A Real-Time Signal Processing System for Cochlear Implants

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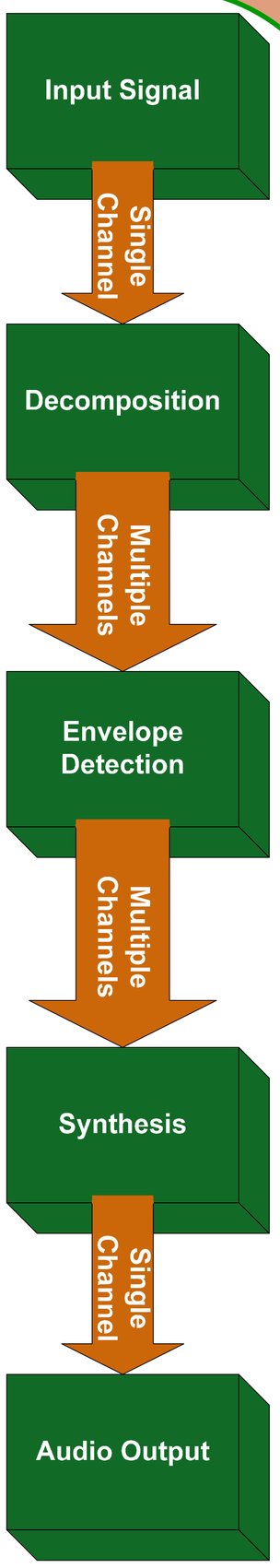
Project Goals:

- Design and implement a real-time signal processing system based upon Continuous Interleaved Sampling (CIS) strategy as deployed in cochlear implants
- Evaluate designs using qualitative hearing tests to determine the best method for deployment in a limited hardware implementation.
- Develop easy-to-use GUI environment with extensible user selectability including frequency limits, filter-order, and data word length

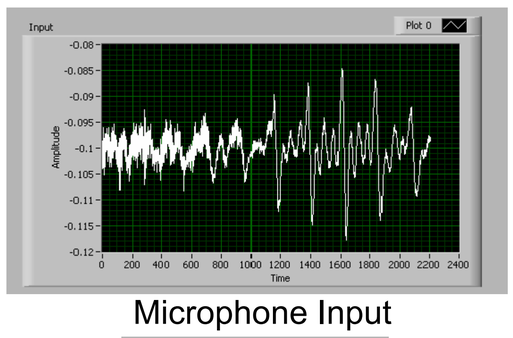
Project Overview:

- **Cochlear Implants** provide profoundly deaf patients the sensation of hearing. The implants work by generating electrical signals from sound and using the electrical signals to stimulate electrodes implanted in the patients' ears.
- The cochlear implant system consists of an input device (microphone), a signal processing system, and the output device (electrodes).
- This system simulates a cochlear implant signal processing system. The system runs using a Windows based PC with a microphone and speakers (or headphones).
- LabVIEW was used for the development of this system.
- Some processing was also done using Matlab scripting.
- **Decomposition-** Once an input sound is received, the system divides it into separate frequency bands (channels) corresponding to the number of implant electrodes.
- **Envelope Detection**- The amplitude information of each decomposed band is then extracted.
- **Synthesis-** In order to create sound output similar to what a cochlear implant patient hears, the system generates an output signal based on a noise or sinusoidal signal as well as the envelope. Synthesizing an output signal allows for analysis of the performance of the system.

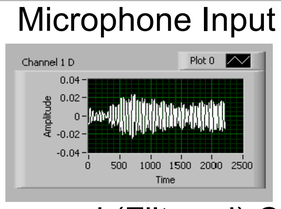


Project Results:

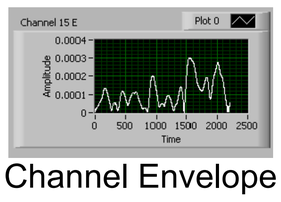
- Real-time input and output was acheived using the LabVIEW runtime environment and tools.
- The front panel environment is easy to use, allowing users to configure the system for their personal listening needs.
- Input signals are successfully decomposed; channel bands are generated using linear division of the supplied listening frequency range.
- The channel envelopes are then extracted.
- The envelopes are then supplied to the user-selected synthesizer in order to generate simulated cochlear implant hearing.
- The processing time for decomposition, envelope detection and synthesis is less than 100 milliseconds, which was determined to be the optimum real time frame length.



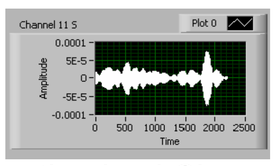
Microphone Input



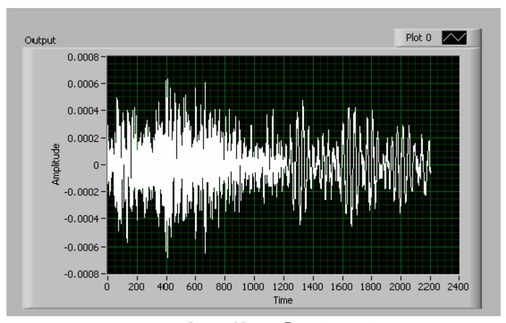
Decomposed (Filtered) Channel



Channel Envelope



Synthesized Channel



Audio Output

Project Conclusions:

- Using a greater number of channels improves output quality but slows processing time
- Smaller frame sizes reduce output delay, but stability is lost below 100ms.
- Noise band synthesis produces more natural sounding output.
- Sinusoidal harmonic synthesis produces a more metallic sounding output signal.

Possible further development:

- Resolve hardware dependencies to allow for optimum performance and portability for any hardware.
- Further perfect synthesis output.
- Add more synthesis methods.
- Further optimize performance by implementing DLL's.