FILTER - Study Material

In signal processing, a **filter** is a device or process that removes some unwanted components or features from a signal. Filtering is a class of signal processing, the defining feature of filters being the complete or partial suppression of some aspect of the signal Most often, this means removing some frequencies or frequency bands. However, filters do not exclusively act in the frequency domain; especially in the field of image processing many other targets for filtering exist. Correlations can be removed for certain frequency components and not for others without having to act in the frequency domain. Filters are widely used in electronics and telecommunication, in radio, television, audio recording, radar, control systems, music synthesis, image processing, and computer graphics.

There are many different bases of classifying filters and these overlap in many different ways; there is no simple hierarchical classification. Filters may be:

- linear or non-linear
- time-invariantor time-variant, also known as shift invariance. If the filter operates in a spatial domain then the characterization is space invariance.
- causal or not-causal: A filter is non-causal if its present output depends on future input. Filters processing time-domain signals in real time must be causal, but not filters acting on spatial domain signals or deferred-time processing of time-domain signals.
- analog or digital
- discrete-time(sampled) or continuous-time
- passiveor active type of continuous-time filter
- infinite impulse response(IIR) or finite impulse response (FIR) type of discrete-time or digital filter.

Linear continuous-time filters

Linear continuous-time circuit is perhaps the most common meaning for filter in the signal processing world, and simply "filter" is often taken to be synonymous. These circuits are generally designed to remove certain frequencies and allow others to pass. Circuits that perform this function are generally linear in their response, or at least approximately so. Any nonlinearity would potentially result in the output signal containing frequency components not present in the input signal.

The modern design methodology for linear continuous-time filters is called network synthesis. Some important filter families designed in this way are:

- Chebyshev filter, has the best approximation to the ideal response of any filter for a specified order and ripple.
- Butterworth filter, has a maximally flat frequency response.
- Bessel filter, has a maximally flat phase delay.
- Elliptic filter, has the steepest cutoff of any filter for a specified order and ripple.

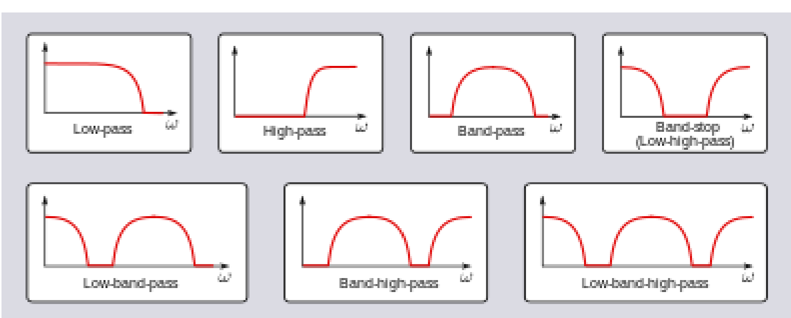
The difference between these filter families is that they all use a different polynomial function to approximate to the ideal filter response. This results in each having a different transfer function.

Another older, less-used methodology is the image parameter method. Filters designed by this methodology are archaically called "wave filters". Some important filters designed by this method are:

- Constant k filter, the original and simplest form of wave filter.
- m-derived filter, a modification of the constant k with improved cutoff steepness and impedance matching.

Terminology

Some terms used to describe and classify linear filters:



- The frequency response can be classified into a number of different bandforms describing which frequency bands the filter passes (the passband) and which it rejects (the stopband):
- Low-pass filter– low frequencies are passed, high frequencies are attenuated.
- High-pass filter– high frequencies are passed, low frequencies are attenuated.

- Band-pass filter– only frequencies in a frequency band are passed.
- Band-stop filteror band-reject filter – only frequencies in a frequency band are attenuated.
- Notch filter– rejects just one specific frequency - an extreme band-stop filter.
- Comb filter– has multiple regularly spaced narrow passbands giving the bandform the appearance of a comb.
- All-pass filter– all frequencies are passed, but the phase of the output is modified.
- Cutoff frequency is the frequency beyond which the filter will not pass signals. It is usually measured at a specific attenuation such as 3 dB.
- Roll-off is the rate at which attenuation increases beyond the cut-off frequency.
- Transition band, the (usually narrow) band of frequencies between a passband and stopband.
- Ripple is the variation of the filter's insertion loss in the passband.
- The order of a filter is the degree of the approximating polynomial and in passive filters corresponds to the number of elements required to build it. Increasing order increases roll-off and brings the filter closer to the ideal response.

One important application of filters is in telecommunication. Many telecommunication systems use frequency-division multiplexing, where the system designers divide a wide frequency band into many narrower frequency bands called "slots" or "channels", and each stream of information is allocated one of those channels. The people who design the filters at each transmitter and each receiver try to balance passing the desired signal through as accurately as possible, keeping interference to and from other cooperating transmitters and noise sources outside the system as low as possible, at reasonable cost.

Multilevel and multiphase digital modulation systems require filters that have flat phase delay—are linear phase in the passband—to preserve pulse integrity in the time domain, giving less inter symbol interference than other kinds of filters.

On the other hand, analog audio systems using analog transmission can tolerate much larger ripples in phase delay, and so designers of such systems often deliberately sacrifice linear phase to get filters that are better in other ways—better stop-band rejection, lower passband amplitude ripple, lower cost, etc.