Music 421A
Winter 2011-2012
Homework #7
COLA, Noise reduction, Dynamic Range Normalization
Due in one week

Theory Problems

1. (10 pts) Suppose the window transform $W(\omega)$ is a *lowpass filter* with cut-off frequency $\omega_c = 2\pi/R$ and infinite side-lobe suppression. That is, $W(\omega) \approx 0$ for $|\omega| \geq \omega_c$.

   (a) (5 pts) In this case, show that

   $$\sum_{m=-\infty}^{\infty} w(n - mR) = \frac{1}{R} W(0)$$

   irrespective of the shape of w or the shape of $W(\omega)$ in the interval $(-\omega_c, \omega_c)$.

   (b) (5 pts) Specify the set of useable frame step sizes R' such that

   $$\sum_{m=-\infty}^{\infty} w(n - mR') = \text{constant.}$$

2. (10 pts) **Constant-Overlap-Add Condition**

   (a) (5 pts) For windows in the 1-, 2-, and 3-term Blackman-Harris families, (*i.e.*, rectangular, generalized Hamming, and Blackman family), use the Poisson Summation Formula to determine the set of all hop-size values giving constant overlap-add.

   (b) (5 pts) Why does the Kaiser window not overlap-add exactly for $R > 1$? What ranges of hop sizes should be used and why? [Characterize the valid hop sizes in terms of one or more spectral properties of the window. The matlab function `ckola.m`[1] may be used to check your conclusions.]

[1] `http://ccrma.stanford.edu/~jos/sasp/hw/ckola.m`

Lab Assignment

1. (35 pts) **Bandpass Filtering for Noise Reduction**
   Write a program to denoise the corrupted birdsong in `wrenpn1.wav`[2] by means of a time-varying FIR bandpass filter. Assume that the birdsong consists of a single frequency- and amplitude-modulated sinusoid.

   (a) (5 pts) Read in frames of signal using a Hann window of length $M = 256$ and a hop-size which will give a constant overlap-add. *State clearly the hop-size you use.*

   (b) (5 pts) For each frame of noisy signal, find the single largest peak of the spectrum using `findpeaks.m` that you made for the previous homework.

   (c) (5 pts) Design a narrow bandpass filter $H(\omega)$ centered at the frequency just obtained using the same window used in analysis with the same length, $L = M$. Be sure that the filter has a unity DC gain in order to preserve our sinusoid's amplitude. [Hint: Use the window method for FIR digital filter design.]

   (d) (5 pts) Multiply $H(\omega)$ and $X_w(\omega)$ to obtain the filtered output spectrum where $H(\omega)$ and $X_w(\omega)$ is the Fourier transform of the filter impulse response and the windowed signal respectively. Be sure that your zero-padding is enough to avoid time-aliasing. *State clearly what FFT length you use.*

   (e) (5 pts) IFFT the resulting spectrum and overlap-add the frame into the output buffer. Make sure that your overlap-add is correctly aligned.

   Submit the denoised output sound.

   Also submit

   (a) a spectrogram plot of the original input signal,

   (b) a spectrogram plot of the denoised output signal, and

   (c) an overlay plot (on the same scale) of the dB magnitude spectrum of the input and output signals, for one frame near the middle.

[2] `http://ccrma.stanford.edu/~jos/sasp/hw/wrenpn1.wav`