MPEG LAYER II

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```
clear
clc
close all
```

Reading Audio File

```
file = 'project.wav';
[signal,Fs] = audioread(file);
file_info = audioinfo(file)
```

```
file_info = struct with fields:
             Filename: 'D:\FALL 2023\Projects\MPEG-Audio-Compression\project.wav'
    CompressionMethod: 'Uncompressed'
          NumChannels: 1
           SampleRate: 24000
         TotalSamples: 240000
             Duration: 10
                Title: []
              Comment: []
               Artist: []
        BitsPerSample: 16
```

```
bps = file_info.BitsPerSample;
sound(signal,Fs);
```

Filter Bank

PQMF

- Divides the spectrum into 32 subbands.
- Has 32 modulated coefficients, each of order 512.

Getting Filter Coefficients

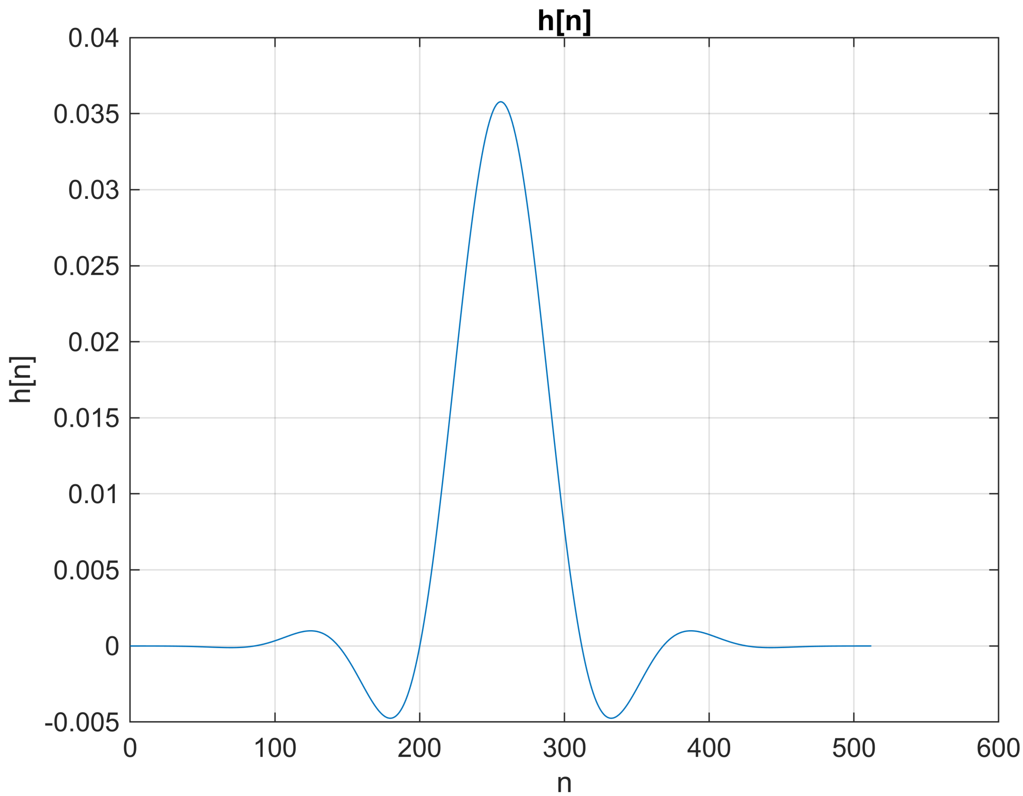
Here, we simply load the filter coefficients provided in the standard.

```
num_filters= 32;
filter_order=512;
h_all = zeros(num_filters,filter_order);  % A Matrix that will contain all coeffs
h_coeffs = PQMF_ReadCoeffs();             % Coefficients read from the standard
```

```
L = 257
```

Plotting Coefficients

```
plot([0:512],h_coeffs)
grid on
title('h[n]')
xlabel('n')
ylabel('h[n]')
```



Getting all Modulated Filters

Now, we perform the modulation to get the modulate coeffiecients for each subband, h_k[n].

According to the following equation:

$$h_k[n] = h[n]\cos\left[\left(k+\frac{1}{2}\right)(n-16)\frac{\pi}{32}\right]$$

$$g_k[n] = 32\ h[n]\cos\left[\left(k+\frac{1}{2}\right)(n+16)\frac{\pi}{32}\right]$$

$$k = 0,1,\ldots,31$$

$$n = 0,1,\ldots,511$$

(Bosi, Goldberg, sect 4.2)

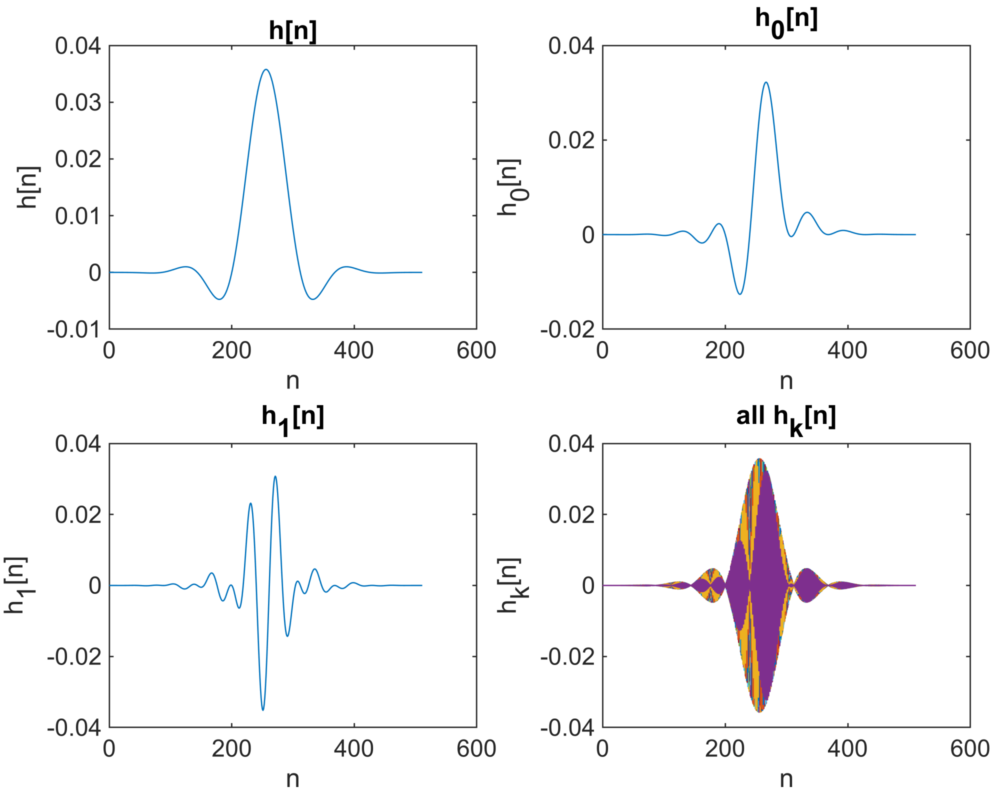
```
for k= 0 :num_filters-1
```

```
    for n=0: filter_order-1
        h_all(k+1,n+1) = h_coeffs(n+1) * cos((k+0.5)*(n-16)*(pi/32));
    end
end
```

Plotting all Coefficients

i.e. h[n], and the modulated ones, i.e. h_k[n]

```
subplot(2,2,1)
plot([0:511],h_coeffs(1:512))
title('h[n]')
xlabel('n')
ylabel('h[n]')
subplot(2,2,2)
plot([0:511],h_all(1,:))
title('h_0[n]')
xlabel('n')
ylabel('h_0[n]')
subplot(2,2,3)
plot([0:511],h_all(2,:))
title('h_1[n]')
xlabel('n')
ylabel('h_1[n]')
subplot(2,2,4)
plot([0:511],h_all)
title('all h_k[n]')
xlabel('n')
ylabel('h_k[n]')
```



Filtering the signal

Now, we use the filters obtained in the previous steps, to filter the audio signal itself.

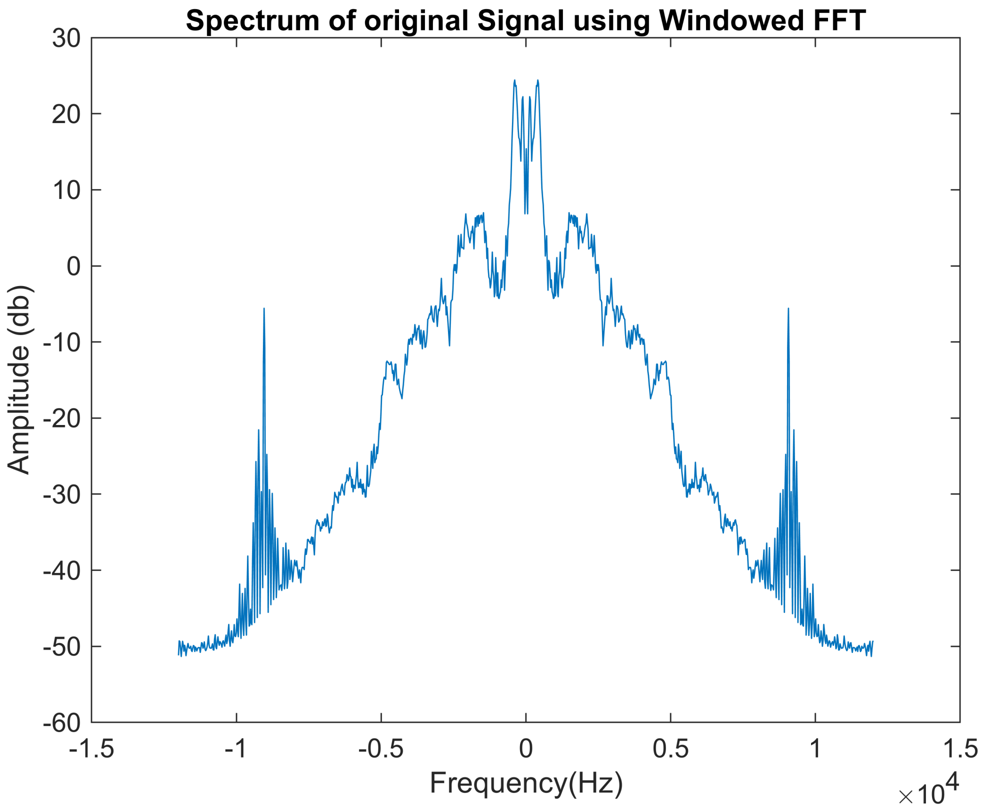
```
sig_filtered =[];
for k=0:1:31
    sig_f= filter(h_all(k+1,:),1,signal);  %Filtering
    sig_down = downsample(sig_f,32);        %Down Sampling
    sig_filtered=[sig_filtered sig_down];  %Appending
end
```

Windowed FFT

Computing Signal Frequency Spectrum

Applying FFT, to get the spectrum of the signal that will be used later.

```
frequncy_signal = windowed_FFT(signal, 1024);
figure()
freq = linspace(-Fs/2,Fs/2,length(frequncy_signal));
plot(freq, 20*log10(abs(frequncy_signal)));
title('Spectrum of original Signal using Windowed FFT')
xlabel("Frequency(Hz)");
ylabel("Amplitude (db)");
```



Psychoacoustic Model I

We apply this model to the audio signal, to analyse which frequency bands are to be removed, and performing the non-tonal masking, as we will see.

Computing SPL

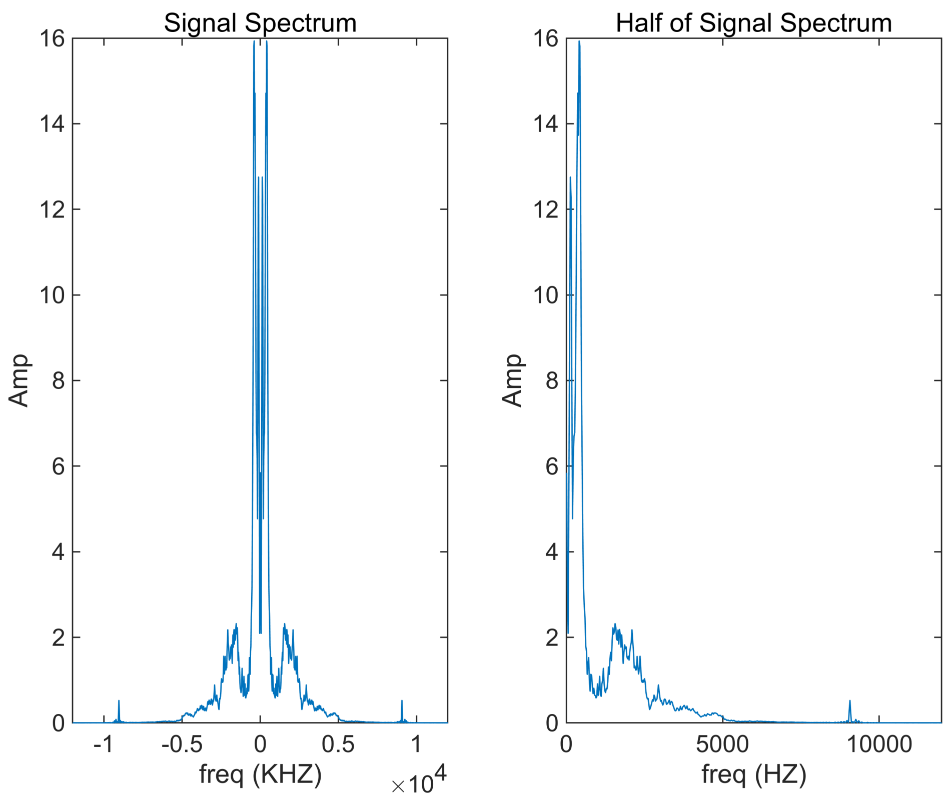
Signal Spectrum after FFT

First of all, we take FFT fir the signal.

Note: here we used the given spectrum as per the last tutorial instructions.

```
N = 1024;           % number of points
% load the Signal_Spectrum.mat
X = load('Signal_Spectrum.mat');
Signal_Spectrum = (X.Signal_Spectrum);
% set the freq vector for the signal
f_vector = linspace(-Fs/2,Fs/2,N);

% get the right half of the Spectrum
half_spectrum = X.Signal_Spectrum(513:end);
half_freq = f_vector(N/2+1:end);
```

```
% plot the Signal Spectrum
figure(1);
subplot(1,2,2);
% the whole signal spectrum
subplot(1,2,1);
plot(f_vector,Signal_Spectrum)
xlabel('freq (KHZ)')
ylabel('Amp')
subtitle('Signal Spectrum')
% half Signal Spectrum
subplot(1,2,2);
plot(half_freq,half_spectrum);
xlabel('freq (HZ)');
ylabel('Amp');
subtitle('Half of Signal Spectrum');
```



SPLs Computing and Evaluating

Now, let's find the SPLs corresponding to each frequency sample in the signal.

Note: Calculations are made for half of the spectrum.

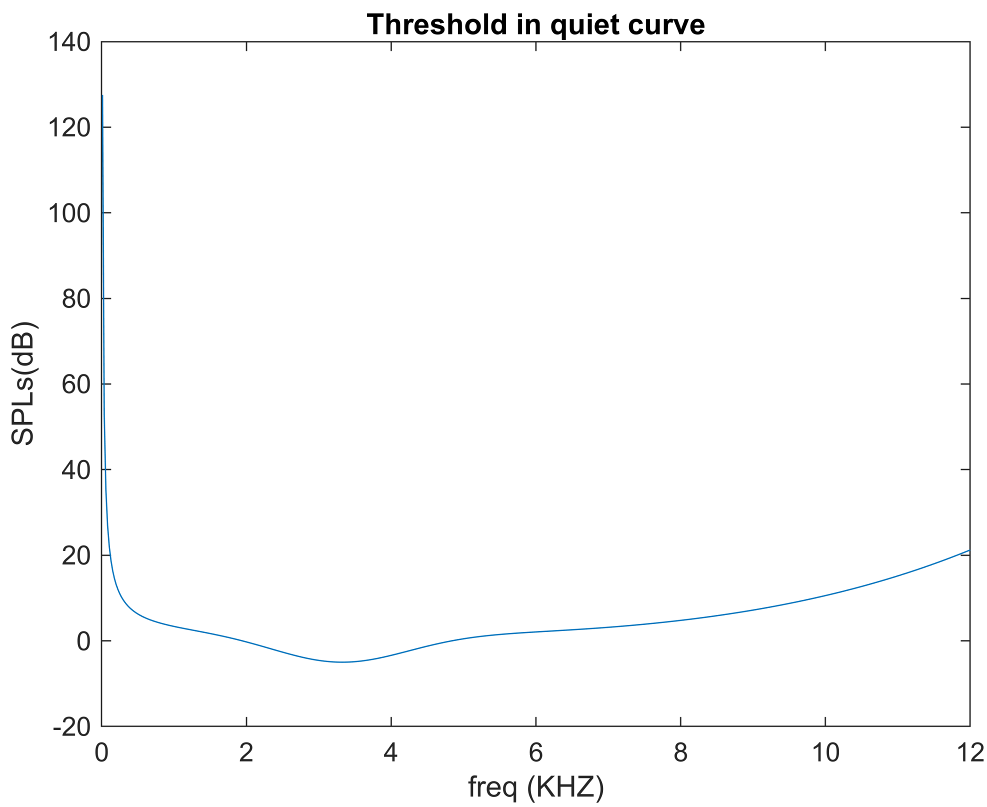
```
% Calculate the Lks for each value at half_spectrum
Lk = Lk_fn(N,half_spectrum);
```

$$L_k = 96\,\text{dB} + 10\log_{10}\left(4/N^2\,|X[k]|^2\,8/3\right) \quad \text{for } k=0,\ldots,N/2-1$$

```
% Calculate the threshold_in_quiet for each freq
q_threshold = threshold_in_quiet(half_freq);
```

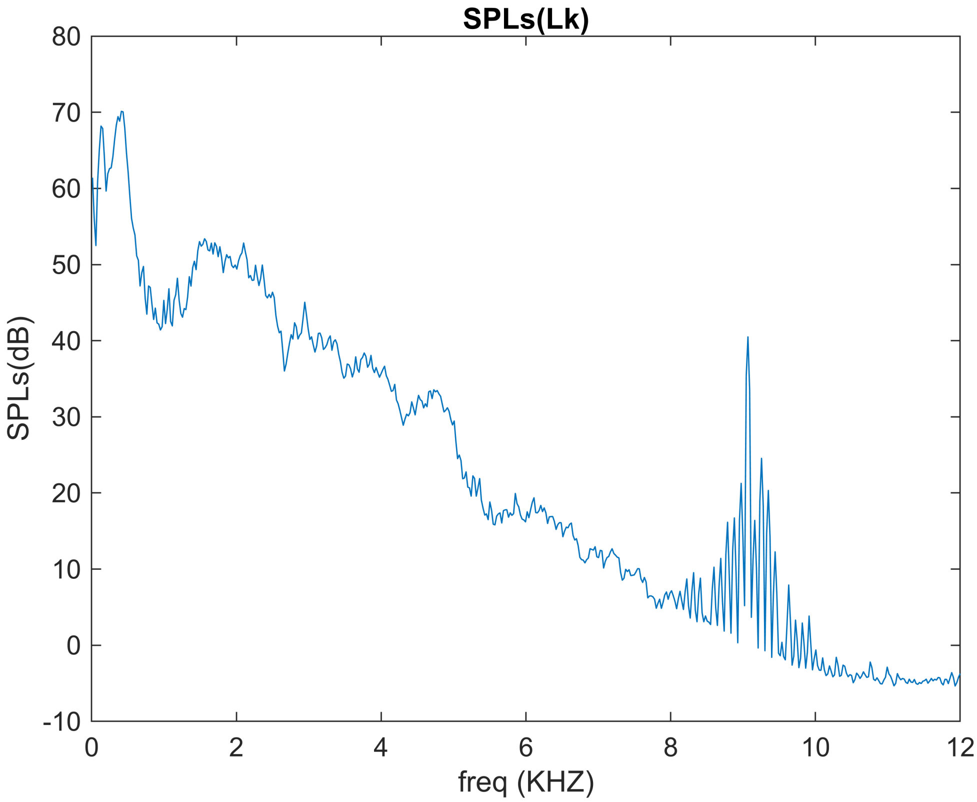
$$A(f)/\text{dB} = 3.64(f/\text{kHz})^{-0.8} - 6.5e^{-0.6(f/\text{kHz}-3.3)^2} + 10^{-3}(f/\text{kHz})^4$$

```
close all;
% plot threshold_in_quiet
plot(half_freq/1000,q_threshold);
title('Threshold in quiet curve')
xlabel('freq (KHZ)')
ylabel('SPLs(dB)')
```

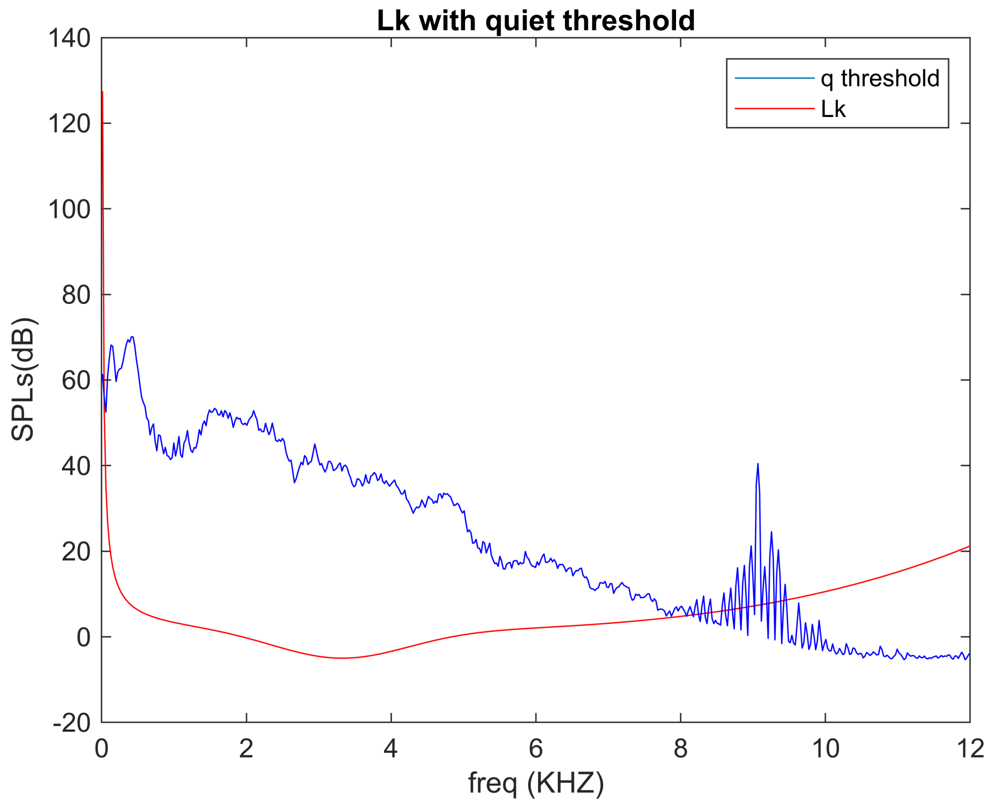



```
% plot Lks
plot(half_freq/1000,Lk);
title('SPLs(Lk)');
```

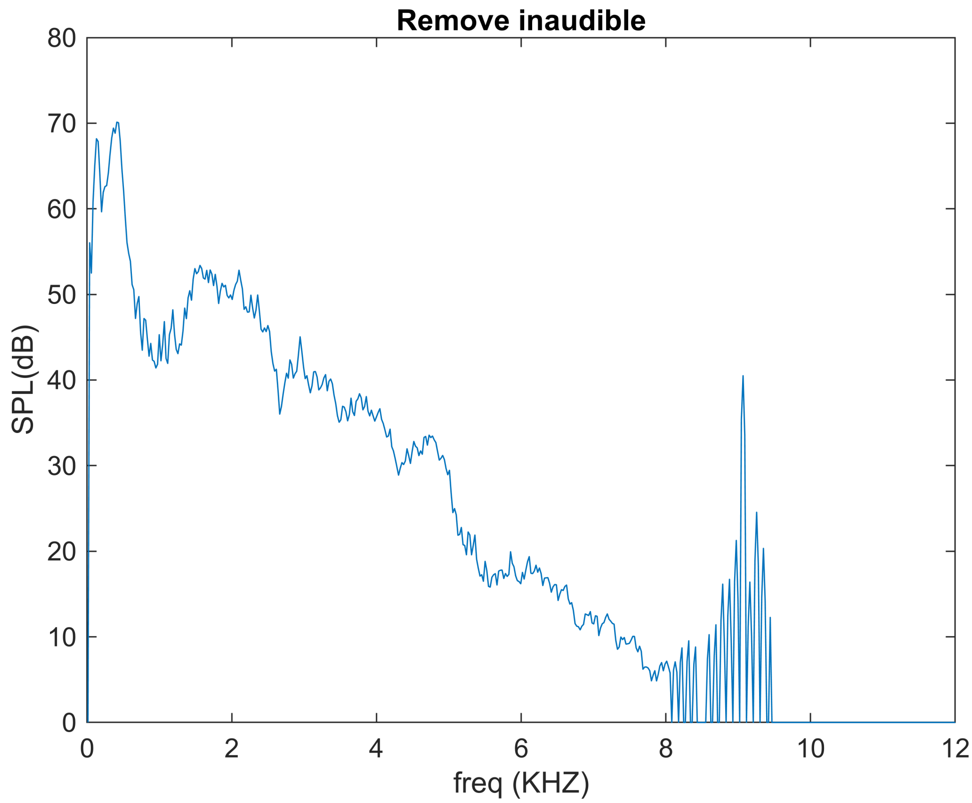
```
xlabel('freq (KHZ)');
ylabel('SPLs(dB)');
```




```
title('Lk with quiet threshold');
hold on
plot(half_freq/1000,q_threshold,Color='r');
plot(half_freq/1000,Lk,Color='b');
legend({'q threshold','Lk'});
hold off
```




```
% set zeros to Lks that is less than the quite threshold
indices = q_threshold>Lk;
Lk(indices) = 0;
% plot
plot(half_freq/1000,Lk);
title('Remove inaudible')
xlabel('freq (KHZ)')
ylabel('SPL(dB)')
```



Psychoacoustic Model II

Separation of Tonal & Non-Tonal Components

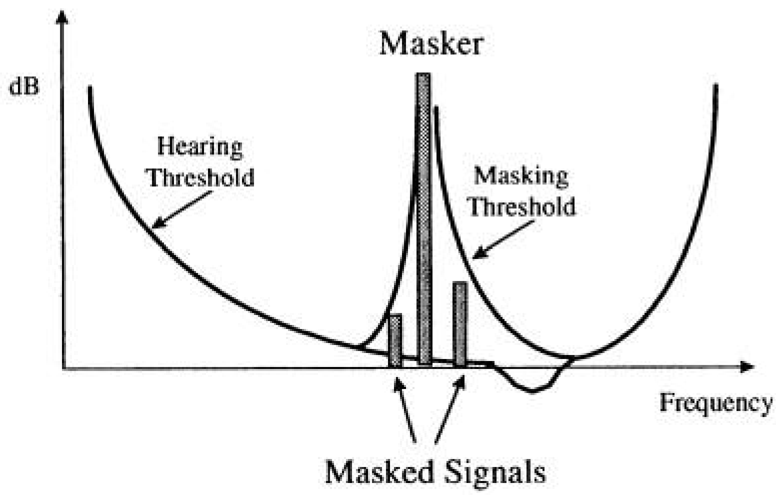
In this part:

- Find maskers, that are the maxima of each subband.
- Find the right most freq affected by this masker.
- Interpolate a curve between them, that is the threshold.
- Any value below this threshold is to be masked.

Divide the Lk specturm into 32 sub-bands

```
% convert the SPLs and the freq into sub-bands
sub_num = 32;
SPL_matrix = vector2matrix(Lk,sub_num);
freq_matrix = vector2matrix(half_freq,sub_num);
```

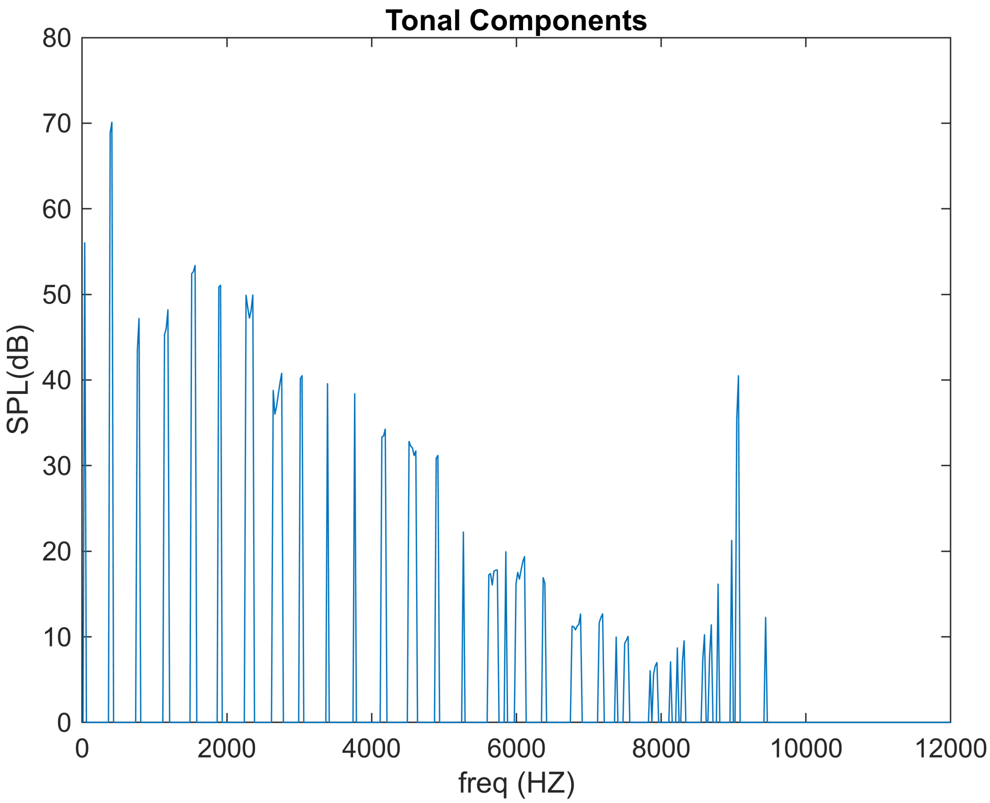
Get the masker Masker frequencies



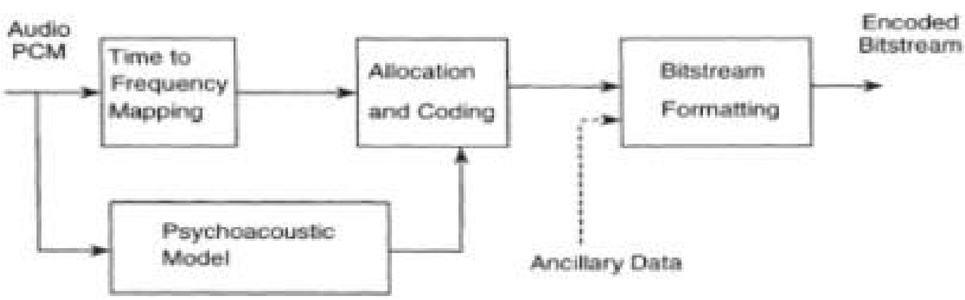

```
% Looping over the sub-bands
for i = 1 : sub_num

    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%% Get the golobal and local maximas  %%%%%%%%%%%%%%%%%%%%%%%%%%%
    %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

    % the SPL and the freq of the i sub-band
    sub_band_SPL  = SPL_matrix(i,:);
    sub_band_freq = freq_matrix(i,:);
    % get the golobal and local maximas
    [peakValues, peakIndices] = Find_maximas(sub_band_SPL);     % peakValues is the spl value
                                                                % peakIndices is its position

    % looping over the peaks in the sub-band
    for j = 1: length(peakValues)

        % set the masker freq and its SPL
        masker_freq = sub_band_freq(peakIndices(j));
        masker_SPL  = peakValues(j);

        %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
        %%%%%%%%%%%%%%%%%%%%%%% The spread function for each maxima %%%%%%%%%%%%%%%%%%%%%%%%%%
        %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

        % check that SPL of the masker is not zero
        if masker_SPL ~= 0

            % calculate the most right freq affected by the masker
            f2 = masking_threshold(masker_SPL,masker_freq);
            % and its SPL
            f2_SPL = threshold_in_quiet(f2);
            % Two points to interp the spreading function
```

```
            y_spl  = [masker_SPL f2_SPL];
            x_freq = [masker_freq f2];
            % p = spreading_fn(x,y);
            p = polyfit(x_freq,y_spl,1);
            delta = 1.5;
            p = p*delta;

            %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
            %%%%%%%%%%%%%%% Masking the Non-Tonal Components for each sub-band %%%%%%%%%%%%%%%
            %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

            % looping over the affected part of the sub-band
            for k = peakIndices(j)+1 : 16
                if sub_band_freq(k) < f2
                    % get the masking threshold for tonal components
                    m_threshold = polyval(p,sub_band_freq(k));
                    if sub_band_SPL(k) < m_threshold
                        sub_band_SPL(k)=0;
                    end
                end
            end

        end
    end
    SPL_matrix(i,:) = sub_band_SPL;             % set the changes to the SPL_matrix
end
```

```
% get the vector of the Lks after the
Lk_after_masking = matrix2vector(SPL_matrix);
% plot
plot(half_freq,Lk_after_masking);
title('Tonal Components');
xlabel('freq (HZ)');
ylabel('SPL(dB)');
```



Decoder



Applying FFT to PQMF Sub-bands

```
% get the FFT to the output of the filter bank
sig_filtered_matrix = sig_filtered.';        % Transpose of sig_filtered to make rows represent
sig_filtered_fft    = zeros(32,1024);        % Initialize the output of FFT to the sig_filtered

% Apply FFT at each row (sub-band)
for i = 1:sub_num
    sig_filtered_fft(i,:) = abs(fftshift(fft(sig_filtered_matrix(i,:),1024)));
end

% get half the FFT output
```

```
half_sig_filtered_fft = sig_filtered_fft(:,513:end);
```

Masking Out the Undesired Frequencies

Here, we get the SPLs for each sub-band to mask out the undesired frequencies of the input data.

```
% calc the SPLs for each sub-band
SPLs_sig_filtered = Lk_fn(1024,half_sig_filtered_fft);

% apply masking
sig_filtered_masked = SPLs_sig_filtered.*Lk_after_masking;
```

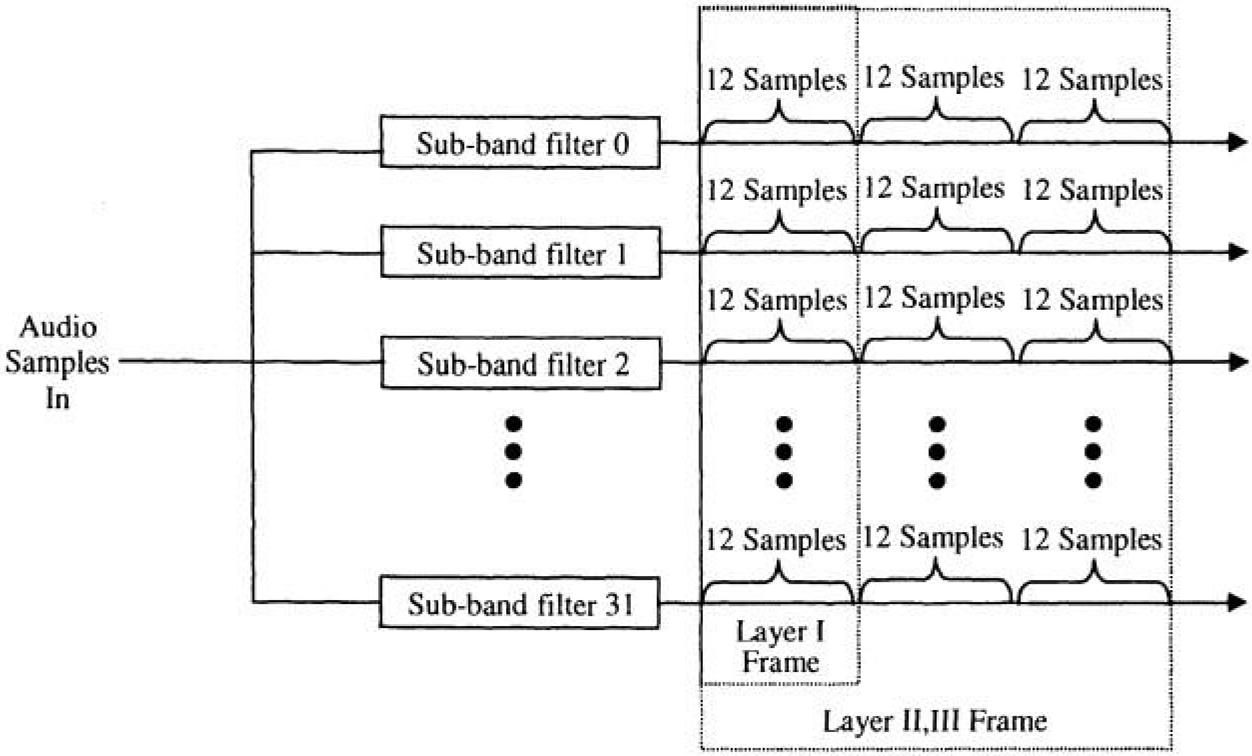
Bit Allocation

```
% calc the sum of sub-bands SPLs
sum_spl = sum(sig_filtered_masked,2).';
% bit allocation
BPS = file_info.BitsPerSample;                  % BitsPerSample from the audio info (16-bits)
sub_band_bits = bit_allocation(BPS,sum_spl);    % Apply the bit_allocation function
                                                % to get the num of bits for each sub-band
```

Quantization

We implemented this part as follows:

- Dividing data into frames, each contains 12 samples from each subband.
- Mid-Tread Quantization.
- Quan() function is used for this purpose.




```
quantized_sig = [];                              % Initialize quantized_sig Vector
% looping over each 12 cols (frame)
for j = 1:12:size(sig_filtered_matrix,2)
    frame = [];                                  % Initialize empty Frame Vector
    % looping over each row (sub-band)
    for i = 1:size(sig_filtered_matrix,1)
        % Apply Quan function on each 12 sample and set its number of bits
        slice_12 = sig_filtered_matrix(i,j:j+12-1);                         % slice 12 samp
        [maxi,mini,encoded_decimal] = Quan(slice_12,sub_band_bits(i));      % Apply Quan fu
        encoded_binary = dec2bin(encoded_decimal,sub_band_bits(i));         % convert the (
        granule = struct('max',maxi,'min',mini,'binary_code',encoded_binary);  % set max min b
        frame = [frame granule];                                            % append the gr
    end
    quantized_sig = [quantized_sig  frame];      % Append the frame to the quantized_sig
end
```

Decoder

Now, it is time to reconstuct, i.e. restore the signal values.

De-Quantization

Notes:

- As shown in the previous section, in the encoder, we stored the frames into a struct.

- The struct has values of min, max, and the binary encoded bits.
- We simply use these values for dequantization, by reversing the quantization process.
- DeQuan() function is used for this purpose.

```
dequantized_signal = [];
%extracting granules sequentially
for i=1:length(quantized_sig)
    current_granule = quantized_sig(i);
    %extracting binary representation
    encoded_binary = current_granule.binary_code;
    %converting back to decimal
    decoded_granule = bin2dec(encoded_binary);
    %calculating bit size
    bits_num = size(encoded_binary, 2);
    %de-quantizing
    dequantized_frame = DeQuan(current_granule.max, current_granule.min, bits_num, decoded_gran
    %adding to dequantized signal in a sequential manner
    dequantized_signal = [dequantized_signal dequantized_frame];
end
```

Reshaping

```
decoded_subbands_matrix = [];
num_granules = length(dequantized_signal)/(size(sig_filtered_matrix,1)*12);

index = 1;
for i=1:num_granules
    %collecting granules belonging to one frame
    frame = [];
    for j=1:32
        %selecting current granule
        current_granule = dequantized_signal(index:index+11);
        %adding to current frame
        frame = [frame;current_granule];
        index = index+12;
    end
    %adding frame to decoded subbands matrix
    decoded_subbands_matrix = [decoded_subbands_matrix frame];
end
```

Synthesis Filters

Constructing the Filter

According to the equation of g[n] introduced in the Filter Bank Part:

```
%constructing synthesis fitlers
K = 32;
N = 512;
```

```
g = zeros(K,N);

for k=0:K-1
    for n=0:N-1
        g(k + 1,n + 1) = 32 * h_coeffs(n + 1)*cos((k + 0.5)*(n + 16)*(pi/32));
    end
end
```

Applying the Filter

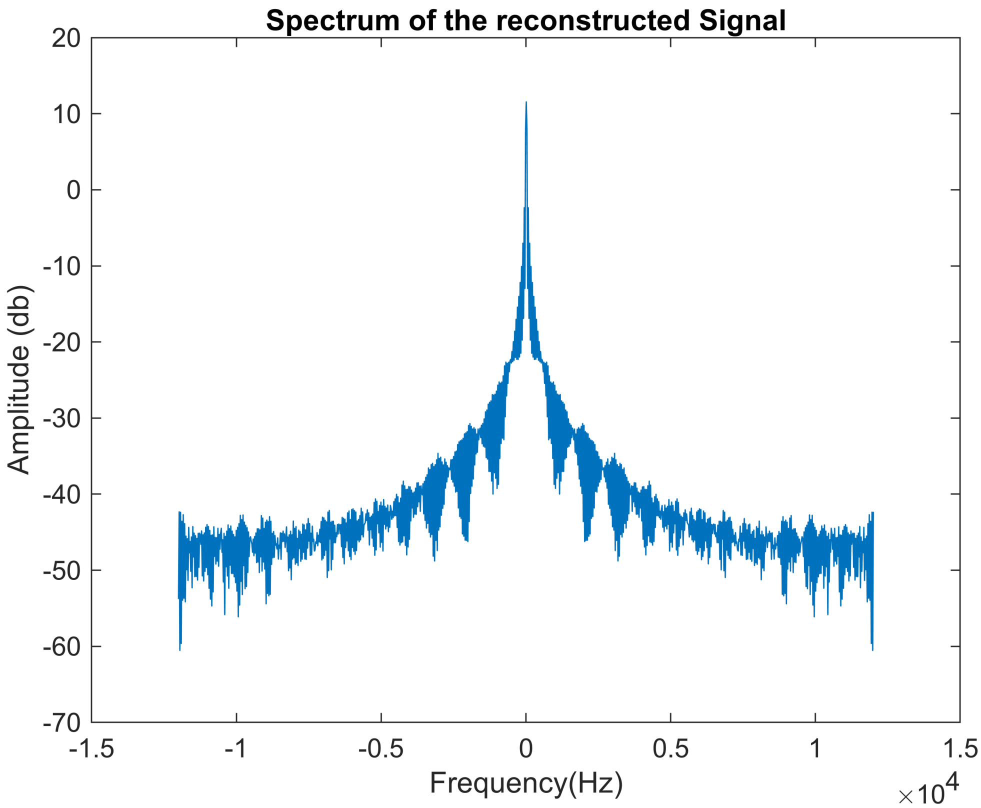
Last step in synthesizing our signal again.

```
%applying synthesis filter
decompressed_signal = zeros(1,size(decoded_subbands_matrix,1)*size(decoded_subbands_matrix,2));
for k=1:K
    %performing upsampling on extracted subband
    rec_signal = upsample(decoded_subbands_matrix(k,:), 32);
    %applying filter
    filtered_rec_signal = filter(g(k,:), 1, rec_signal);
    %summing subbands
    decompressed_signal = decompressed_signal + filtered_rec_signal;
end
```

```
sound(decompressed_signal,Fs);
```

Finally, we plot the spectrum of our reconstructed signal.

```
figure()
Signal_Spectrum_recieved = fftshift(fft(decompressed_signal , 1024));
freq = linspace(-Fs/2,Fs/2,length(Signal_Spectrum_recieved));
plot(freq, 20*log10(abs(Signal_Spectrum_recieved)));
title('Spectrum of the reconstructed Signal')
xlabel("Frequency(Hz)");
ylabel("Amplitude (db)");
```



Compression Ratio

One las thing, we calculate the Compression Ration as the ration between the size of the original and the compressed signals.

```
num_bits_original = bps*length(signal);
num_bits_compressed = sum(sub_band_bits)*length(signal)/32;
CompRatio = num_bits_original/num_bits_compressed
```

```
CompRatio = 4.1290
```