BANDPASS FILTERS

A bandpass filter is an electrical device that passes the spectral components of an electrical signal around a certain frequency f_0 with little or no attenuation, while it mostly rejects other spectral components (Fig. 1). Bandpass filters are common building blocks of many electronic systems. They find applications in diverse fields such as communications, audio, instrumentation, and biomedicine. To illustrate a typical application of a bandpass filter, consider the simultaneous transmission of n signals in one communications channel using frequency multiplexing techniques. The signals can share the same channel because their individual frequency spectra are shifted so that they occupy nonoverlapping regions (or bands) of the frequency spectrum. Frequency modulation is used to shift the baseband spectrum of each of the signals (centered originally at zero frequency or dc) to center frequencies $f_{01}, f_{02}, \ldots, f_{0n}$ (Fig. 2). In order to avoid overlapping, the center frequencies must have a minimum separation of at least the bandwidth of the signals. Bandpass filters are used on the receiver side in order to select or "filter out" only one of the signals transmitted in the channel. A process called demodulation is used to shift the spectrum of the selected signal back to baseband (to a zero center frequency).

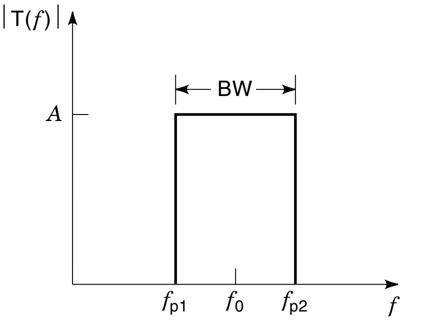

Figure 1. Frequency response of ideal bandpass filter.

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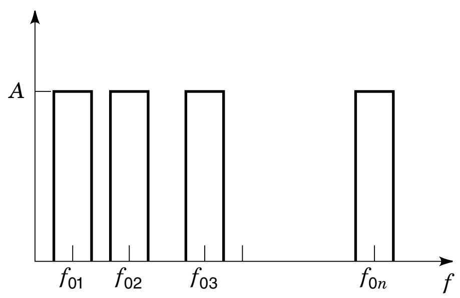

Figure 2. Spectrum of n signals in a communication channel.

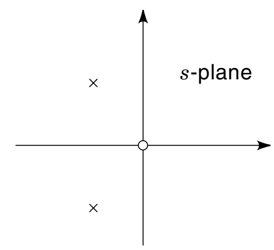

Figure 4. Pole–zero plot of second-order bandpass filter.

IDEAL BANDPASS FILTERS

Ideally, a bandpass filter is characterized by its center frequency f_0 and its bandwidth BW. The bandwidth is defined in terms of upper and lower passband edge frequencies f_{p1} and f_{p2}, respectively, and given by BW $= f_{p2} - f_{p1}$. Typically, f_{p1} and f_{p2} specify frequency points at which the gain decreases by a factor $\sqrt{2}$. The frequency response plot of Fig. 1 shows the gain A as a function of frequency f of an ideal bandpass filter. An important parameter of a bandpass filter is its selectivity factor S_F, which is expressed by $S_F = f_0/\text{BW}$. The selectivity is a measure of the filter's ability to reject frequencies outside its passband. It is also a relative measure of the reciprocal bandwidth. Highly selective filters (also denoted narrowband filters) have a bandwidth that constitutes only a small fraction of f_0. Narrowband filters with large selectivity factors (e.g., $S_F > 50$) are commonly required in communication systems. For ideal bandpass filters, the frequency range is divided into three regions: the passband region where the gain is approximately constant ($f_{p1} < f < f_{p2}$) and the lower and upper stopbands ($f < f_{p1}$ and $f > f_{p2}$, respectively), where the gain is ideally zero or the attenuation or reciprocal gain is infinite. The ideal bandpass filter has sharp transitions between the passband and the two stopbands.

PRACTICAL BANDPASS FILTERS

The typical frequency response of a practical passband filter is shown in Fig. 3. It has a relatively constant gain possibly with fluctuations (ripple) with maximum amplitude A_p within the passband. The passband edge frequencies f_{p1} and f_{p2} are defined, in this case, as the minimum and maximum frequencies with gain, $A - A_p$. The center frequency f_0 is commonly the center of geometry of the passband edge frequencies: $f_0 = \sqrt{f_{p1}f_{p2}}$. In a practical filter, there is a gradual transition between the passband and the upper and lower stopbands. Therefore, besides the passband and stopband there are also two transition bands. Two frequencies f_{s1} and f_{s2}, each denoted lower and upper stopband edge frequencies, respectively, define the stopbands in a practical bandpass filter. Stopband regions are characterized by a small gain with maximum value, $1/A_s$, where A_s is the stopband attenuation. Passband ripple A_p and stopband attenuation A_s are commonly specified in a logarithmic unit denoted decibel according to $A_s\,(\text{dB}) = 20 \log A_s$. Practical bandpass filters can have monotonic decreasing gain or equal gain fluctuations (ripple) both within the passband and the stopband.

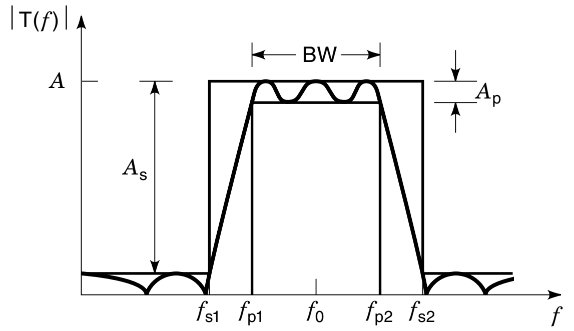

Figure 3. Practical bandpass filter.

MATHEMATICAL CHARACTERIZATION OF A BANDPASS FILTER

The simplest bandpass filter is the second-order filter. It is characterized by the input–output relation or transfer function:

$$T(f) = \frac{j\text{BW}f}{f_0^2 + j\text{BW}f - f^2} \tag{1}$$

This transfer function is a complex function with magnitude and phase. The magnitude (frequency response) obtained from Eq. (1) has the form

$$|T(f)| = \frac{\text{BW}f}{\sqrt{(f_0^2 - f^2)^2 + (\text{BW}f)^2}} \tag{2}$$

The pole–zero plot (in the complex frequency plane also known as the s-plane) of a second-order bandpass filter is shown in Fig. 4. It has a zero at the origin (shown as a circle) and two complex conjugate poles (shown as marks). The selectivity is associated with the relative distance of the poles from the imaginary axis in the s-plane. The distance from the origin to the poles corresponds to the center frequency in radians, $\omega_0 = 2\pi f_0$.

BANDPASS FILTER IMPLEMENTATIONS

Depending on the center frequency and selectivity, there exist several possible implementations of bandpass filters. In general, the higher the selectivity and the center frequency, the more complex the filter becomes. Two very important aspects are: miniaturization and compatibility with very large scale integration (VLSI) systems technology. These have dictated

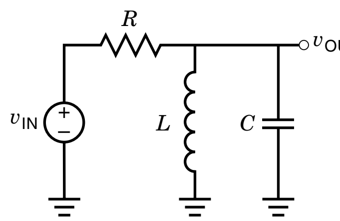

Figure 5. Second-order *RLC* bandpass filter.

the evolution of bandpass filters and of most other electronic circuits. Compatibility refers to the fact that the filter can be fabricated on a CMOS integrated circuit as a part of a VLSI system in CMOS technology. This technology allows the fabrication of very complex analog and digital electronic systems with hundreds of thousands or even millions of transistors on a single integrated circuit. CMOS compatibility is crucial in order to reduce manufacturing costs and to increase system reliability. Automatic tuning and low power consumption are also important aspects for bandpass filters. Tuning consists in the process of adjusting the filter's response to compensate for unavoidable fabrication tolerances in the values of the filter elements. These tolerances cause the filter's response to deviate from the ideal (nominal) response. Low power is important for portable equipment like cellular phones and bioimplantable devices.

Passive Filters

For many years, most filters were exclusively implemented as *passive RLC* circuits using resistors, inductors, and capacitors (1). Figure 5 shows an example of a resonant RLC circuit used as a bandpass filter. This filter is characterized by the transfer function of Eq. (1) with $f_0 = 1/2\pi \cdot 1/\sqrt{LC}$ and BW = $1/(CR)$. The transfer function of this circuit has the pole–zero plot shown in Fig. 4. Ceramic filters based on bulk electromechanical resonance phenomena in piezoelectric materials have been used in the radio frequency range specially for requirements of high selectivity and accurate center frequency. Inductors have several problems: they have large physical dimensions and cannot be integrated unless their value is very small; they are temperature dependent; and their value is subject to large manufacturing tolerances. They also have poor characteristics at low frequencies, and their value tends to change with time. *RLC* passive filters cannot be automatically tuned at the manufacturing stage. Additionally, *RLC* and ceramic filters are not compatible with CMOS VLSI technology.

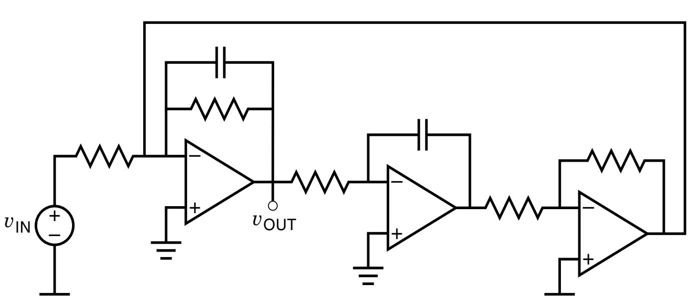

Figure 6. Second-order *RC* active filter.

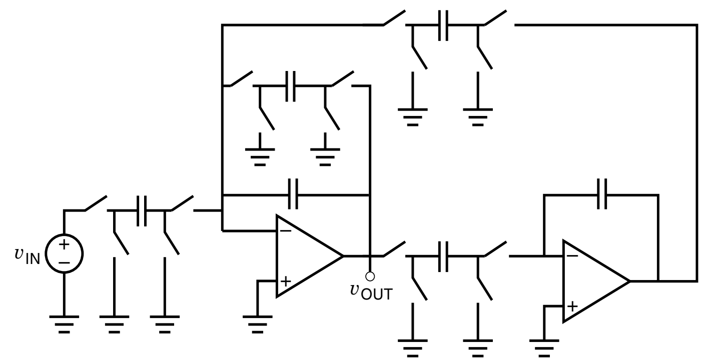

Figure 7. Second-order switched-capacitor filter.

RC Active Filters

In the 1960s, *RC* active filters using operational amplifiers, capacitors, and resistors led to the implementation of highly selective low-cost filters without inductors (Fig. 6). These filters are known as *RC* active filters (2). They constituted the first step toward filter miniaturization. *RC* active filters were mass produced in miniaturized form using thin-film and/or thick-film technology. These filters still required individual tuning. This was automatically done using computer-controlled laser beams that burn sections of resistor material to change the resistance and achieve the desired filter response. Within a short time, *RC* active filters replaced most inductor-based filters in the audio frequency range.

Monolithic Analog Filters

In the mid 1980s, *RC* active filters were replaced by fully integrated switched-capacitor filters (3). These use only switches, operational amplifiers, and capacitors and can be fabricated in CMOS VLSI technology (see Fig. 7). Switched-capacitor filters require no tuning and can operate at center frequencies

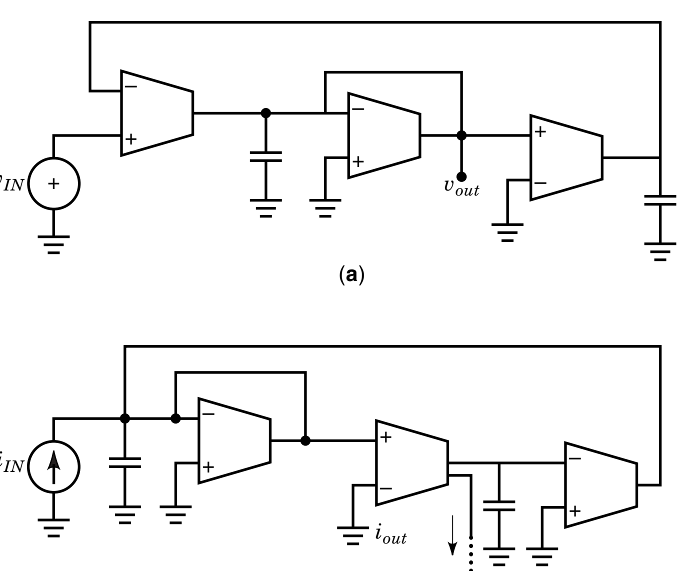

Figure 8. Second-order OTA-C bandpass filter: (a) voltage mode implementation; (b) current mode implementation.

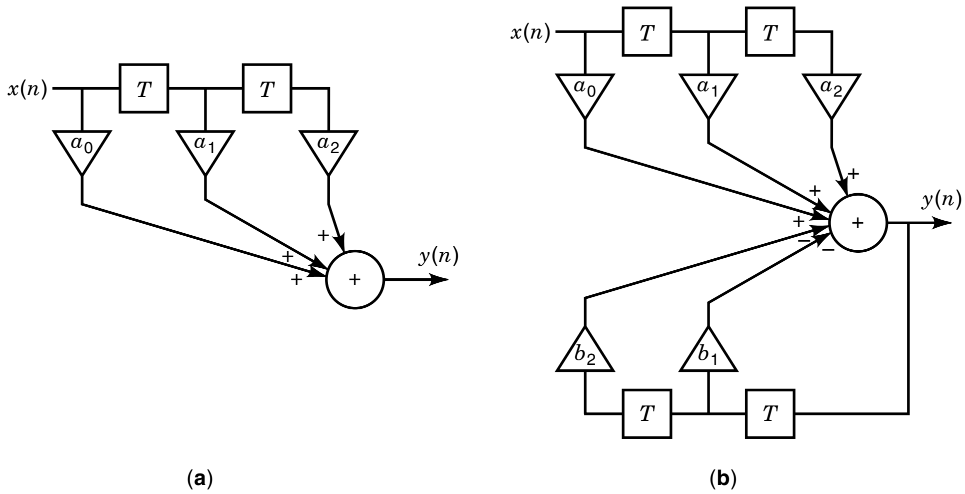

Figure 9. Second-order digital filters (a) FIR filter; (b) IIR filters.

up to a few hundred kilohertz. Starting in the mid 1970s surface acoustic wave (SAW) filters—another family of filters—were developed to implement filters for center frequencies spanning from 10 to several hundred megahertz. These filters are miniaturized filters that also require no tuning. They are based on the controlled propagation of electromechanical waves on the surface of a piezoelectric crystal with very small dimensions. SAW filters are not compatible with CMOS technology. But because of their excellent performance characteristics and low cost, they are currently being used to implement the intermediate frequency filter of most televisions.

Recently, electronically tunable elements in CMOS technology have allowed the integrated circuit implementation of active filters for center frequencies up to a few tens of megahertz. These filters are compatible with CMOS VLSI technology. Most of these implementations are based on electronically tunable elements called operational transconductance amplifiers (transconductors or OTAs) and include on-chip automatic tuning circuitry (4). The tuning circuit continuously (or periodically) monitors the filter's response and electronically adjusts the gain of the OTAs to tune the frequency response of the bandpass filter. This family of filters is known as OTA-C filters (see Fig. 8). They require no tuning and can be integrated as part of a VLSI system (5). Figure 8a shows a conventional "voltage mode" filter, while Fig. 8b shows a "current mode" filter where input, output, and intermediate variables are represented by electrical currents rather than voltages as it is done in conventional "voltage mode" filters (6). In analog signal processing systems, the range above 100 MHz is still based, to a large exent, on passive *RLC* filters. At these frequencies, high-quality, small-dimension inductors are available. A very recent trend for the implementation of high-frequency high-selectivity integrated analog filters is based on microelectromechanical (MEM) structures, which are based on electromechanical resonance. This approach has shown potential for the implementation of bandpass filters operating in the radio frequency range and with very high selectivity factors similar to those achievable with ceramic and SAW structures. These MEM-based filters, unlike the SAW filters, are CMOS VLSI compatible.

Digital Filters

Currently, the availability of low-cost digital circuitry in VLSI systems has made it convenient to replace many traditional analog functions by digital ones and to process signals in the digital domain rather than in analog form. Successful implementation of digital filters for center frequencies up to a few megahertz, as stand-alone units or as part of a VLSI system has recently been achieved. Digital filters are fully integrated filters that are compatible with CMOS technology and require no tuning (7). Their characteristics (f_0, BW, or Q) can be easily reprogrammed for many different applications. A digital filter consists basically in various addition, multiplication, and delay operations applied to an ordered sequence of numbers that represent the digitized values of a signal. These digitized values are obtained by sampling the signal at regular time intervals ($t = T, 2T, 3T, \ldots$) and transforming the sampled values into binary codes. In digital filters, present values $x(n) = x(nT)$ and past values $x(n-1) = x((n-1)T)$, $x((n-2) = x((n-2)T) \ldots$ of both input and output signals are processed. The steps performed on these digitized values lead to an output sequence $y(n), y(n-1), y(n-2), \ldots$, which can be transformed by means of a digital-to-analog converter into a filtered output signal $y(t)$. A digital filter is, therefore, an algorithm or sequence of mathematical steps that relates input and output sequences by means of multiplication, addition, and delay operations. A digital filter is mathematically characterized by a difference equation. There are two types of digital filters: finite impulse response (FIR) filters, where the current value of the output signal $y(n)$ depends only on the current and past values of the input signal, and infinite impulse response (IIR) filters, where the current value of the output signal depends also on past values of the output signal. For example, the difference equation of a second-order FIR filter is expressed by

$$y(n) = a_0 x(nT) + a_1 x(n-1) + a_2 x(n-2) \tag{3}$$

while the difference equation of a second-order IIR digital filter has the form

$$\begin{aligned} y(n) = {} & a_0 x(n) + a_1 x(n-1) + a_2 x(n-2) \\ & - b_1 y(n-1) - b_2 y(n-2) \end{aligned} \tag{4}$$

where a_0, a_1, a_2, b_1, and b_2 are multiplying coefficients; $x(n)$ and $y(n)$, stand for $x(t = nT)$ and $y(t = nT)$, are the current values of the input and output signals; and $x(n-1)$, $x(n -$

2), $y(n-1)$, and $y(n-2)$ correspond to the two previous values of the input and output signals, respectively. The values of the multiplying coefficients and the sampling frequency, $f_s = 1/T$, determine the selectivity and center frequency of the digital filter and can be easily reprogrammed. Figures 9(a, b) illustrate the block diagram of second-order FIR and IIR filters, respectively.

High-speed digital filters can be implemented using special-purpose VLSI hardware in the form of digital signal processors (6) or as relatively low-speed filters using software in general-purpose digital systems such as computers or microprocessors.

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JAIME RAMIREZ-ANGULO
New Mexico State University

BAND-REJECT FILTERS. See BAND-STOP FILTERS.