EE125 Matlab Project 4 Part 1: Circular Convolution

Exploring circular convolution by looking at how acousticians design concert halls

```
% Reading the wav file for the data
[x,Fs]=audioread('onscreen.wav'); % Time series, sample frequency

% Load and listen to the data
% soundsc(x,Fs) % Guy says on screen

% Plot the data
tx = 0:1/Fs:(length(x)-1)/Fs;
figure(1)
subplot(2,1,1)
plot(tx,x)
ylabel('Signal')
xlabel('Time (sec)')
title('Original Signal')

% Load the small church impulse response, sample rate matches
load('smallChurch_fs22k.mat')
h = hChurch;

% Plot church's impulse response
th = 0:1/Fs:(length(h)-1)/Fs;
subplot(2,1,2)
plot(th,h)
ylabel('Signal')
xlabel('Time (sec)')
title('Church Impulse Response')

suptitle('Plots for steps 1 and 2')

% Use linear convolution with the impulse response to see what the
 signal
% would sound like in a small church
s = conv(h,x);
% soundsc(s,Fs)
```

This signal sounds much "deeper" or "fuller" - like there is some sort of echo, which makes sense when looking at what the impulse response of the church is, the convolution will have some sort of decreasing repetition of copies overlaid of the original signal giving it an echo effect

In order to have FFT based circular convolution equal to linear convolution, you just have to pad the signals being convolved to be the same length and linear conv. So for this case, the math is shown below:

```
L = length(x);
M = length(h);
N = M + L - 1;
xpad = [x; zeros(M-1,1)];
hpad = [h; zeros(L-1,1)];
```

```
% With linear convolution the length of the output is going to be the
 sum
% of the length of the inputs minus the overlap, which in this case is
 1
xp = [x; zeros(M-L,1)];
cc = ifft(fft(xp).*fft(h));
ccp = ifft(fft(xpad).*fft(hpad));

% Checking that the padded circular conv is the same as the linear
 conv
 MSE_Padded_Circ_Conv_Lin_Conv = immse(ccp,s) % ~0

% Listening to the two outputs
% soundsc(cc,Fs)
% soundsc(ccp,Fs)
% Maybe I did this wrong, but I can't really hear much of a difference
% between the two signals. I know that there is a difference because
 there
% will be aliasing when using, I guess after listening to it several
 times
% there is some sort of cutoff at the end of the circular convolution
% without padding, but it could just be a sort of placebo that I'm
 hearing

% Plot the required stuff for step 5
figure(2)

% The original signal
subplot(5,1,1)
plot(tx,x)
ylabel('Signal')
xlabel('Time (sec)')
title('Original Signal')

% The linearly convolved signal
ts = 0:1/Fs:(length(s)-1)/Fs;
subplot(5,1,2)
plot(ts,s)
ylabel('Signal')
xlabel('Time (sec)')
title('Linearly Convolved Signal')
axis([0 2.5 -10 10])

% The difference between linear conv and fft based padded conv
diff = s - ccp;
subplot(5,1,3)
plot(ts,diff)
ylabel('Difference')
xlabel('Time (sec)')
title('Difference Between Linear Conv and FFT Based Padded Conv')
axis([0 2.5 -10 10])

% FFT based convolution without padding
```

```
tcc = 0:1/Fs:(length(cc)-1)/Fs;
subplot(5,1,4)
plot(tcc,cc)
ylabel('Signal')
xlabel('Time (sec)')
title('Circular Conv Without Padding')

% The difference between linear conv and unpadded circ conv
diff2 = s(1:length(cc)) - cc;
subplot(5,1,5)
plot(tcc,diff2)
ylabel('Signal')
xlabel('Time (sec)')
title('Difference Between Linear Conv and Unpadded Circular Conv')
```

Based off of the graphs, the time domain aliasing is going to occur mostly in the beginning of the signal, since thats where the difference between the circular unpadded and linear convolutions are the greatest. This makes sense because the aliasing occurs because of the periodic nature of the tranformations so the signal should "leak" over into the left side of the time domain, but I was expecting the aliasing to be more pronounced than it was.

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EE 125 Matlab 4 Part 2: Signal Compression

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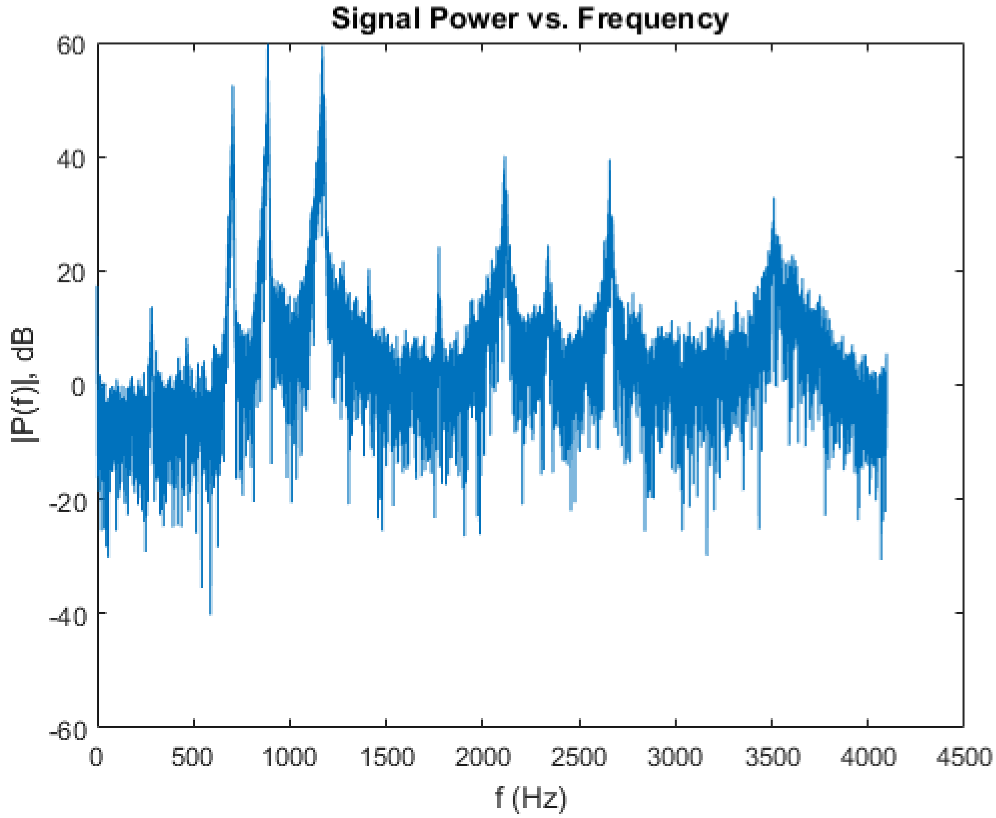
FFT Compression

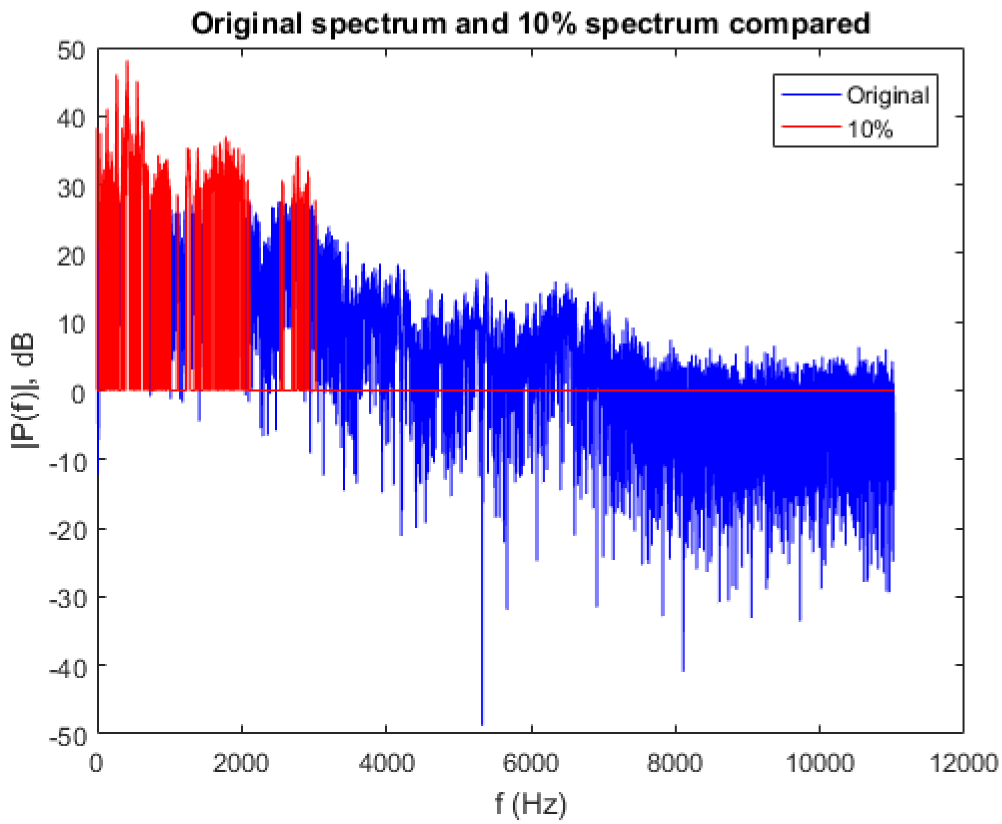
```
% Step 1
% Loading the train file
load train

% Listening to the train file
% soundsc(y,Fs)

% Step 2
% Take the N pt FFT using the next highest power of 2 from length(y)
 for N
n = 2^nextpow2(length(y));
f = Fs*(0:(n/2))/n;
Y = fft(y,n);

% Convert to decibels
P = mag2db(abs(Y));

figure(3)
plot(f,P(1:n/2+1))
xlabel('f (Hz)')
ylabel('|P(f)|, dB')
title('Signal Power vs. Frequency')

% The signal has its most powerful components at 750, 900, 1100, 2100,
% 2700, and 3500 Hz. If you look at the euler decomposition of the
 Fourier
% transform this means the signal is primarily composed of cosines of
 those
% frequencies

% Step 3
% Print the FFTcompression function
type('FFTcompression.m')

% Step 4
type('SNRoverall.m')
```

```
% Step 5
% Compress using FFT
Y100 = FFTcompression(y,100);
Y50 = FFTcompression(y,50);
Y20 = FFTcompression(y,20);
Y10 = FFTcompression(y,10);

% Reconstruct in time domain
recon100 = ifft(Y100);
recon50 = ifft(Y50);
recon20 = ifft(Y20);
recon10 = ifft(Y10);

% Step 5 a
% Listen to the different sounds
% soundsc(recon100,Fs) % Sounds like the normal signal
% soundsc(recon50,Fs) % Sounds pretty good, but not quite the same
% soundsc(recon20,Fs) % Not terribly different from 50% but not
 amazing
% soundsc(recon10,Fs) % Not great quality but definitely still sounds
 like
% a train.
% I did some testing in the command window to try to find where I
 could
% stop recognizing the signal and I found that to be at about 0.25
 percent
% retained, which I found really surprising (I was expecting to not be
 able
% to remove such a high percentage of the original signal), I could
 even
% kind of recognize it at 0.1%.

% Step 5 b
% Plot the original magnitude spectrum and the 10% spectrum together
P10 = mag2db(abs(Y10));
P10(P10<-10^6) = 0;
figure(4)
plot(f,P(1:n/2+1),'b',f,P10(1:n/2+1),'r')
xlabel('f (Hz)')
ylabel('|P(f)|, dB')
title('Original spectrum and 10% spectrum compared')
legend('Original','10%')
% The unadulterated signal is much more organic because its composed
 of a
% much larger variety of sinusoids where the 10% sounds fake because
 it's
% made up of less harmonics.

% Step 5 c
yp = ifft(Y); % Just doing this to preserve the signal length
snr100 = SNRoverall(yp,recon100);
snr50 = SNRoverall(yp,recon50);
snr20 = SNRoverall(yp,recon20);
snr10 = SNRoverall(yp,recon10);
```

```
T = table(snr100,snr50,snr20,snr10)

% Step 6
clear
[y,Fs] = audioread('onscreen.wav');

% Compress using FFT
Y100 = FFTcompression(y,100);
Y50 = FFTcompression(y,50);
Y20 = FFTcompression(y,20);
Y10 = FFTcompression(y,10);

% Reconstruct in time domain
recon100 = ifft(Y100);
recon50 = ifft(Y50);
recon20 = ifft(Y20);
recon10 = ifft(Y10);

% Listening to the outputs
% soundsc(recon100,Fs) % Sounds like the normal signal
% soundsc(recon50,Fs) % Sounds pretty good, guys voice sounds
 shallower
% soundsc(recon20,Fs) % The guy sounds like he's speaking through an
% intercom
% soundsc(recon10,Fs) % The guy sounds even more fake, like a textbook
% example of a digital recording
% This compression works until just about 2 percent when you start to
% barely be able to understand what is being said

% Plot the original magnitude spectrum and the 10% spectrum together
n = 2^nextpow2(length(y));
f = Fs*(0:(n/2))/n;
Y = fft(y,n);
P = mag2db(abs(Y));
P10 = mag2db(abs(Y10));
P10(P10<-10^6) = 0;
figure(4)
plot(f,P(1:n/2+1),'b',f,P10(1:n/2+1),'r')
xlabel('f (Hz)')
ylabel('|P(f)|, dB')
title('Original spectrum and 10% spectrum compared')
legend('Original','10%')
% Looking at the graph it makes sense why you can still understand
 what he
% is saying even when retaining only 10 percent of the signal power.
 This
% is because you can see the frequency waveforms for the important
 things
% he says like "on screen" and the beep or whatever that is.

% Calculating the SNR values
yp = ifft(Y); % Just doing this to preserve the signal length
snr100 = SNRoverall(yp,recon100);
```

```
snr50 = SNRoverall(yp,recon50);
snr20 = SNRoverall(yp,recon20);
snr10 = SNRoverall(yp,recon10);

T = table(snr100,snr50,snr20,snr10)


function Yout = FFTcompression(signal,percentRetained)
% function Yout = FFTcompression(signal,percentRetained)
% This function takes in a signal and a percent of the signal to
 retain,
% computes the fourier transform and removes the undesired components,
 and
% returns the FFT of only the percent you want to keep.
% If the percent retained is 90, then the top 90% of the signal will
 be
% kept.

% Take the Fourier transform
n = 2^nextpow2(length(signal));
S = fft(signal,n);

% Zero out the weaker frequency components and return the signal
dummy = sort(abs(S));
index = round((1-percentRetained/100)*length(dummy))+1;
q = dummy(index);
S(abs(S)<=q) = 0;
Yout = S;

return

function snr = SNRoverall(signal,reconSignal)
% function snr = SNRoverall(signal,reconSignal)
% This function computes the overall SNR of a signal

snr = 10*log(sum(signal.^2./(signal-reconSignal).^2));

return

T =

    snr100    snr50     snr20    snr10
    ______    ______    _____    ______

    448.61    289.98    300.5    274.96


T =

    snr100    snr50     snr20     snr10
    ______    ______    ______    _____

    477.95    239.07    231.94    271.2
```


Signal Power vs. Frequency
|P(f)|, dB
f (Hz)
60
40
20
0
-20
-40
-60
0
500
1000
1500
2000
2500
3000
3500
4000
4500


Original spectrum and 10% spectrum compared
Original
10%
|P(f)|, dB
f (Hz)
50
40
30
20
10
0
-10
-20
-30
-40
-50
0
2000
4000
6000
8000
10000
12000

Storage savings due to compression

Step 1 It makes sense that the Fourier transform is doubling storage space of the signal because the FFT returns the information in both the positive and negative frequencies. If I were going to save them, since I know the original signal will always be real valued, I would save only half of the Fourier transform because the negative frequencies are going to be symmetric to the positive frequencies about the y axis. If you look at the length adjusted yp, then the amount of data is going to be the same whether it is stored in the Fourier or time domain (if you take only one half of Fourier data). This would equate to the same amount of memory usage storing data in the time or frequency domain assuming you make the signal to the next power of 2

```
whos y
whos yp
whos Y
whos Y100
% whos Y50
% whos Y20
whos Y10

% Step 2
Ysp = sparse(Y10);
whos Ysp % Oh wow, Y10 is 524288 bytes where Ysp is 78616 bytes.
% The space saving here is pretty substantial, but not 10x. This is
% probably because the array still needs to save the corresponding
% positions for each of the nonzero values from the previous matrix.
 Instead
% of just storing all of the values and keeping the zeros in the
 places
% they were in before you have to store all the position values of the
% nonzero values.
```

```
  Name          Size             Bytes  Class     Attributes

  y         25680x1             205440  double

  Name          Size             Bytes  Class     Attributes

  yp        32768x1             262144  double

  Name          Size             Bytes  Class     Attributes

  Y         32768x1             524288  double    complex

  Name          Size             Bytes  Class     Attributes

  Y100      32768x1             524288  double    complex

  Name          Size             Bytes  Class     Attributes

  Y10       32768x1             524288  double    complex

  Name          Size            Bytes  Class     Attributes
```

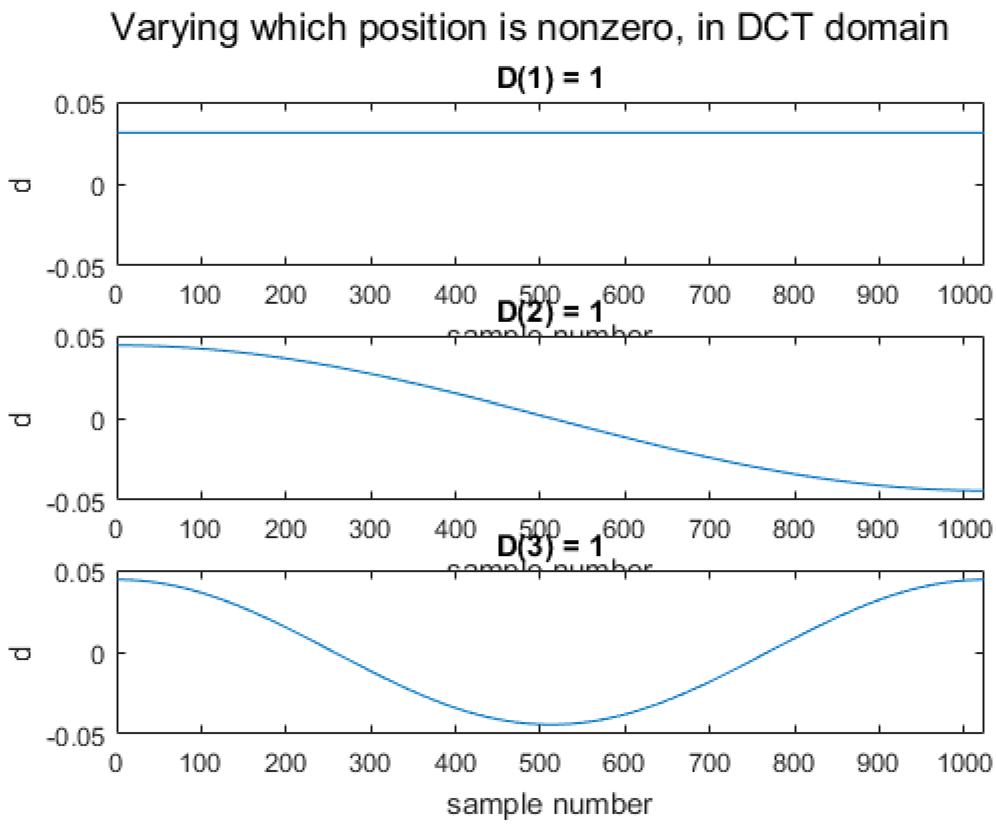
```
  Ysp       32768x1            78616  double    sparse, complex
```

Discrete Cosine Transform

```
% Step 1
D = zeros(1024,1);

D(1) = 1;
d = idct(D);
figure(5)
subplot 311
plot(d)
axis([0 1024 -0.05 0.05])
xlabel('sample number')
ylabel('d')
title('D(1) = 1')

D(1) = 0;
D(2) = 1;
d = idct(D);
subplot 312
plot(d)
axis([0 1024 -0.05 0.05])
xlabel('sample number')
ylabel('d')
title('D(2) = 1')

D(2) = 0;
D(3) = 1;
d = idct(D);
subplot 313
plot(d)
axis([0 1024 -0.05 0.05])
xlabel('sample number')
ylabel('d')
title('D(3) = 1')

suptitle('Varying which position is nonzero, in DCT domain')

% What this step is doing is varying which composite harmonics are
 being
% represented in the signal. The discrete cosine transformation
 represents
% a time domain signal as a composite superposition of a bunch of
 different
% cosine functions. When you set D(1) = 1 you're setting the magnitude
 for
% what is the DC component of the signal. When you set D(2) = 1 you're
% looking at only the first harmonic (lowest nonzero frequency), and
 when
% you set D(3) = 1 you're looking at the next harmonic. The results
 make
```

```
% sense since the first plot is just a DC shift, the second plot is a
% cosine, and the third plot is a higher frequency cosine.

% Step 2
type('DCTcompression.m')

D100 = DCTcompression(y,100);
D50 = DCTcompression(y,50);
D20 = DCTcompression(y,20);
D10 = DCTcompression(y,10);

dctRecon100 = idct(D100);
dctRecon50 = idct(D50);
dctRecon20 = idct(D20);
dctRecon10 = idct(D10);

dctSnr100 = SNRoverall(yp,dctRecon100);
dctSnr50 = SNRoverall(yp,dctRecon50);
dctSnr20 = SNRoverall(yp,dctRecon20);
dctSnr10 = SNRoverall(yp,dctRecon10);

% DCT SNR table
dctT = table(dctSnr100,dctSnr50,dctSnr20,dctSnr10)

% Listening to the differences between FFT, DCT compressions, 20%
 retained
% soundsc(recon20,Fs)
% soundsc(dctRecon20,Fs)
% Oh wow, I actually found there to be a kind of significant
 difference in
% the amount of noise you hear in the reconstructed signal. The DCT
 was
% much cleaner.


function Dout = DCTcompression(signal,percentRetained)
% function Dout = DCTcompression(signal,percentRetained)
% This function takes in a time domain signal, performs a DCT,
 retaining
% only a desired percentage of the signal strength.

% Take the n point DCT of the function (making sure it's an even power
 of
% 2)
n = 2^nextpow2(length(signal));
D = dct(signal,n);

% Find where you want to start zeroing out data
dummy = sort(abs(D));
index = round((1-percentRetained/100)*length(dummy)) + 1;

% Zero out the desired data points
Dout = D;
Dout(abs(Dout)<dummy(index)) = 0;
```

```
return

dctT =

    dctSnr100    dctSnr50    dctSnr20    dctSnr10
    _________    ________    ________    ________

    NaN          298.61      278.77      210.38
```



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