**Lab 7 – Hilbert Transform**

* The file to be processed is fireflyintro.wav
* Design an odd length Hilbert filter and save it to a binary file (or use hilbert\_filter.bin)
* Write a C program to:
  + Compute the real and imaginary components of the Hilbert transform of the input (do this by modifying the filtering program that you wrote previously)
  + Use the atan2 function to compute the phase of the Hilbert transformed signal
  + Unwrap the phase
  + Compute the instantaneous frequency
  + Filter the instantaneous frequency through a 301-point normalized Gaussian filter (parameter = 2), see file gaussian\_2\_filter.bin
  + Save the filter instantaneous frequency

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* Make a spectrogram of the original signal (, Hamming window, 90% overlap)
* Overlay the filtered instantaneous frequency over the top of the spectrogram.

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* Describe the relationship between the instantaneous frequency and the spectrogram.

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* Plot the magnitude and phase response for the Hilbert filter and the delay filter .

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* Plot the magnitude and phase of the Hilbert transformer .

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* Describe in words the action of the Hilbert transformer based on the frequency response.

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* Plot the magnitude and phase of the cascade combination of the 1st-order difference filter and the Gaussian filter.

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