7. Filter Banks

7.1 Johnston Filters for QMF Banks

Write a script to verify the usefulness of Johnston‘s lowpass filter 12B in QMF banks.

Determine and plot the gain responses of the two analysis filters.

Compute and plot the magnitude response of the distortion transfer function.

7.2 Perfect Reconstruction Two-Channel FIR Filter Bank

Write a script to design and simulate a perfect reconstruction two-channel FIR filter bank given in the Simulink model ***perrec2cfb***.

The function ***firpr2chfb*** can be used to design the four FIR filters of the perfect reconstruction two-channel filter bank. Design the filters and visualize their magnitude responses.

Verify the perfect reconstruction property of the filter bank.

Note that the given model is used for frame-based processing.

Frame-based processing is advantageous because it allows multiple samples to process at once. A frame is a collection of samples of data and in frame-based processing, the Simulink blocks process data one frame at a time.

Initialize the model with the following parameters:

```
f1 = 350;                        % frequency f1 in Hz
f2 = 4000;                       % frequency f2 in Hz
a1 = 2;                          % amplitude a1
a2 = 10;                         % amplitude a2
fs = 10000;                      % input sampling frequency
Ts = 1/fs;                       % input sampling period
Npf = 256;                       % samples per frame
tfinal = 3;                      % stop time for the simulation
Nfr = ceil(tfinal/(Npf*Ts));     % number of frames
L = 128;                         % filter length
wp = 0.45;                       % normalized passband edge frequency
```

7.3 Fixed-Point Simulation of the Two-Channel Filter Bank for ITU Rec. G.722

The ITU recommendation G.722 describes an audio coding system for higher quality speech applications. The coding system uses sub-band adaptive differential pulse code modulation (SB-ADPCM) within a bit rate of 64 kbit/s.

The audio input signal is coded using 14 bits with 16 kHz sampling rate. The frequency band is divided into two sub-bands. A 24-coefficient QMF bank is used to compute the lower and higher sub-band signals.

The ADPCM signals are quantized to 6 and 2 bits for the lower and higher sub-bands with 8 kHz sampling rate.

The QMF coefficient values are given as follows:

```
h[0] = h[23] = 0.366211E-03, h[1] = h[22] = -0.134277E-02
h[2] = h[21] = -0.134277E-02, h[3] = h[20] = 0.646973E-02
h[4] = h[19] = 0.146484E-02, h[5] = h[18] = -0.190430E-01
h[6] = h[17] = 0.390625E-02, h[7] = h[16] = 0.441895E-01
h[8] = h[15] = -0.256348E-01, h[9] = h[14] = -0.982666E-01
h[10] = h[13] = 0.116089E+00, h[11] = h[12] = 0.473145E+00
```

Develop a Simulink model ***G722*** to simulate the QMF bank of the G.722 rec. with finite wordlength for the input samples and the filter coefficients.

Use the block **From Multimedia Device** as input for a mp3 file and the block **To Audio Device** as output. Use the blocks **FIR Rate Conversion** and **Data Type Conversion** for the required rate and type conversions.

You can accelerate the simulation by changing the simulation mode. The Rapid Accelerator mode creates a Rapid Accelerator standalone executable from your model. MATLAB and Simulink run in one process, and if a second processing core is available, the standalone executable runs there. This allows to simulate your model much faster than in the default normal mode.

7.4 DFT Analysis Filter Bank

Design a M-channel DFT analysis filter bank using a prototype linear-phase FIR filter that is designed with the M-files ***firpmord*** and ***firpm***.

Specify M as an even integer, and the passband and stopband ripple values in dB. Assume a transition band of width 0.05π around π/M.

From the prototype filter determine and plot the magnitude responses of each filter into a single figure. Plot the sum of all magnitude responses.

7.5 Cosine-Modulated 32-Channel Filter Bank

Design a cosine-modulated 32-channel filter bank.

To this end, use the function ***unicmfb*** of Doblinger with the following specifications:

```
N = 32;             % Number of filter bank channels
L = 512;            % length of prototype FIR filter
delta_f = 1/(2*N);  % normalized transition band width
alpha = 1e8;        % weighting factor for error criterion
Nfreq = L;          % Number of frequency points in passband
```

Analyse the plots generated by the function. Apply ***fvtool*** to analyse all filters in more detail.

Develop a Simulink model ***unicm32cfb*** to simulate your filter bank with the block **From Multimedia Device** as input for a mp3 file and the block **To Multimedia Device** for the audio output file.

Check the quality of the output wav-file e.g. with an appropriate player software.