ELECTRICAL AND ELECTRONICS ENGINEERING INSTITUTE

EE 286: Digital Audio Signal Processing

Exercise 2: Reverberation Time

Reverberation time is the time required for the sound to "fade away" or decay in a closed space. Sound in a room will repeatedly bounce off surfaces such as the floor, walls, ceiling, windows or tables.

When these reflections mix, a phenomenon known as reverberation is created. Reverberation reduces when the reflections hit surfaces that can absorb sound such as curtains, chairs and even people.

The reverberation time of a room or space is defined as the time it takes for sound to decay by 60dB. For example, if the sound in a room took 10 seconds to decay from 100dB to 40dB, the reverberation time would be 10 seconds. This can also be written as the T_{60} time.

However, it is often very difficult to accurately measure the T_{60} time as it may not be possible to generate a sound level that is consistent and stable enough, especially in large rooms or spaces.

To get around this problem, it is more common to measure the T_{20} and T_{30} times and to then multiply these by 3 and 2 respectively to obtain the overall T_{60} time.

The T_{20} and T_{30} values are usually called "late reverberation times" as they are measured a short period of time after the noise source has been switched off or has ended.

The objective of this exercise is to measure the reverberation time of a room, and to determine its effect on different audio signals.

Measuring the T_{20} and T_{30} values

To measure these values, a sound source is used and this can either be an interrupted source, such as a loudspeaker or an impulsive noise source such as a starting pistol. The interrupted method is most commonly used as the sound source can be calibrated and controlled accurately, allowing for more repeatable measurements.

The measurement of reverberation time typically follows this process:

1. Create a stable sound field using a sound source
2. Start a sound measurement instrument, such as a sound level meter
3. Switch off the sound source and allow the sound to decay
4. Wait for the background sound to stabilise and stop the measurement (avoiding creating any noise that may disturb the measurement data)

The calculation of the T_{20} and T_{30} times starts after the sound has decayed by 5dB and ends after the level has dropped by 20dB and 30dB respectively. The measured data must have at least 10dB headroom above the noise floor.

The image below shows an example using an interrupted sound source. This is shown using one frequency only whereas a real measurement would be assessing the decay using typically, 1:3 octave

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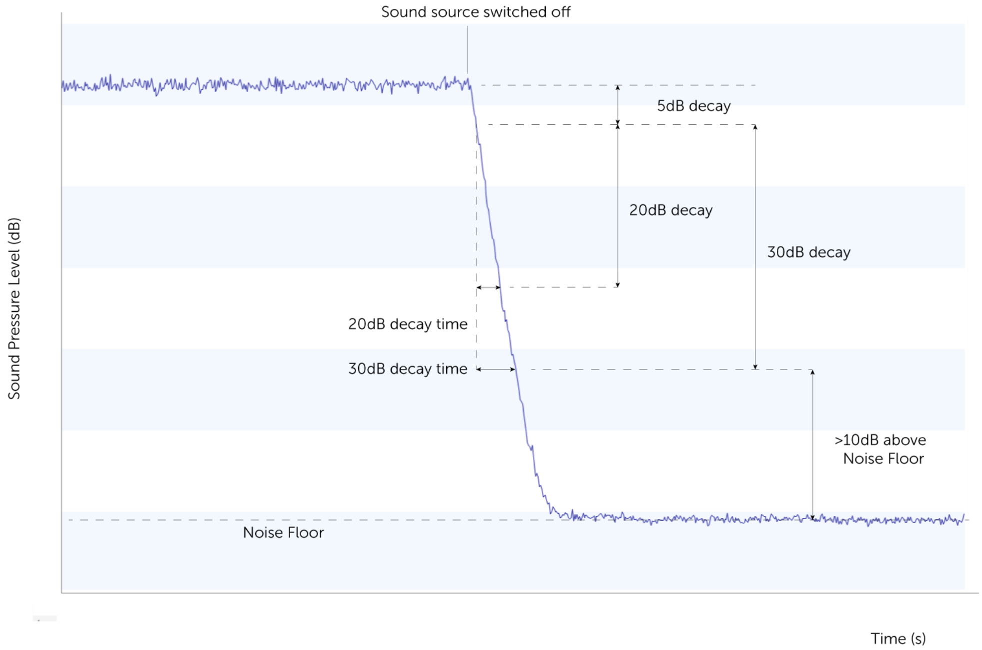
bands or 1:1 octave bands depending upon the application and the requirements of any applicable measurement standards.



The 20dB and 30dB decay times are calculated and the 60dB decay time calculated by multiplying these by 3 and 2 respectively.

Let's try this in MATLAB to measure the reverberation time of a small drum room. Our interrupted source is a 2-second Gaussian noise.

```
fs=44100; %sampling rate
t=(0:1/fs:2)'; %2 second signal
x=randn(1,length(t))'; %generate nise
x=x./max(x); %ormalize the signal
soundsc(x,fs)

h=wavread('Small Drum Room.wav'); %load room impulse response

pause %pause the script

y=conv(x,h); %convolve input with room impulse response

L = 882; %segment size in samples
step=L; %hop size of segments in samples
N=length(y); K=fix((N-L+step)/step);
time=(1:L)';
P=zeros(1,K);
for k=1:K
   y_rms=rms(y(time)); %get rms value of the windowed signal
   P(k)=20*log10(y_rms); %get the power level
   time=time+step;
end
t=linspace(0,L/fs*K,K); %assign x-axis values
plot(t,P);
```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```
xlabel('Time (sec)');
ylabel('Power (dB)');
```

Using the Data Cursor function in the MATLAB figure window, determine T_{20} and T_{30}.

Repeat the measurements for the St Nicolaes church. Superimpose the new reverberation plot in red color by adding this script:

```
hold on %superimpose figures in the same window
plot(t,P,'r'); %where P is the power level of the new room
```

Put the resulting plot in your documentation. Compare the new computed values with the previous room.

To understand more the behavior of the room, we can use sinusoids of varying frequencies as sources instead of noise. To generate a 2-second 250 Hz sinusoid:

```
x=cos(2*pi*250*t);
```

Using the small drum room, plot and superimpose the reverberation profiles using the following frequencies: 250 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, 8 kHz. Put this in your documentation. Compare the resulting plots. What does this infer about the absorption capabilities of a room?

Given that you have a sense of the reverberation times of the two rooms, verify the effects of reverberation time on a speech signal, singing voice, and instrumental music. In terms of intelligibility, and richness or fullness of sound, recommend which room is more appropriate for the three types of audio signals.

Together with your plots and answers to the questions, submit the Matlab script yoursurname_reverb.m. Compress your files as a zip file then upload in UVLE. **You are on your honor that your submission is your own work.**

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