ACTIVE FILTERS - AN OVERVIEW

This paper intends to give an overview and short introduction to the reader not familiar with the topic.

Definition:

Active filters realize transfer or filter functions - i.e. linear operators in the time- and frequency-range, which are applied to an input signal - by means of active components.

One distinguishes between time-continous and time-discrete filters:

- Continous-time filters use the information of the analog input signal for every point in time. Continous-time filters are described by Laplace- or Fourier transform in the frequency domain or by linear differential equations in the time domain.
- Discrete-time filtering means, that a sampling of the input signal takes place and only the sampled values are used for further computations. So the input signal in general is (represented by) a stream of numbers, e.g. after an AD-conversion of the analog input signal. Discrete-time filters are described by the z-transform in the frequency domain and linear difference equations in the time domain.

Types of active filters:

The active filters are essentially devided into three groups:

- Digital filters: The actual digital filter is a calculating process which is performed by digital components like DSPs or FPGAs. Should there exist AD- and DA-converters as analog interfaces, additional continous-time filters have to be applied in general as antialiasing and reconstruction filters.
- RC-filter: They consist of RC-networks and amplifiers. The RC-filter is the ‘classical active filter’ (analog, continous-time).
- SC-filter (Switched-Capacitor-Filter): They sample signals by switching MOSFET’s. The signal conditioning however, occures in an analog way, although in principle the filter is discrete in time.

Besides there exist some exotic types of active filters for special purposes like active LCR-filters or active quartz-filters.
RC- and SC-filter are often summarized under the term ‘analog filters’.

Application of active filters:

Active filters are used instead of passive filters for different reasons, to name a few:

- For low frequencies, the values of capacitances and inductances used in a passive filter can become uncomfortable high. That means large dimensions, low quality factors, polarity and current leakage in electrolytic capacitors etc.
- The tolerance and quality factor requirements on the components in an equivalent passive filter cannot be met.
- Inductances are to be totally avoided in an application (because of size, weight, nonlinearity, tolerance, low quality and sensibility with respect to EMI…).
- The filter has to do some signal amplification, anyway.

- Electronic control of some filter parameters is necessary.
- The filter does not exist in passive form (e.g. linear phase filter).
- The signals to be filtered already have digital format

The different types of filters also have different scopes of application which nowadays may overlap in many parts, mainly due to the advances in AD- and DA-converters and numerical processing power.

Digital Filters:
- Realization of almost any transfer function with very high precision.
- Advantageous if signals are already present in digital format (e.g. in image-processing) and if the application already provides a microprocessor platform.
- The filterparameters can be easily changed by reprogramming.
- Stand-alone filters are relatively complicated and clocked systems can cause considerable noise or expense in preventing this noise.
- DSP's, AD- and DA-converters with high performance aren't cheap and in some cases still space and current consuming.
- Digital filters are used mainly in complex signal conditioning when the transfer function needed cannot be realized with reasonable expense or not at all by analog filters.

SC-Filter:
- Affordable, space saving and uncomplicated in use.
- Frequency range mostly up to 250kHz, order typically 4 to 12, mostly low pass will be offered.
- Very suitable for monolithic integration, but not for discret realization - many IC-types are available.
- The filterparameters can all be re-normalized to a new reference frequency simply by changing the clock frequency.
- Precision comparable to the RC-filter.
- Used predominantly in consumer applications.

RC-Filter:
- Very low noise possible, no generation of non-harmonic spurious signals (except random noise).
- Applications are mainly in audio frequency range, but possible up to about 100MHz.
- Order typical 2 to 12, high quality factors can be realized.
- Used in antialiasing or reconstruction filters, audio technics, ultrasonic, instrumentation.

<u>Parameters of active filters:</u>

The design of active filters is, as with passive filters, built upon gain and phase specification, with the gain specifications often provided in the form of a graphical (tolerance-) scheme. Besides, the following parameters which result from the use of active and therefore nonlinear components, have to be considered:
- Operating voltage range and current consumption
- Input- and output voltage range.
- Noise / dynamic range (intermodulation etc.)
- Nonlinear distortion (in the time domain)

Source- and load impedances in contrary are generally no problem in active filters – exceptions are, of course, very low noise filter or e.g. filters, which directly have to drive heavy loads (lines).

Implementation of active analog filters:

A short outline of the implementation of active RC-filters will be given in the following. Regarding the SC-filter, which is only realized monolithically, the reader may refer to the detailed informations of the manufacturers (e.g. Linear Technology, MAXIM) or the technical literature on design of integrated circuits.

The implementation has to consider, besides the abovementioned parameters, the sensitivity of the circuit to the tolerances of the components and the phase- and frequency deviation of the amplifiers used. These sensitivities have to be minimized by choosing the optimal

- global filter structure,
- circuit arrangement,
- passive and active components.

The "global filter structure" results from the method of its synthesis out of the transfer function.
The simpliest transfer function fulfilling the requirements of the specifications will indeed lead to the simpliest filter structure, but the realization of a more complex transfer function fulfilling the specifications also will generally show smaller sensitivities and can therefore allow higher tolerances in the components. So, use of a transfer function of higher order than theoretically necessary can lead to a preferable hardware solution.
The following methods for synthesis are used among others:

> <u>Cascade realization:</u>
> Under cascade realization or cascading of filterblocks, one understands the series connection of several filtersblocks of first or second order to realize a total transfer function of higher order. For low- or highpasses, the total transfer function is decomposed for this purpose into factors (i.e. fractions) of first and second order (real pole/zero or conjugated complex pole/ zero pairs). Each factor corresponds to one filterblock of first or second order. With bandpasses or band supressors, the factorization is done already in the underlying lowpass transfer function from which the bandpass or –supressor will be derived. After factorization, a lowpass-bandpass-transformation will be done for each factor or filter block.
> There is, of course, an infinite number of possible factorizations. The numerators can be allocated to different denominators and in the choice of gain for the individual block there are as many degrees of freedom as filter blocks exist. The specifications to be fulfilled by the filter as well as the transfer functions itself determine which factorization is suitable. Compromises have to be made in almost every case. For instance, one achieves particularly small sensitivities in general, when poles and zeros of the individual factors/ filterblocks lie far apart – this factorization though, is not optimal in terms of the dynamic range of the filter.
> The technique of cascading is simple and allows a wide spectrum of electronical realizations. But especially in regard of passband sensitivity for higher order filters, the following methods are favorable:

Direct Synthesis:

The method of direct synthesis means the simulation of doubly terminated passive filters, either by directly replacing the inductances of the passive filter by active circuitry or indirectly by simulating the nodal- or mash equations of the passive filter by means of a block structure (Leapfrog Structure, Ladder Simulation). The procedure is called direct synthesis, because an active filter is directly designed out of the circuit and the component values of its passive counterpart rather than via factorization of its transfer function.

The simulation of inductances is generally accomplished by impedance converters. One possible impedance converter is the gyrator, which can be employed when OTAs (Operational Transconductance Amplifiers) are available, usually in integrated circuits. If ordinary operational amplifiers or current feedback amplifiers are to be used, the so called general impedance converter (GIC) is the better structure, also allowing the replacement of grounded an floating inductors (Gorski-Popiel method).

Follow-the-Leader-Feedback (FLF) Structure:

The idea of the FLF-Structure is to get complex filter structures by feedback of the outputs of a chain of basic filterblocks (i.e. filters of first and second order, simply called "resonators" here) to the input. So from simple filterblocks a multiplicity of complex filters of arbitrary order can be derived. The FLF-filter is in principle a cascaded filter with feedback. These filters synthesized by means of FLF-structures have generally low sensitivities, resulting from the error compensating properties of (negative) feedback.

Zeros in the transfer function can be realized by feed-forward of the output signals of particular resonators to the output of the whole structure. For the construction of resonators also many possibilities exist. The resonators again should be optimized for the task.

The circuit relevant realization of the filter depends, for example, on filter blocks of second order. They are also called biquads or resonators (of second order) and are used in the cascading technique or the FLF-structure.

Frequently used resonators of second order are:

- Filter with single negative feedback, e.g. Sallen-Key-Filter.
- Filter with double negative feedback with one or two amplifiers (MFB-Filter)
- Biquad-Circuits, which solve linear differential equations by integrators and amplifiers (State-Variable-Filter).
- GIC-Circuits, meaning impedance converters or inverters.
- Double-T-Circuits and related band supressors. Bandpass filters can be derived from supressors by subtracting the output of the suppressor from the input signal.

What a resonator of second order could look like and how the different filter parameters stand in competition with one another is for example described in our article "A relation between filter parameters of the Tow-Thomas-Biquad" on our web-site.