The Design of A flexible FIR Band-Pass Filter To Detect Hippocampal Sharp Wave Ripple (SWR) Implemented On Arduino Due

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1. Background

(a) What is Hippocampal Sharp Wave?

High-frequency oscillations ,known as sharp-wave/ripple complexes occurring in hippocampus during slow-wave sleep (SWS), have been proposed to promote synaptic plasticity necessary for memory consolidation. During slow wave sleep (SWS) and quiet wakefulness, the hippocampus generates high-frequency field oscillations (ripples) during which pyramidal neurons replay previous waking activity in a temporally compressed manner.
As a result, reactivated firing patterns occur within shorter time windows propitious for synaptic plasticity within the hippocampal network and in downstream neocortical structures. This is consistent with the long-held view that ripples participate in strengthening and reorganizing memory traces, possibly by mediating information transfer to neocortical areas.

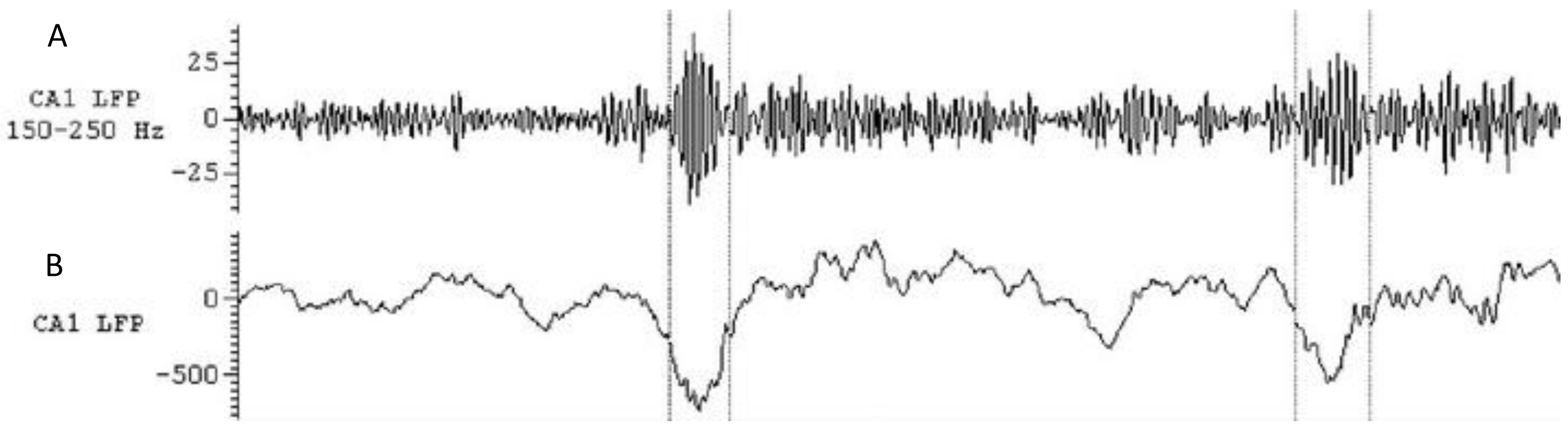
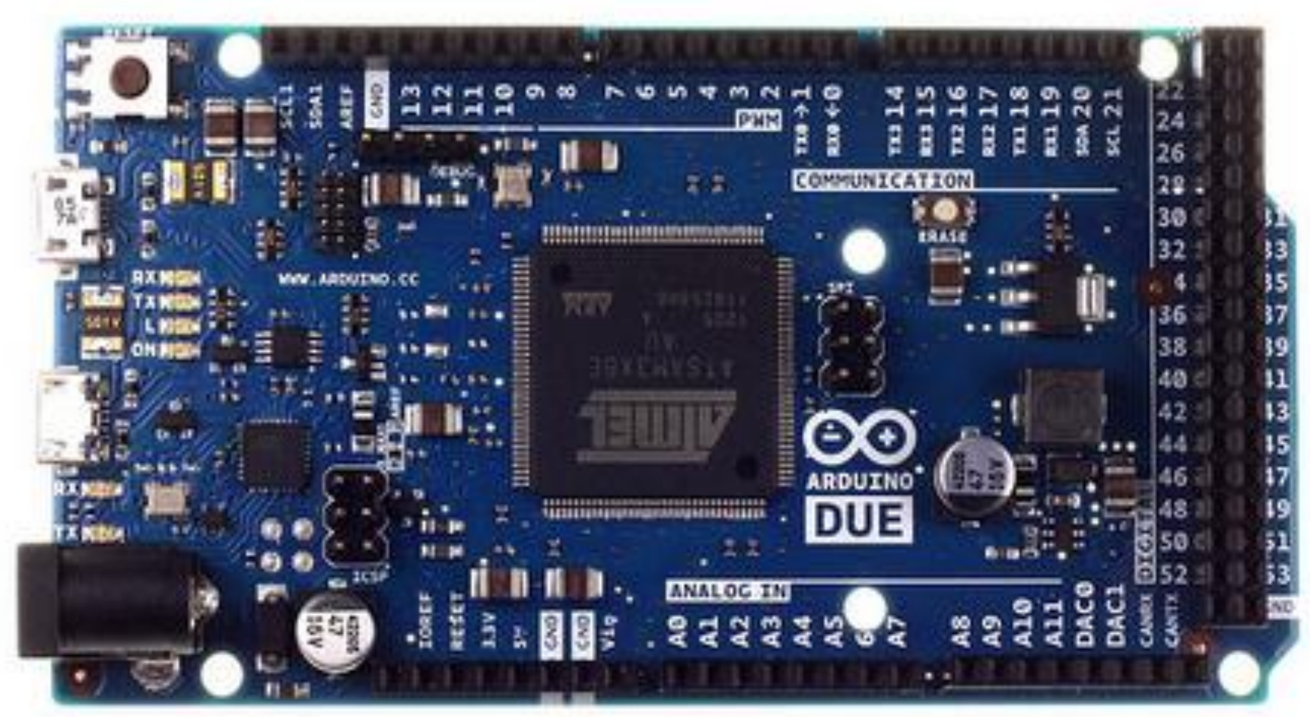


Fig A: The CA1 LFP signal after passing the band-pass filter (150Hz to 250Hz)
Fig B: The CA1 LFP signal before passing the band-pass filter

(b) What is Arduino due?

The Arduino Due is a microcontroller board based on the Atmel SAM3X8E ARM Cortex-M3 CPU. It is the first Arduino board based on a 32-bit ARM core microcontroller.



Arduino Due Front View

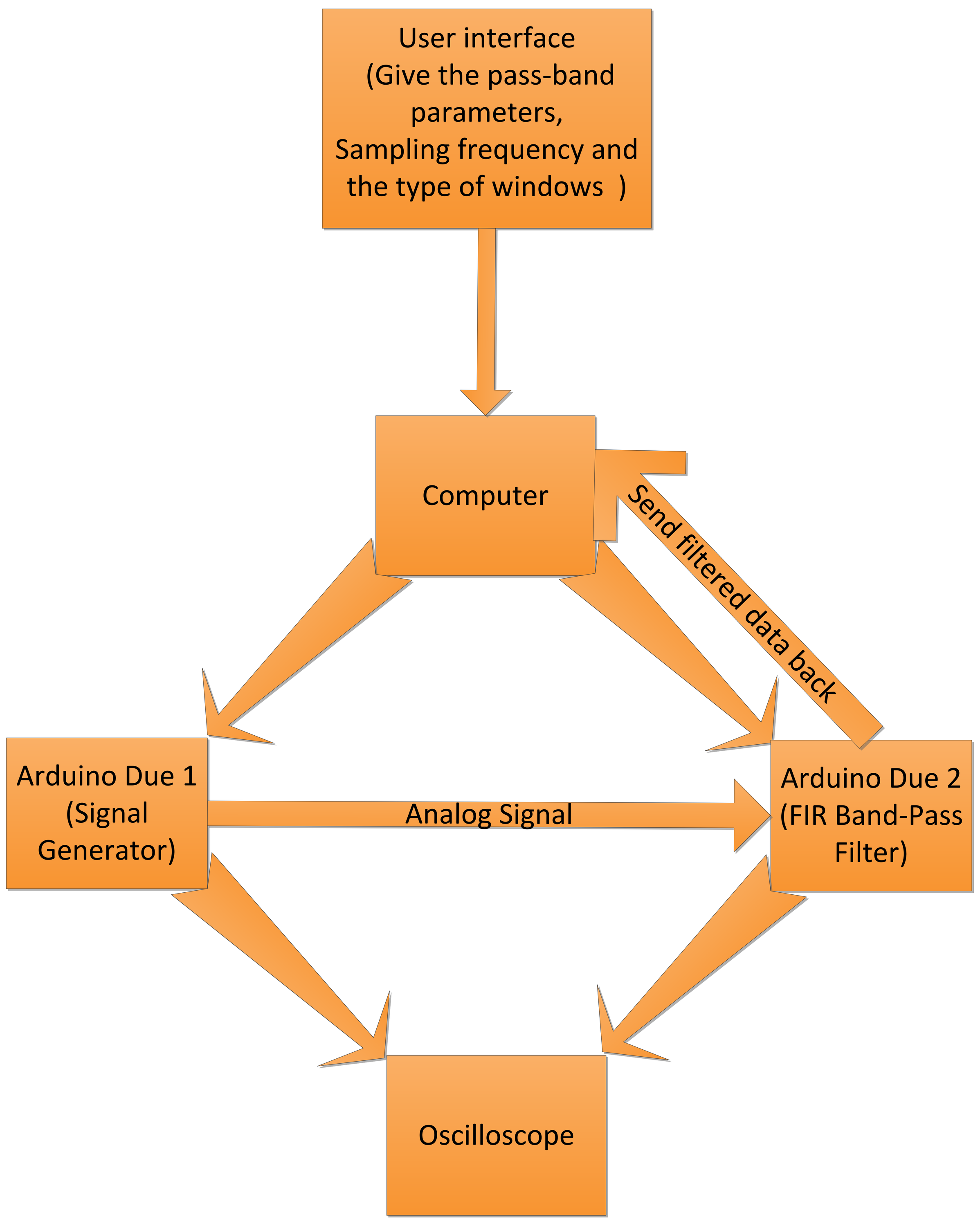
2. Basic principle & Flow diagram

Totally, two Arduino boards are used in this band-pass filter design.

One is used as the signal generator which stored the signal data in its ROM, and output the signal into analog waveform using digital-to-analog converter.

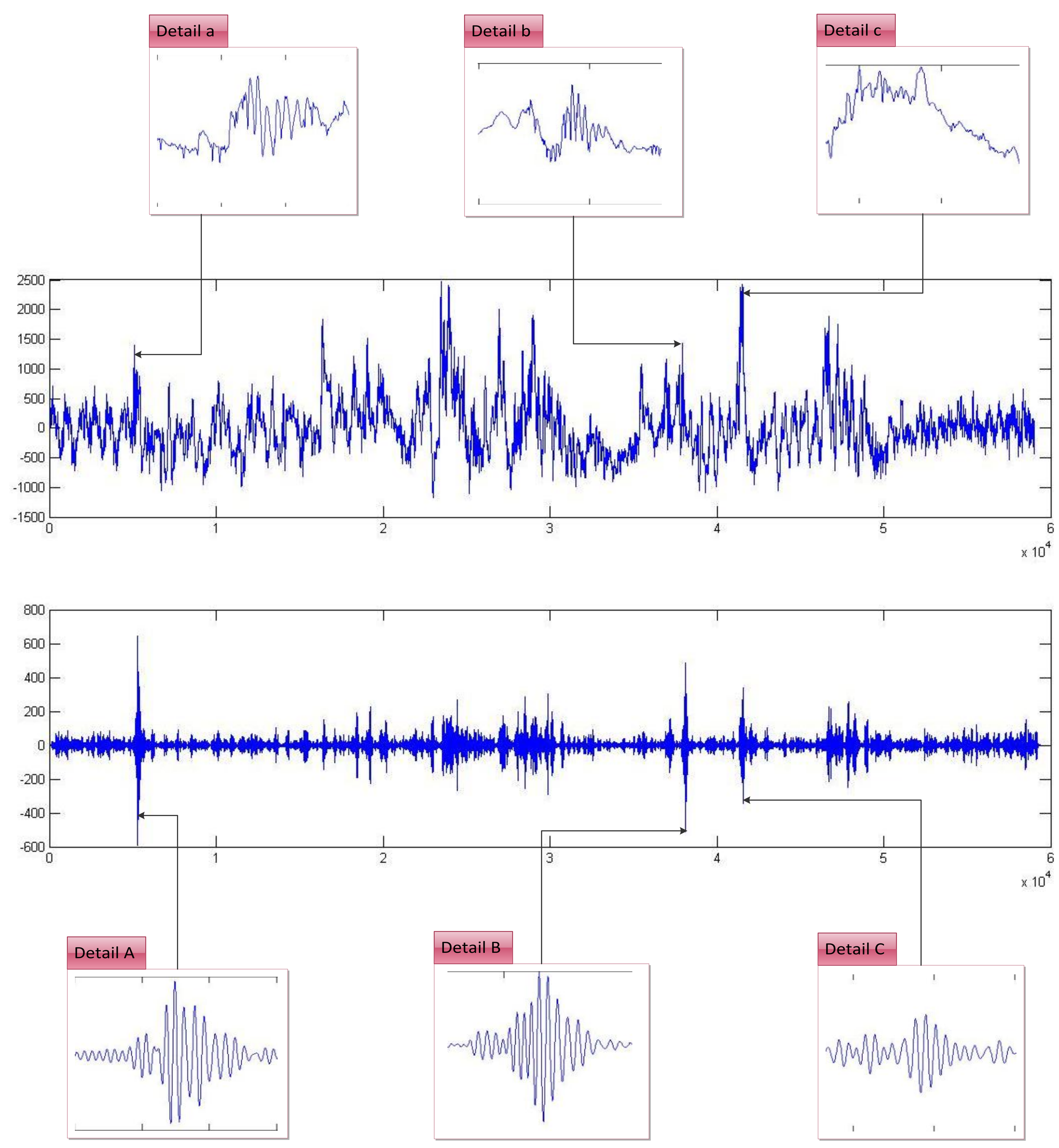
The other is used as the band-pass filter, which received the analog signal provided by the first board and do the filtering algorithm then output the filtered signal data by digital-to-analog converter.

The basic algorithm is convolution:
$y[n]=x[n]*h[n]=\sum_{m=0}^{N-1} x[m]h[n-m]$
Where h[n] is the filter coefficient, x[n] is the input sequence and y[n] is the output sequence. ($h[n]=w[n]*h_d[n]$, w[n] is window function and $h_d[n]$ is impulse response of the FIR filter.)



3. Results

The following figure shows the results of a sample CA1 LFP signal with 60000 data points and 5kHz sampling frequency. The pass band is set from 150Hz to 250Hz.



4. Further Improvements

1. Build a GUI to make it more convenient for the Filter parameters input and window types selection.
2. Build a interrupt of the system, then the user can control the sequence length to be filtered and select the data length to analysis
3. Using some algorithms of average to analysis signals from multi-channels.