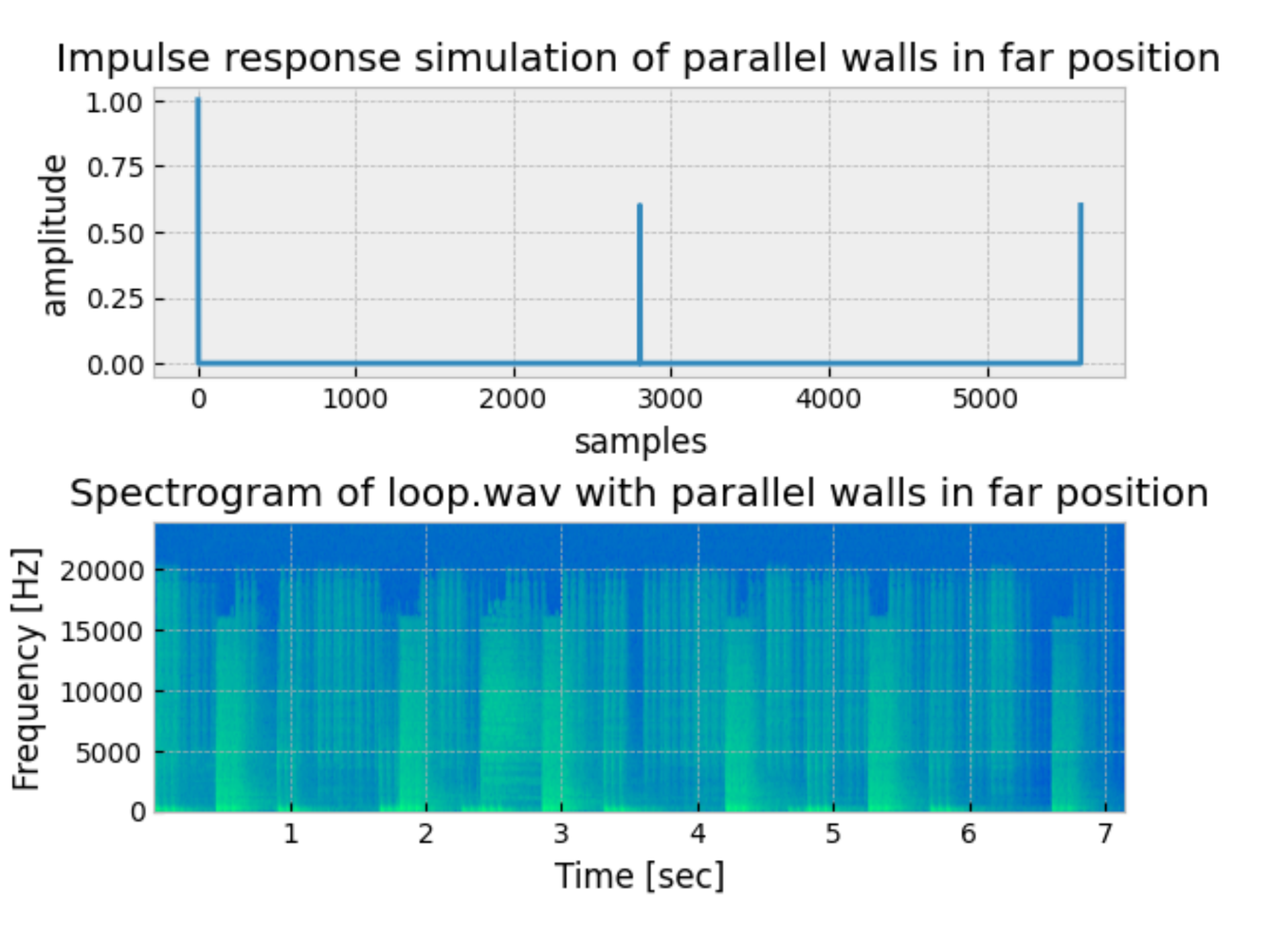


In [ ]:
```python
### PART 1B ###

reflect_gain = 0.6


# Returns a filter that simulates 2 parallel walls' reflections
def parallel_walls(dist_vec, reflect_gain, fs):
    dist1 = np.sqrt((dist_vec[0] / 2)**2 +
                    dist_vec[1]**2) * 2  # Pythagoras * 2
    dist2 = np.sqrt((dist_vec[0] / 4)**2 + dist_vec[1]**2) * 4

    samples1 = int(dist1 / 343 * fs)  # Convert to number of samples
    samples2 = int(dist2 / 343 * fs)

    filter = np.zeros(int(samples2))
    filter[0] = 1
    filter[int(samples1)] = reflect_gain
    filter[-1] = reflect_gain

    return filter


# Simulate parallel walls in close position
filter = parallel_walls([0.1, 1], reflect_gain, fs)
parallel_walls_close = convolve(input_sound, filter)

sound(parallel_walls_close,
      rate=fs,
      label=f'{fname} with parallel walls in close position')
plot_filter_spectrogram(
    filter,
    parallel_walls_close,
    fs,
    filter_title=
    "Impulse response simulation of parallel walls in close position",
    spec_title=f"Spectrogram of {fname} with parallel walls in close position")

# Simulate parallel walls in far position
filter = parallel_walls([1, 10], reflect_gain, fs)
parallel_walls_far = convolve(input_sound, filter)

sound(parallel_walls_far,
      rate=fs,
      label=f'{fname} with parallel walls in far position')
plot_filter_spectrogram(
    filter,
    parallel_walls_far,
    fs,
    filter_title=
    "Impulse response simulation of parallel walls in far position",
    spec_title=f"Spectrogram of {fname} with parallel walls in far position")
```

loop.wav with parallel walls in close position -0:07



loop.wav with parallel walls in far position -0:07



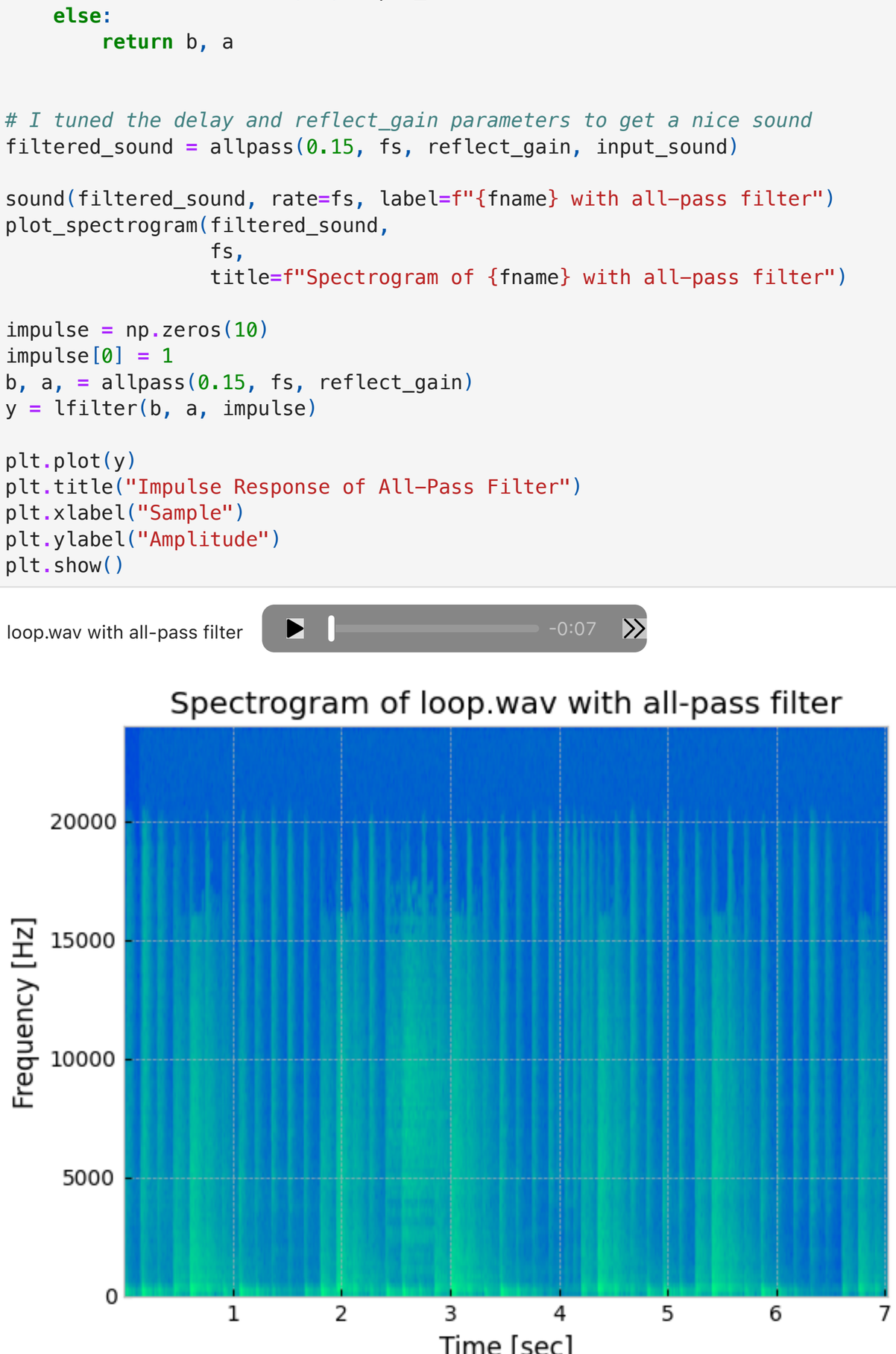
Conclusions

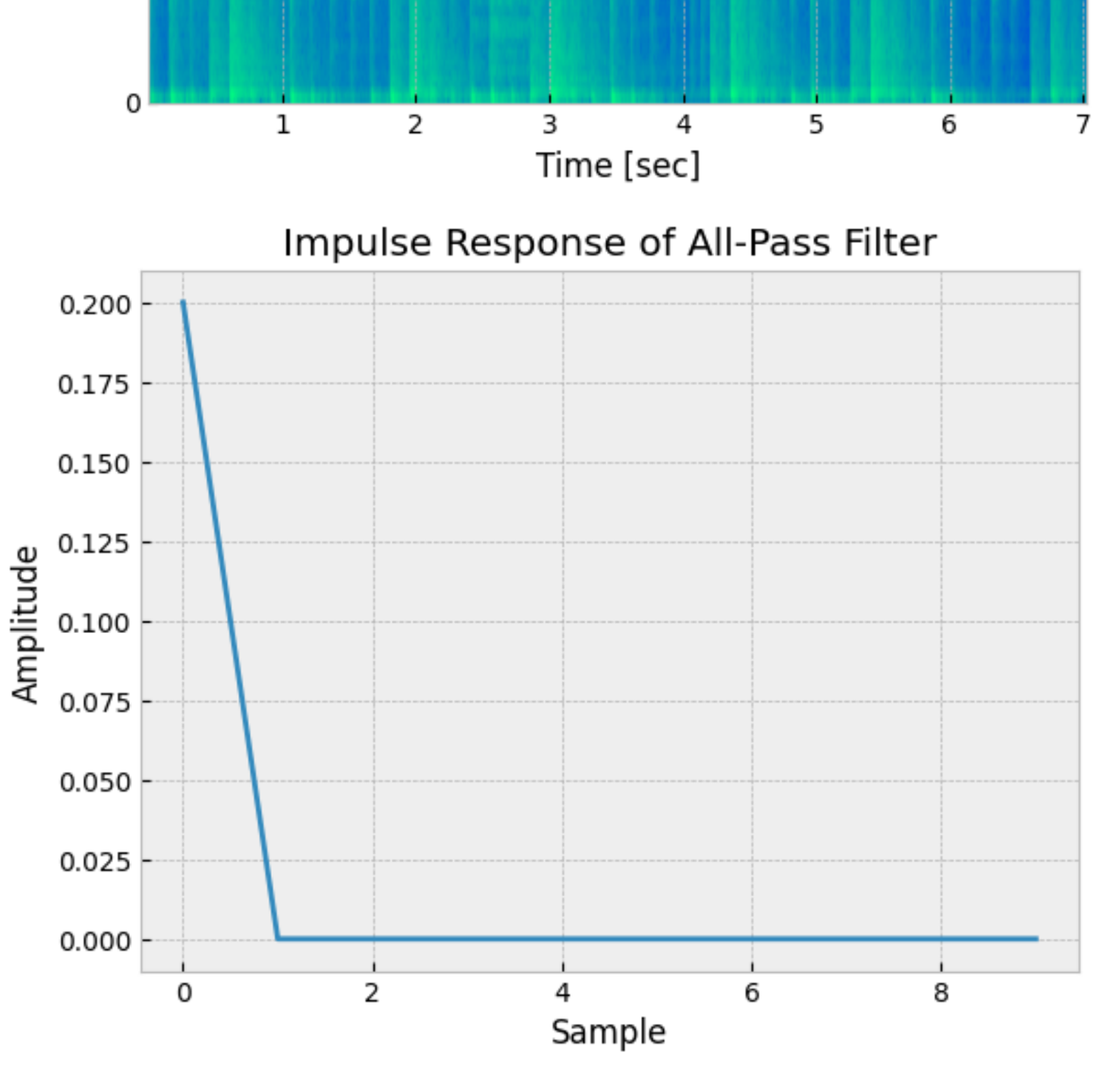
	Close Position	Far Position
Single wall	The audio seems to have lost some of it's power end thump and feels a little hazy to me.	There is a much more apparent fast echo which can be heard very clearly with the snare and hi-hats.
Parallel walls	The audio seems like it has reduced in quality even further than in the single wall scenario almost as though a lot of frequencies have been scooped out using an equalizer.	The echos are a lot more apparent than in the single wall simulation and in opinion a little too much. While tweaking the reflection gain, I was able to find a few sweet spots where the echo sounded nice and "musical".

c) Since for small values of d the output above gets its spectrum altered, we will use an allpass filter to make a more natural sounding echo pattern. Use the formulation shown in the lecture slides and compare that filter's frequency response with the response from the filter in the section b) above.

In [ ]:
```python
### PART 1C ###

reflect_gain = 0.2


# Returns an all-pass filter
def allpass(delay, fs, g, input_sound=None):
    # Initialize the numerator and denominator of the filter
    samples = int(delay * fs)
    a = np.zeros(samples + 1)
    b = np.zeros(samples + 1)

    # Set the coefficients
    a[0] = 1
    b[0] = g
    a[samples] = g
    b[samples] = 1

    if input_sound is not None:
        return lfilter(b, a, input_sound)
    else:
        return b, a


# I tuned the delay and reflect_gain parameters to get a nice sound
filtered_sound = allpass(0.15, fs, reflect_gain, input_sound)

sound(filtered_sound, rate=fs, label=f"{fname} with all-pass filter")
plot_spectrogram(filtered_sound,
                 fs,
                 title=f"Spectrogram of {fname} with all-pass filter")

impulse = np.zeros(10)
impulse[0] = 1
b, a, = allpass(0.15, fs, reflect_gain)
y = lfilter(b, a, impulse)

plt.plot(y)
plt.title("Impulse Response of All-Pass Filter")
plt.xlabel("Sample")
plt.ylabel("Amplitude")
plt.show()
```

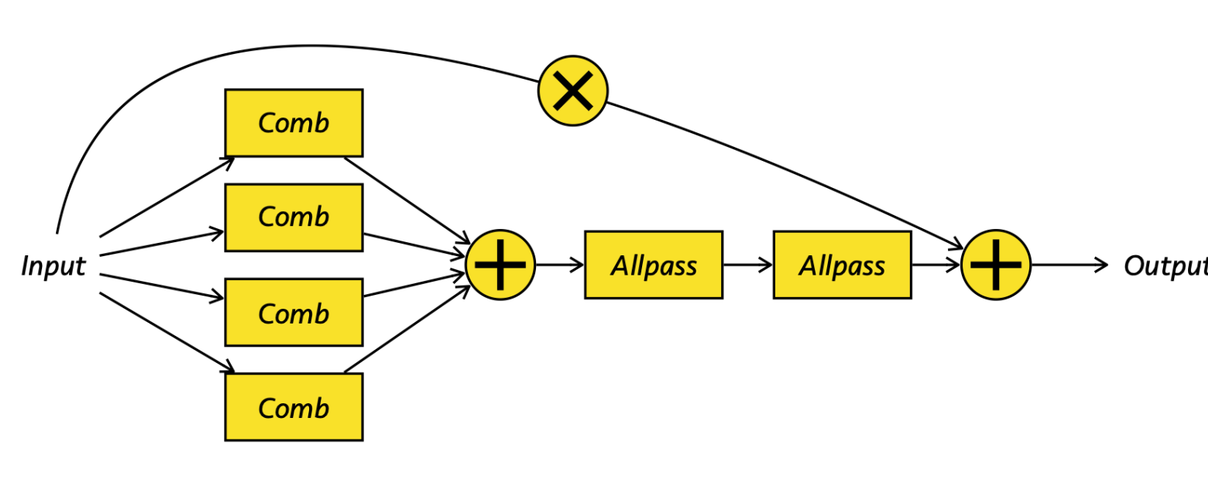
loop.wav with all-pass filter -0:07





Part 2. Schroeder reverberators

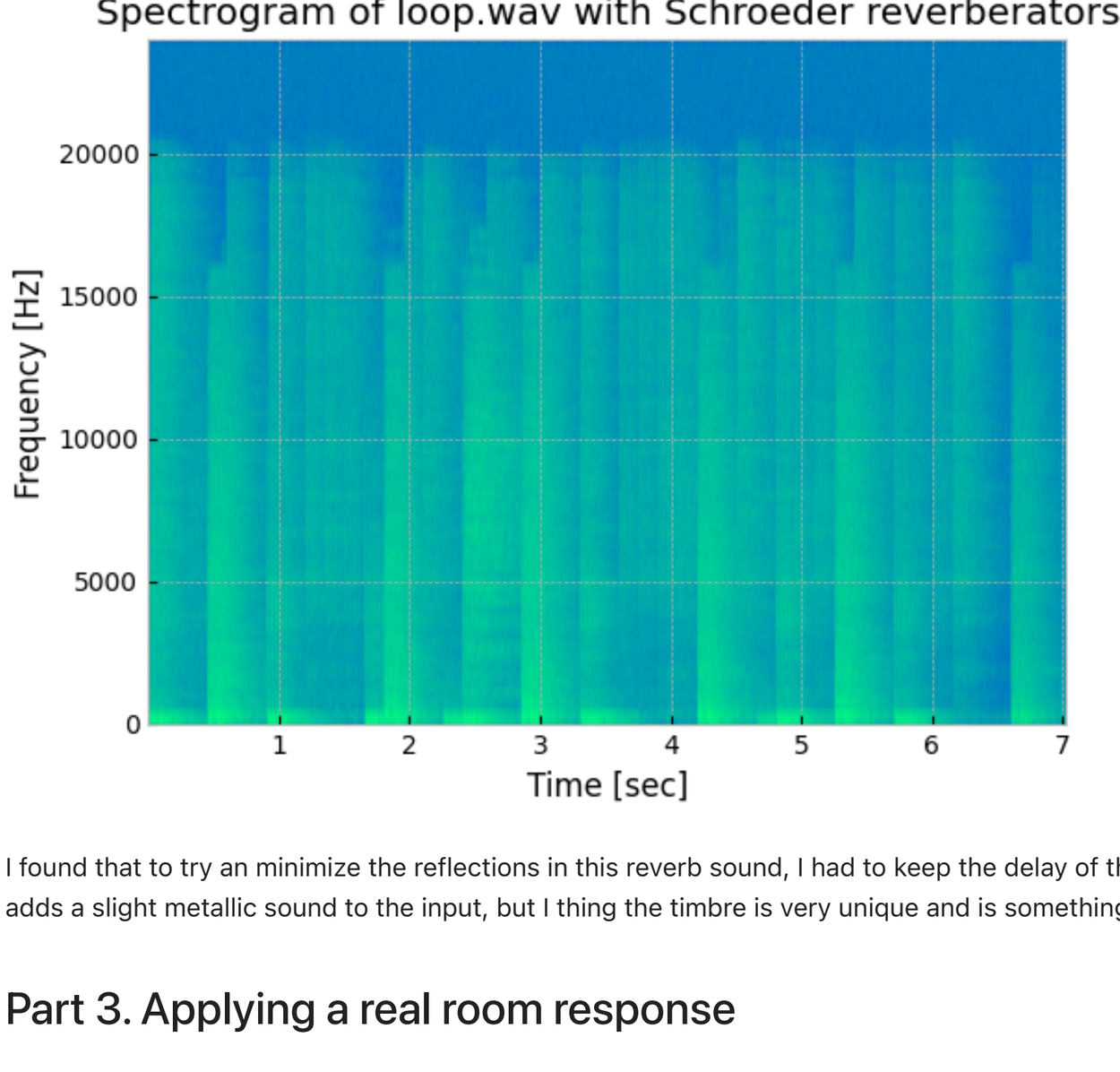
Now we will combine multiple filters to make a more serious sounding reverb. Implement the structure shown below.



Try to find the necessary parameters for the above structure that makes it sound as good as possible (not too busy, not too subtle). Hint: There is no right answer! Try a few things, see what sounds best and explain why you used the parameters you decided to use. As you come across some bad sounding cases, describe what the problem was.

In [ ]:
```python
### PART 2 ###
# Returns a comb filter
def comb_filter(delay, fs, g, input_sound=None):
    # Initialize the numerator and denominator of the filter
    samples = int(delay * fs)
    a = np.zeros(samples + 1)
    b = np.zeros(samples + 1)

    # Set the coefficients
    a[0] = 1
    b[0] = 1
    a[samples] = g

    if input_sound is not None:
        return lfilter(b, a, input_sound)
    else:
        return b, a


# Setting the parameters for the Schroeder reverberators
comb_1 = comb_filter(0.01, fs, .6, input_sound)
comb_2 = comb_filter(0.009, fs, .2, input_sound)
comb_3 = comb_filter(0.015, fs, .7, input_sound)
comb_4 = comb_filter(0.02, fs, .3, input_sound)

comb_sum = comb_1 + comb_2 + comb_3 + comb_4

allpass_1 = allpass(0.037, fs, .6, comb_sum)
allpass_2 = allpass(0.03, fs, .6, allpass_1)

# Adding the input sound to the Schroeder reverberators
effected_sound = input_sound + allpass_2

sound(effected_sound, rate=fs, label=f"{fname} with Schroeder reverberators")
plot_spectrogram(effected_sound,
                 fs,
                 title=f"Spectrogram of {fname} with Schroeder reverberators")
```

loop.wav with Schroeder reverberators -0:07



I found that to try an minimize the reflections in this reverb sound, I had to keep the delay of the allpass filters < 0.04. I kept messing around with various combinations of delays and gains, and I found that this combination sounded really cool to me. It adds a slight metallic sound to the input, but I thing the timbre is very unique and is something I would add to my own productions. I would characterize the reverb as very short and tight.

Part 3. Applying a real room response

To apply a more realistic filter we need to use a real room response. Download the St. Andrew's church room impulse response (Stereo) from:

[https://openairlib.net/?page_id=683]

Make sure that the sound you will convolve with it will be at the same sampling rate. This impulse response captures that church's RIR using two channels. To synthesize a sound in that room you need to convolve it with the impulse response. If your test sound is a single-channel recording, you can simply convolve it with each of the two impulse responses and then use the resulting outputs as the left and right channel. You will notice that using the `scipy.signal.lfilter` command for this operation can take very long to compute. For faster convolutions we need to perform this filtering in the frequency domain (using the fast convolution algorithm that we mentioned in the filtering lecture). The function `numpy.convolve` automatically does that (if you use an up to date version), otherwise you can directly use `scipy.signal.fftconvolve`. Verify that the resulting output sounds like it has been placed inside a church.

In [ ]:
```python
fname_RR = "st-andrews-church (stereo).wav"
fs_RR, curch_response = wavfile.read(f'./data/{fname_RR}')

left_channel = curch_response[:, 0]
right_channel = curch_response[:, 1]

left_convolved = fftconvolve(left_channel, input_sound)
right_convolved = fftconvolve(right_channel, input_sound)

convolved_sound = np.array([left_convolved, right_convolved])

sound(convolved_sound, rate=fs_RR, label=f"{fname} convolved with {fname_RR}")
```

loop.wav convolved with st-andrews-church (stereo).wav -0:09

Part 4 (extra credit). Measuring a room response.

Measuring a room response is crucial when trying to emulate a real room. In this exercise we will make a simple room response measurement. We will use the technique shown in the lecture slides.

a) Generate a chirp to use as our room excitation signal.

b) Go to a (preferably quiet) room that you want to measure, play the chirp from a speaker and record the outcome (the easiest way to do that is to save this sequence as a wave file and play it while you record a new sound file). You can instead use the real-time I/O code from Lab 0. In doing so you will obtain the sequence $y[t]$ from slide 36.

c) Now you will need to perform the actual deconvolution. Take $x[t]$ and $y[t]$, compute their DFTs to get $X[\omega]$ and $Y[\omega]$, and estimate $H[\omega]$ (remember to zero pad the inputs since the output of the convolution will be about twice as long as the inputs). Take the inverse DFT of H to get the time domain representation of the room impulse response. Does it look like one?

d) Convolve the resulting response with the test sound above (make sure you use the same sample rates). Does it sound reverberated? This won't be perfect. What sounds "wrong" with it? Can you speculate why?

In [ ]:
```python
# YOUR CODE HERE
raise NotImplementedError()
```