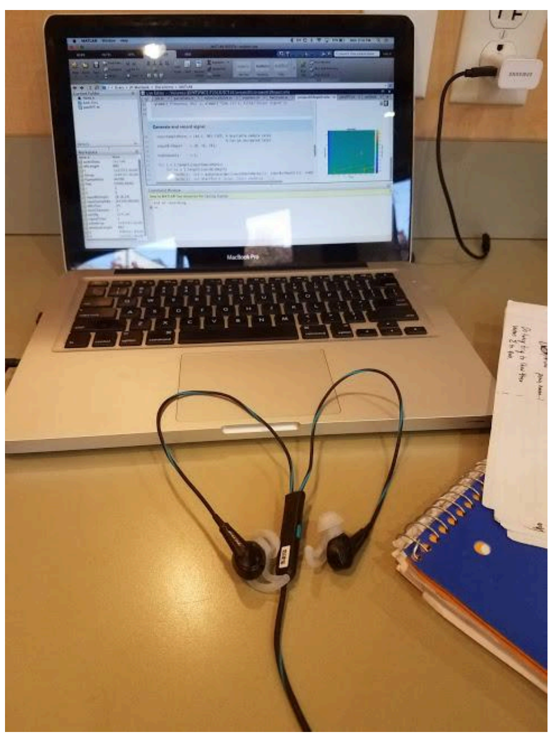
```
%%%%%%%%%%%%%%%%%%%%%%%%%%
%%%  EC516 Project 01  %%%
%%% Jeremiah Sullivan  %%%
%%%    ML, IC, or RL   %%%
%%%   November 2018    %%%
%%%%%%%%%%%%%%%%%%%%%%%%%%
%%% Project Requirements
% (a) Record and display a speech signal
% (b) (3 points) Compute and display the discrete TDFT of the
%            rectangular (“box”) window, duration ~20 ms.


% (c) (4 points) Use GFBS Method to compute signal back from discrete TDFT
clc; clear all; close all;
```

A: Generate and display signal

Choosing to use 2013 13" Macbook Pro // Bose Headphones as a recording platform of convenience. Sticking to default sampling rates, can later decimate if desired. Standard 3.5 mm audio connector, so assuming the Mac handles all active conditioning. Not using the "noise-cancelling" mode on the headphone, because I bet this geometry would cause... unique... problems with the algorithms. May be interesting to try later though.

Initial trials were completed in a noisy coffeeshop (update this if location changes).



For replicability, let's also generate some sound. Let's try an LFM chirp. We now have further unkowns (internal afg / transmit chain).

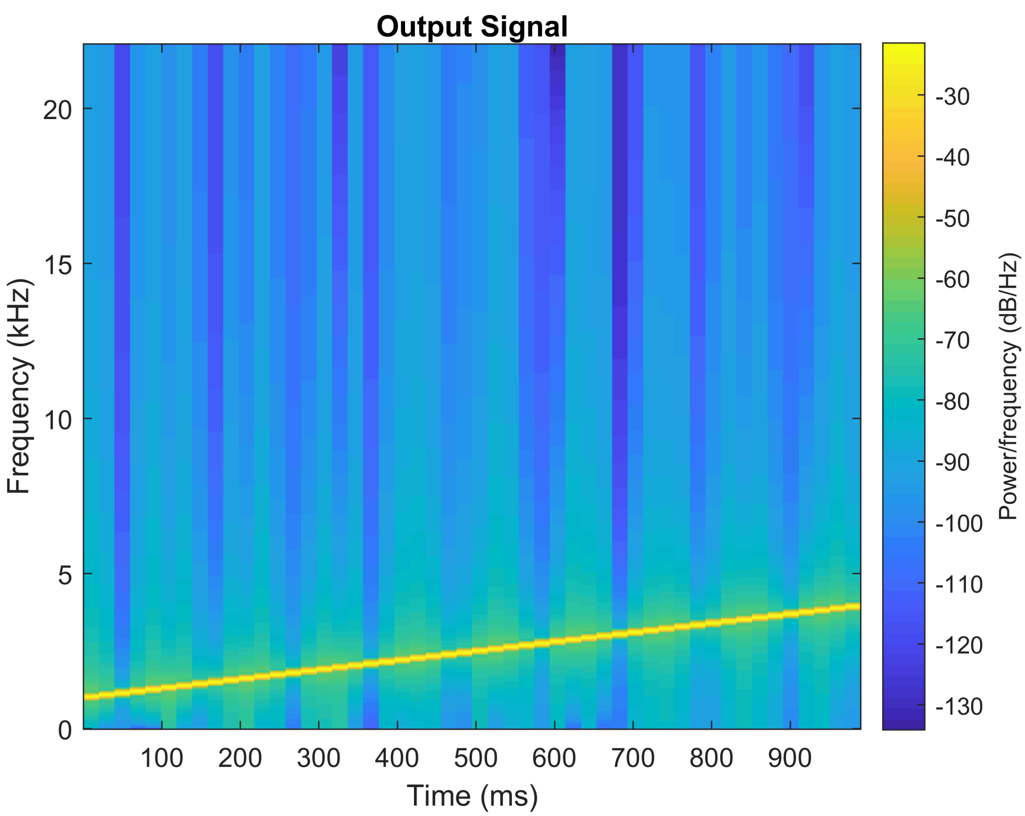
Max audio-settings, 24-bit, 96 kHz... This seems unnecessary.

```
% Set up signal
fSampleOut = 44.1E3;
signalTime = 1; % second
fVec       = [1000, 4000]; % from 100 Hz to 4 kHz
timeArray  = 0:(1/fSampleOut):signalTime; % Time (seconds)
fArray     = linspace(fVec(1), fVec(2), length(timeArray) );
nBitsOut   = 24;
y          = chirp(timeArray, fVec(1), timeArray(end), fVec(end));
```

Verifying output signal

```
% Y = fftshift(abs(fft(y)));
% Y = Y(round(length(Y)/2):end); % Round signal if odd length
% f = linspace(0, fSampleOut/2, length(Y));
% plot(f, Y); xlabel('Frequency (Hz)'); ylabel('|Y|'); title('Single Sided Spectrum');

windowLength = 20E-3.*fSampleOut;
dftLength = windowLength;
spectrogram(y, windowLength, round(0.01*windowLength), dftLength, fSampleOut, 'yaxis')
%ylabel('Frequency (Hz)'); xlabel('Time (s)');
title('Output Signal');
```



Generate and record signal

```
inputSampleRates = [48 44.1, 96].*1E3; % Available sample rates
% Can be decimated later
inputBitDepth    = [24, 16, 24];

numChannels      = 1;

for i = 1:length(inputSampleRates)
    for ii = 1:length(inputBitDepth)
        recObj{i, ii} = audiorecorder(inputSampleRates(i), inputBitDepth(ii), numChannels);
        recObj{i, ii}.StartFcn = 'disp(''Start speaking.'')';
        recObj{i, ii}.StopFcn = 'disp(''End of recording.'')';
    end
end

record(recObj{1,1}, 3);
```

```
Start speaking.
```

```
tic;
while toc < 2
    sound(y, fSampleOut); pause(signalTime.*0.9);
end
audioData{1, 1} = getaudiodata(recObj{1,1});
```

Signal Processing on Recorded signal

```
y = audioData{1,1};

windowLength = 20E-3.*inputSampleRates(1);
unitWindow   = ones(windowLength, 1);
dftLength    = windowLength;
```

The following spectrograms compare the overlap (L) and window properties

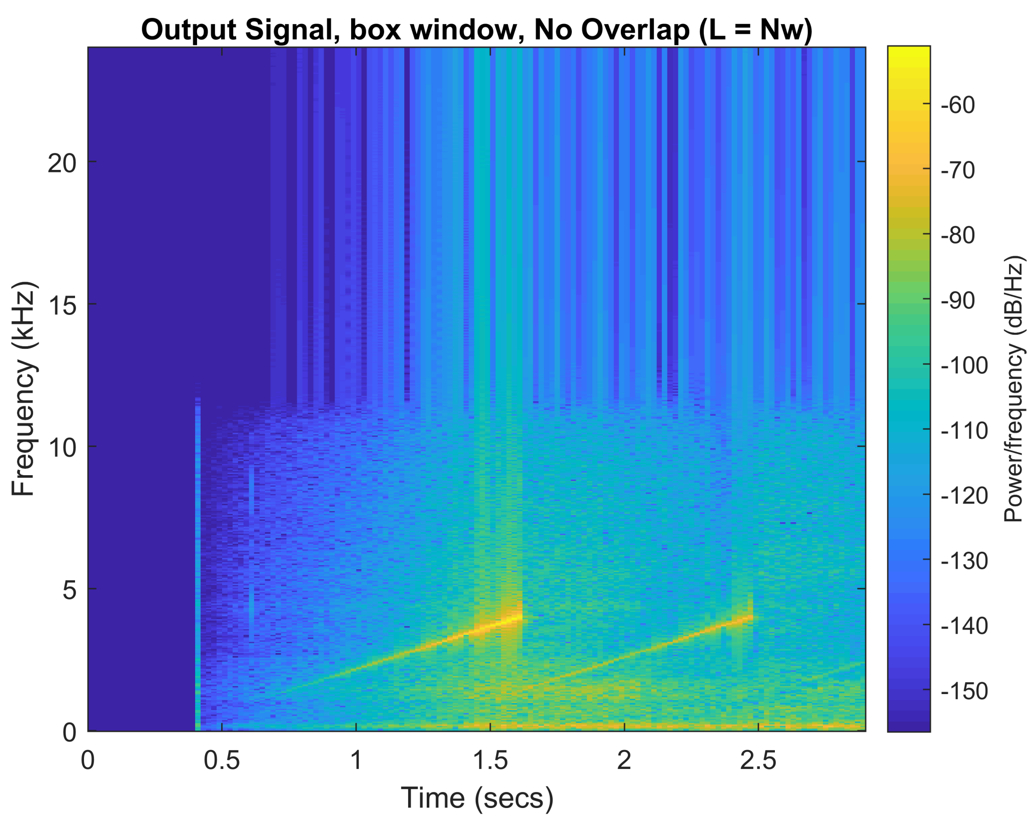
- Hamming vs unit-step
- Full overlap (L=1) vs. no overlap

```
% figure;
% [fbs.ham.s, fbs.ham.w.fbs, fbs.ham.t] =  spectrogram(y, windowLength, windowLength-1, dftLeng
% spectrogram(y, windowLength, windowLength-1, dftLength, inputSampleRates(1), 'yaxis'); % For
% %ylabel('Frequency (Hz)'); xlabel('Time (s)');
% title('Output Signal, hamming, Full Overlap (L = 1)');
```

```
% figure;
% [gfbs.ham.s, gfbs.ham.w.fbs, gfbs.ham.t] =  spectrogram(y, windowLength, 0, dftLength, inputS
% spectrogram(y, windowLength, 0, dftLength, inputSampleRates(1), 'yaxis');
% %ylabel('Frequency (Hz)'); xlabel('Time (s)');
% title('Output Signal, hamming, No Overlap (L = Nw)');

[fbs.UnitStep.s, fbs.UnitStep.w.fbs, fbs.UnitStep.t] =  spectrogram(y, unitWindow, windowLength
    dftLength, inputSampleRates(1), ...
    'twosided', 'yaxis');
%spectrogram(y, unitWindow, windowLength-1, dftLength, inputSampleRates(1), 'yaxis');
%ylabel('Frequency (Hz)'); xlabel('Time (s)');
%title('Output Signal, Full Overlap (L = 1)');

figure;
[gfbs.UnitStep.s, gfbs.UnitStep.w.fbs, gfbs.UnitStep.t] =  spectrogram(y, unitWindow, 0, dftLen
    'twosided','yaxis');
spectrogram(y, unitWindow, 0, dftLength, inputSampleRates(1), 'yaxis');
%ylabel('Frequency (Hz)'); xlabel('Time (s)');
title('Output Signal, box window, No Overlap (L = Nw)');
```



We observe more frequency blurring with the unit step (sharp increases at the start of signal).

We observe a smoother signal with more overlap (computing fft for every point).

Implimenting GFBS Method

First, we must verify the GFBS conditions are satisfied.

L is defined by window overlap...

Number of overlapped samples, specified as a positive integer.

- If `window` is scalar, then `noverlap` must be smaller than `window`.
- If `window` is a vector, then `noverlap` must be smaller than the length of `window`.

M is defined by the DFT length. We have purposefully set these parameters equal so that we have critical sampling in at least one case.

The critically sampled case / unit step makes it very simple to reconstruct using f[n] = 1/w[n] = w[n]

```
j = sqrt(-1);
[M, L]  = size(gfbs.UnitStep.s);
Nw      = M; % Critically sampled

exponentialPart = @(k) (exp(j.*(2.*pi.*(k-1)./M) .* (0:(Nw-1)) ) );

temp    = zeros(1,M);
tempNew = zeros(1,M);
slow    = zeros(1, length(y));
tic;
for ll = 1:floor(length(y)/windowLength)
    indices = ((ll-1)*windowLength+1):((ll)*windowLength) ;
    temp    = zeros(1,M);     % Things breaks when I remove these
    tempNew = zeros(1,M);     % Things breaks when I remove these
    for k = 1:(M)             % Matlab arrays...
        tempNew = (gfbs.UnitStep.s(k,ll).* exponentialPart(k) );
        temp    = temp+tempNew;
        tempNew = zeros(1,M); % Things breaks when I remove these

    end
    slow(indices) = temp;
end
results.slow.time = toc;
%------------------------------------------------------------------------------%
ifftOut = zeros(1,M); tic;
for ll = 1:floor(length(y)/windowLength)
    indices = ((ll-1)*windowLength+1):((ll)*windowLength) ;
    temp = ifft(gfbs.UnitStep.s(:,ll), [], 1, 'symmetric');
    ifftOut(indices) = temp;
end
results.fast.time = toc;

result.fast.y = ifftOut; % (1./M)
result.slow.y = (1./M).* slow;
result.Fs   = inputSampleRates(1);
fprintf('Using fft takes: %01.1f milliseconds', results.fast.time*1E3)
```

```
Using fft takes: 113.3 milliseconds
```

```
fprintf('Using explicit method takes: %01.1f seconds', results.slow.time)
```

```
Using explicit method takes: 19.6 seconds
```

```
tempSum = zeros(1, size(fbs.UnitStep.s, 2));
% Also try the FBS method, for kicks.
% ... Pull data from one of the first spectrograms (L=1)
```

Analyze Results

```
disp('Test for numerical errors, max of real, imag');
```

```
Test for numerical errors, max of real, imag
```

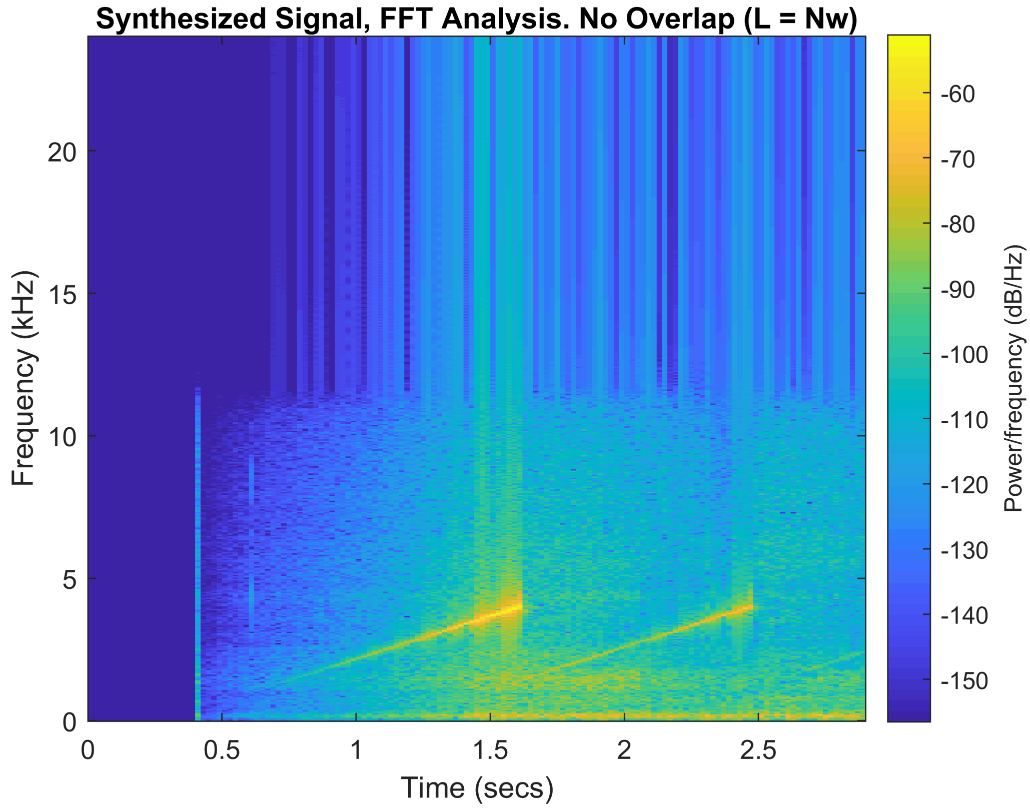
```
imagRoundOff = log10([rms(real(result.fast.y)), rms(real(result.slow.y))]./ ...
    [rms(imag(result.fast.y)), rms(imag(result.slow.y))]);
% This test seems kind of hand-wavy
fprintf('Real component is %01.1f, %01.1f (fft, explicit) orders of magnitude above imaginary'
         imagRoundOff(1), imagRoundOff(2));
```

```
Real component is Inf, 12.9 (fft, explicit) orders of magnitude above imaginary
```

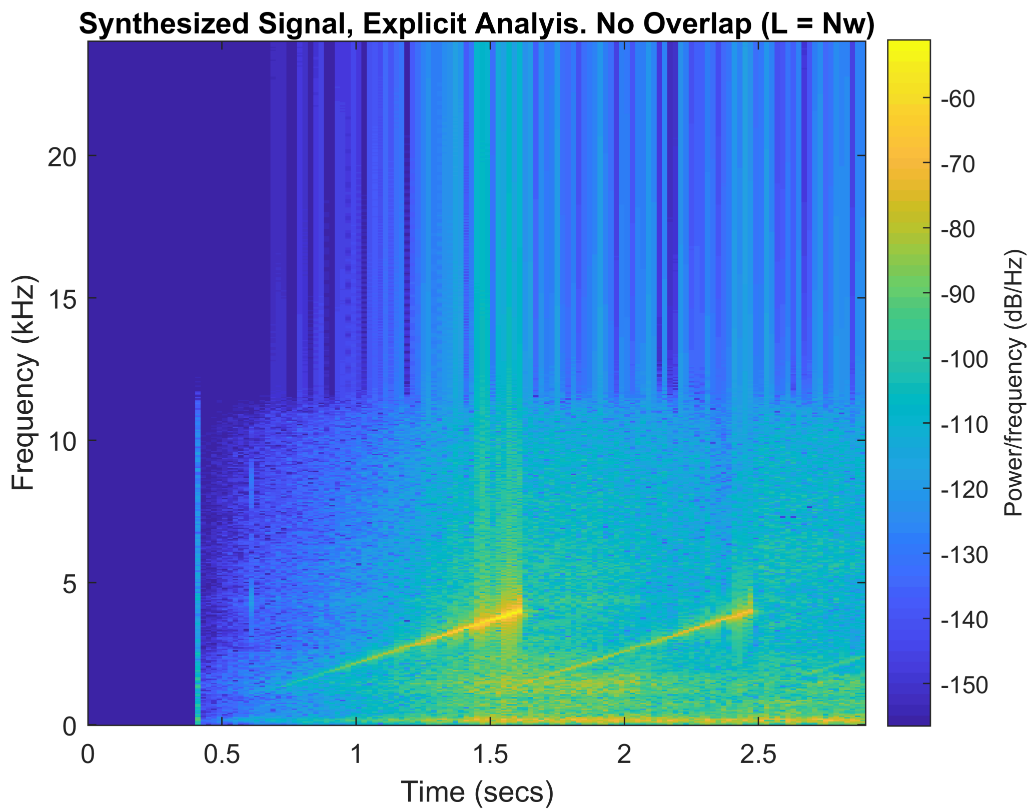
```
if imagRoundOff < 10
    disp('Signal may not be real (Synthesis algorithm is incorrect)');
else
    disp('Imaginary component appears to be simply round-off error, OK to ignore')
end
```

```
Imaginary component appears to be simply round-off error, OK to ignore
```

```
spectrogram(real(result.fast.y), unitWindow, 0, dftLength, inputSampleRates(1), 'onesided','ya
title('Synthesized Signal, FFT Analysis. No Overlap (L = Nw)');
```




```
spectrogram(real(result.slow.y), unitWindow, 0, dftLength, inputSampleRates(1), 'onesided','yax
title('Synthesized Signal, Explicit Analyis. No Overlap (L = Nw)');
```



Trust me, they sound really similar... Compare synthesized signal spectrogram w/ Box Window, no overlap

```
disp('Press a key to play FFT`d signal'); pause
```

```
Press a key to play FFT`d signal
```

```
sound(real(result.fast.y), inputSampleRates(1));pause(2);
disp('Press a key to play explicit signal'); pause
```

```
Press a key to play explicit signal
```

```
sound(real(result.slow.y), inputSampleRates(1));pause(2);

disp('Press a key to play original signal'); pause
```

```
Press a key to play original signal
```

```
sound(y, inputSampleRates(1));pause(2);
```