**FILTERS THEORY**

***DIGITAL FILTER***

* Definition

A digital filter is a system that performs mathematical operations on a discrete and sampled time signal, so as to enhance or reduce certain aspects of that particular signal as may be necessary.

* Uses

1. It is largely used in signal processing and differs from an analog filter, which is an electronic circuit working with continuous signals.
2. Digital filters are expensive compared to analog ones, but they can turn many impractical or impossible designs into possibilities.
3. In everyday life, they can be found in devices like cell phones, radios and audio/video receivers.

* Examples

There are two types of Digital filters, namely

1. **Finite Impulse Response (FIR)** filter or **Non-Recursive** Filter
2. **Infinite Impulse Response (IIR)** filter or **Recursive** Filter
3. Finite Impulse Response (FIR)

**Finite Impulse Response (FIR)** filter is a filter whose impulse response (or response to any finite length input) is of finite duration, because it settles to zero in finite time.

1. Infinite Impulse Response (IIR)

**Infinite Impulse Response (IIR)** is a filter whose impulse response does not settle to zero but continues indefinitely. But practically, the impulse response of this filter becomes Zero after a certain point and can therefore be neglected.

* **FIR** vs. **IIR** filters

1. FIR filters are less efficient while IIR filters are more efficient.
2. FIR filter needs higher order than IIR filter to achieve the same performance.
3. FIR filter consumes low power and IIR filter need more power due to moe coefficients in the design.
4. FIR filters are more stable when compared to IIR filters.

**USE OF DIGITAL FILTER IN HEARING AID**

***Abstract***

To separate speech from noise, a machine learning program breaks a noisy speech sample into a collection of elements called time-frequency units. Next, it analyzes these units to extract 85 features known to distinguish speech from other sounds. Then, the program feeds the features into a deep neural network trained to classify the units as speech or not based on past experience with similar samples. Lastly, the program applies a digital filter that tosses out all the non-speech units to leave only separated speech.

***Application***

* The human ear captures many sound streams at once. A stream is all the sound waves that emanate from a single source, such as a dog. Together, these streams make up an auditory scene (barking + siren + talking).
* If sounds share the same frequency band at the same time, the loudest sound in a scene overpowers the others—a useful principle known as auditory masking.
* An Ideal Binary Mask filter labels Noise and speech that it within segments of sound called time-frequency units, which designate a particular brief interval within a specific frequency band. The filter analyzes each time-frequency unit in a sample of noisy speech and marks each as either 1 or 0. It records a 1 if the “target” sound (in this case, speech) is louder than noise, and a 0 if the target sound is softer.
* The result is a set of 1s and 0s that represent the dominance of noise or speech within a sample. Then, the filter tosses out all units labeled 0 and reconstructs the speech from those that scored 1. To reconstruct an intelligible sentence from noisy speech, a certain percentage of time-frequency units must be labeled 1.
* A [machine-learning program](http://web.cse.ohio-state.edu/~dwang/papers/HYWW.jasa13.pdf) that would run on a neural network and separate speech from noise after undergoing a sophisticated training process was developed. The program would use the ideal binary mask to guide the training of the neural network. And it worked. In a study involving 24 test subjects, we demonstrated that this program could boost the comprehension of hearing-impaired people by about 50 percent.

***References***

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* Kim, G., Lu, Y., Hu, Y., and Loizou, P. C. (2009). “An algorithm that improves speech intelligibility in noise for normal-hearing listeners,” J. Acoust. Soc. Am. 126, 1486–1494.