Reverberation Removal

Now we have been given an audio file with the noise of a highly reverberant room that we want to remove. Because the acoustics of room would be hard to model in terms of exact delays, we send a short pulse p(n), to mimic an impulse, into the room and record the response of the room r(n) to this input. Having the input and output of a system we can determine the transfer function of the system, i.e. the acoustics of the room, and we can use this to run an inverse filtering operation to recover the originally desired audio in the recording.

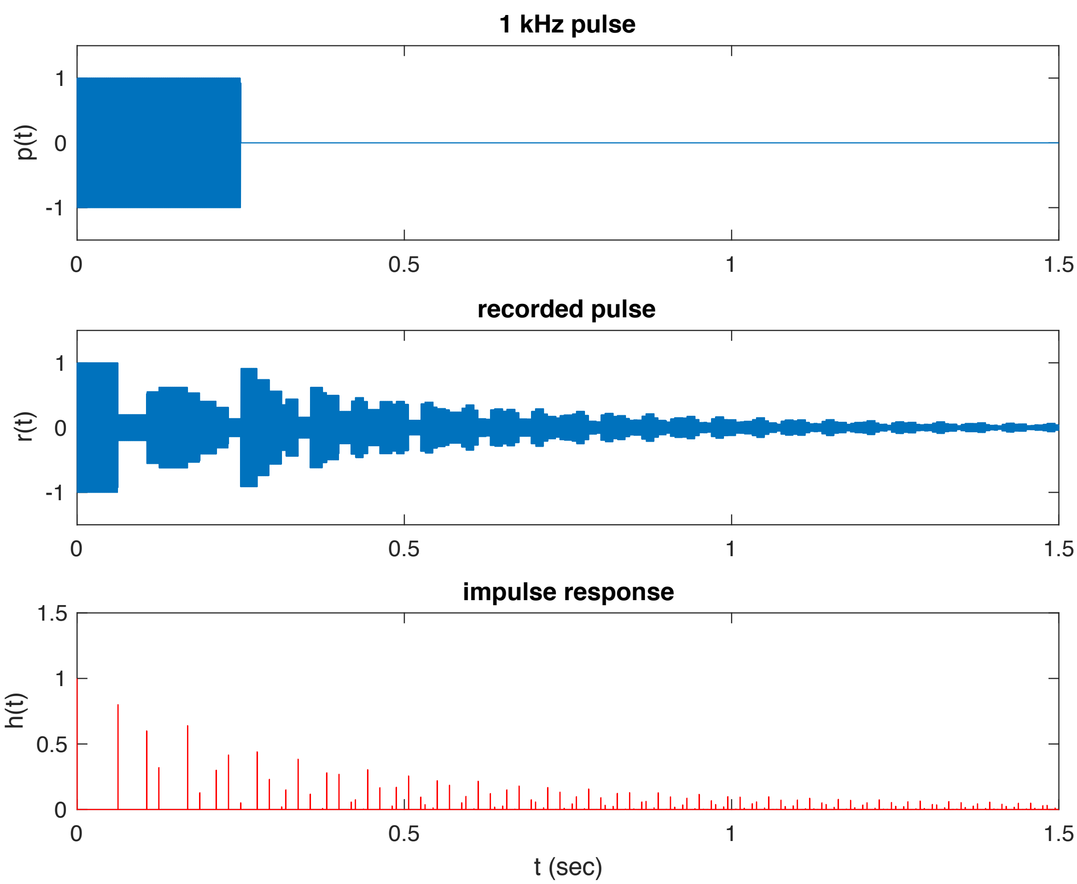
In the z-domain, we have that

(22) P(z)H(z) = R(z)

which gives,

(23) $$H(z) = R(z)\frac{1}{P(z)}$$

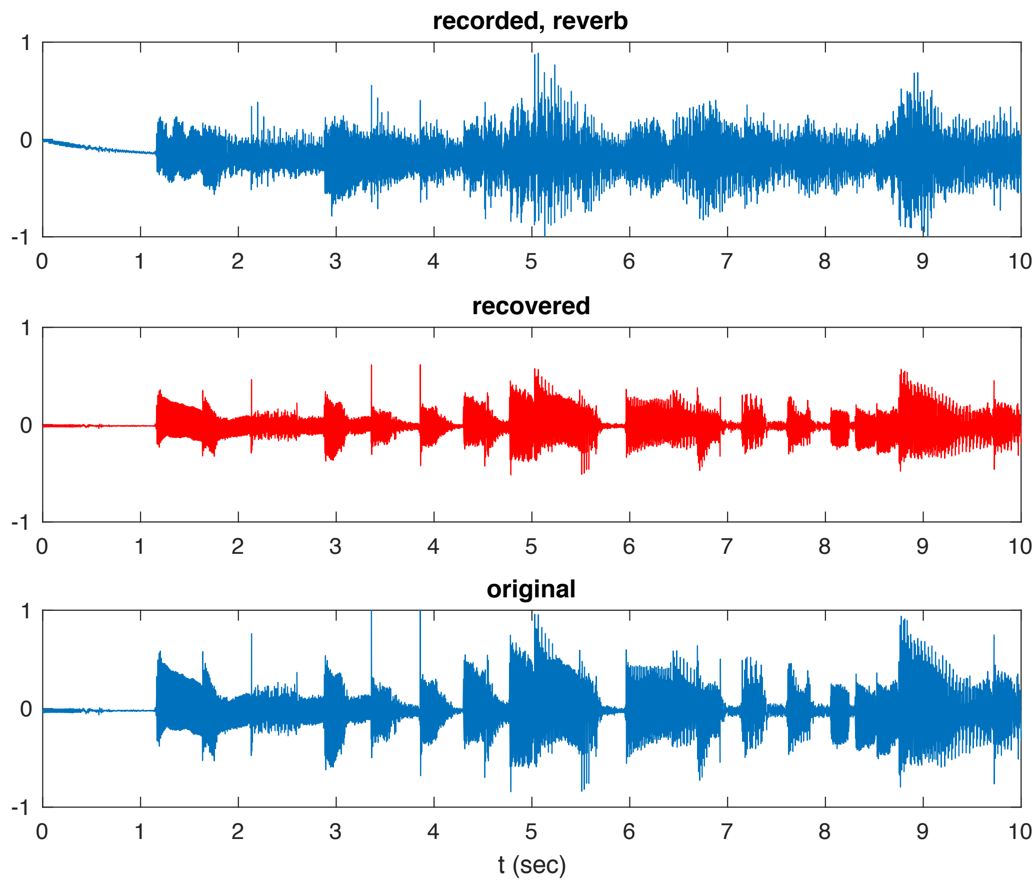
showing that if we run the known signal R(z) through a filter whose transfer function is the inverse of P(z) and take the inverse z transform, we obtain the "impulse" response of the room. Doing this using the filter function in MATLAB we obtain the following plots.



To recover the original signal we send the recording through an inverse filter with this impulse response.

(24) $$Y(z) = \frac{1}{H(z)}X(z)$$

Doing this using the filter function we obtain these results. (See Appendix A.2 for MATLAB code).



Again we can see that the recovered audio is practically the same as the original, By listening to the original and recovered audio, we hear that any small differences are inaudible.

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Appendix 1.b: DeReverb (Reverberation Removal)

Import audio file

```
[x, fs] = audioread('reverb.wav');
% Import original to compare the results
[s, fs] = audioread('original.wav');
% Pulse signal
[p, fs] = audioread('pulse.wav');
% Pulse recorded in room acoustics
[r, fs] = audioread('pulse_rec.wav');
% Obtain the impulse response of the room acoustics
h = filter(1,p,r);
```

Plot the signals

```
t = 1:24000;
subplot(3,1,1);
plot(t/fs,p); title('1 kHz pulse'); axis([0,1.5,-1.5,1.5]);
ylabel('p(t)');
subplot(3,1,2);
plot(t/fs,r); title('recorded pulse'); axis([0,1.5,-1.5,1.5]);
ylabel('r(t)');
subplot(3,1,3);
plot(t/fs,h,'Color','r'); title('impulse response'); xlabel('t
 (sec)');
axis([0,1.5,0,1.5]); ylabel('h(t)');
```

Remove the Room Reverberation

```
y = filter(1,h,x);
```

Plot the Results

```
t = 1:10*fs;
subplot(3,1,1);
plot(t/fs,x); title('recorded, reverb');
subplot(3,1,2);
plot(t/fs,y,'Color','r'); title('recovered');
```

```
subplot(3,1,3);
plot(t/fs,s); title('original'); xlabel('t (sec)');
```

Output the Recovered Signal

```
audiowrite('ReverbRemoved.wav',y,fs);
```

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