Chapter 3 Signal Conditioning

3.1 Introduction of signal conditioning

In the study of electronics, signal conditioning signifies manipulating an analog signal in such a way that meets the requirements of the next stage for further processing.

In communication and control engineering applications, it is common to have a sensor, a signal conditioning stage (generally for amplification and calibration), and a processing stage which includes required analog-to-digital signal conversion, modulation, various operations like filtering in transformed domain and calibration.

Signal conditioning can include amplification, filtering, transformation, conversion, modulation, range matching, isolation, and many more processes which may be required to make sensor output suitable for processing after conditioning.

3.1.1 Filtering

Filtering is the most common signal conditioning function, as not all the frequency components of a signal contains valid data, usually. Filtering is the process of ensuring the usable part of the signal to pass for the next stage of signal processing and rejecting the unwanted part. In electrical signal, to understand the process of filtering, it is essential to study the transformed domain representation of signal and their implications. In two-dimensional signals such as images, a new term spatial frequency is dealt with, which is essentially the rate of change of intensity with respect to space.

3.1.2 Amplifying

Signal amplification performs two important functions: increases the resolution of the input signal, and increases its signal-to-noise ratio. For example, the output of an electronic temperature sensor, which is mostly in the millivolts range, is too low for an analog-to-digital converter (ADC) to process directly. In this case it is necessary to bring the voltage level up to that required by the ADC.

Commonly used amplifiers in signal conditioning include sample and hold amplifiers, peak detectors, log amplifiers, antilog amplifiers, instrumentation amplifiers, or programmable gain amplifiers.

3.1.3 Isolation

Signal isolation must be used to pass the signal from the source to the measurement device without a physical connection: it is often used to isolate possible sources with signal perturbations. It is important to isolate the potentially expensive equipment used to process the signal after conditioning from the sensor.

For this purpose, generally magnetic, optical, or combination of resistive, inductive, and capacitive isolations are used.

3.1.4 Modulation

To maintain realizable antenna height, to provide the facility of multiplexing and channel separation, and to get higher level of noise immunity, a number of modulation schemes are employed in the case of both analog and digital signal conditioning like communication.

Modulation can be defined as a technique of modification of eath or all the three basic properties of analog single tone carrier signal with respect to the baseband signal. As the baseband signal (information) is solely responsible for the task of modulation, it is also known as the modulating signal. A single tone analog carrier signal can be expressed as

$$x(t)=A\sin(\omega t+\varphi), \tag{3.1}$$

where, A is amplitude of the carrier signal; ω is frequency of the carrier signal; φ is initial phase of the carrier signal.

The amplitude A, frequency ω, and phase $\theta(t)=\omega t+\varphi$ are the three fundamental parameters of any signal. Therefore, we can modify/modulate any one of these three parameters with respect to the analog modulating signal. Combining the frequency and initial phase part, the time-dependent angle part $\theta(t)=\omega t+\varphi$ is obtained. So, angle modulation is said to be the mother modulation scheme of frequency and phase modulations.

3.2 Filtering

In this section we will consider the basics of low-pass, high-pass, band-pass, band-elimination filters, and the ideal delay line filters. The filters are specified based on a transfer function $H(\mathrm{j}\omega)$ in terms of its amplitude $|H(\mathrm{j}\omega)|$, phase $\angle H(\mathrm{j}\omega)$, or delay responses $-|\mathrm{d}H(\mathrm{j}\omega)/\mathrm{d}\omega|$, Finding $H(\mathrm{j}\omega)$ from the specifications is the first step. The next step involves the synthesis. We will consider here the ideal filter functions that describe their functions, and simple circuits that can be used as filters.

Low-pass filters allow low frequencies to pass through with small attenuation and attenuate or eliminate high frequencies; high-pass filters eliminate or attenuate low frequencies and allow high frequencies go through with possible small attenuations; band-pass filters allow a band of frequencies to go through with small attenuation and attenuate or eliminate frequencies that are outside this band; band-elimination or band-reject filters let the low and high frequencies pass through and attenuate or eliminate a band of frequencies somewhere in the middle. The amplitude and phase plots of these filters are shown in Fig. 3.1. Filters are used in every communication system. If the frequencies of the two signals are disjoint, then we can remove the undesired signal by using a band-pass filter that allows the desired signal to go through with a small attenuation and attenuate or eliminate the undesired signal. Tuning to a particular radio station involves eliminating, i. e. filtering out all the other signals from the other stations all available at the front end of the radio or TV receivers. The DC component can be removed by using a high-pass filter or a simple bias removal component, a capacitor.

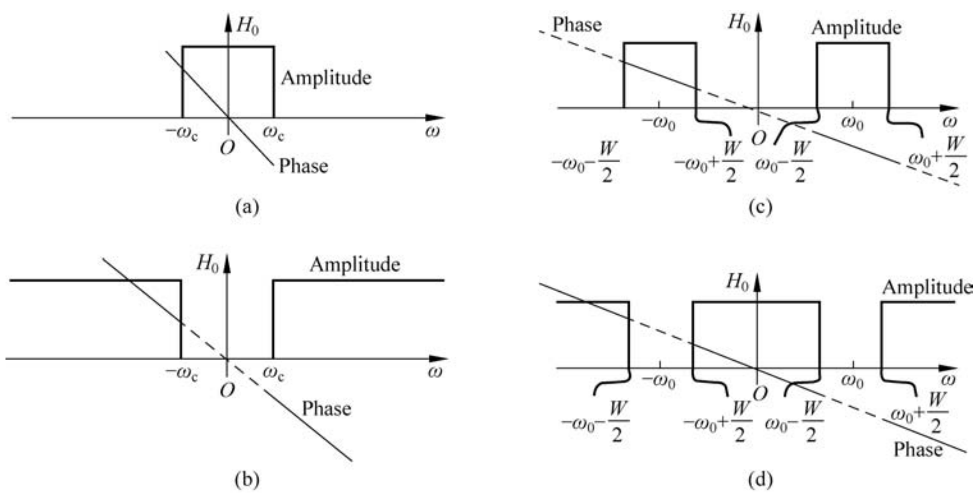

Figure 3.1 Amplitude and phase plots of ideal filters

(a) Low-pass; (b) High-pass; (c) Band-pass, and (d) Band elimination

Filters can be implemented in terms of either analog or digital domain. Next we will consider each of the filter types in a more formal fashion and discuss the generation of simple transfer functions that allow for the analysis of these filters.

3.2.1 Simple low-pass filters

The words low-pass means that when the signal $x(t)$ is passed through a low-pass filter, only low frequencies, say 0 to $f_c = \omega_c/2\pi$, are passed and it blocks all frequencies above the cutoff frequency f_c. When H_0 is assumed to be positive, the amplitude of the ideal low-pass filter transfer function is

$$|H_{LP}(j\omega)| = H_0 \prod \left[\frac{\omega}{2\omega_c}\right] = \begin{cases} H_0 & |\omega| \leqslant \omega_c \\ 0 & |\omega| \geqslant \omega_c \end{cases}, \quad \omega_c = 2\pi f_c. \tag{3.2}$$

Every transmission system takes time, i.e., the signal will be delayed. It is ideal to have this delay to be a constant, say t_0 for all frequencies, which may not be possible. In terms of frequency domain, the output transform of such a system is

$$Y(j\omega) = H_{LP}(j\omega)X(j\omega), \quad H_{LP}(j\omega) = H_0 \prod \left[\frac{\omega}{2\omega_c}\right] e^{-j\omega t_0},$$

$$|H_{LP}(j\omega)| = H_0 \prod \left[\frac{\omega}{2\omega_c}\right], \quad \angle H_{LP}(j\omega) = -\omega t_0. \tag{3.3}$$

The amplitude and the phase response plots are shown with respect to the frequency in Fig.3.2. On the magnitude plot, the band of frequencies from 0 to f_c as the pass band and the band of frequencies from f_c to ∞ as the stop band are shown. Since the amplitude spectrum of a real signal is even and the phase spectrum is odd, the discussion can be limited to only positive frequencies. The phase response is assumed to be linear, i.e., the slope is constant. The group delay is

$$t_0 = -\mathrm{d}\frac{\angle H_{LP}(j\omega)}{\mathrm{d}\omega}. \tag{3.4}$$

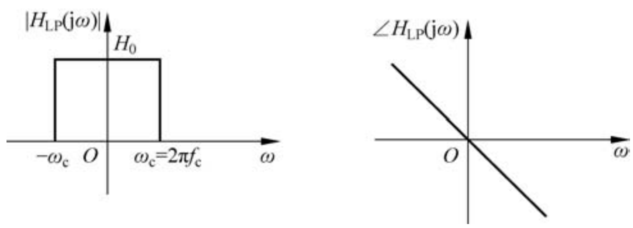

Figure 3.2 Amplitude and phase plots of an ideal low-pass filter

Can we design a real circuit that has the transfer function $H_{LP}(j\omega)$? For a physically realizable system, the impulse response $h(t) = 0$ for $t < 0$, i.e., the system is causal. For a realizable system, the output cannot exist before the input is applied. The impulse response of the ideal low-pass filter can be determined from below.

$$\frac{\sin(a(t-t_0))}{\pi(t-t_0)} \overset{FT}{\Longleftrightarrow} \prod \left[\frac{\omega}{2a}\right] e^{-j\omega t_0}, \tag{3.5}$$

$$h_{\mathrm{LP}}(t)=F^{-1}[H_{\mathrm{LP}}(\mathrm{j}\omega)]=F^{-1}\left[H_0\prod\left[\frac{\omega}{2\omega_c}\right]\mathrm{e}^{-\mathrm{j}\omega t_0}\right]=H_0(2f_c)\frac{\sin(\omega_c(t-t_0))}{\omega_c(t-t_0)}. \tag{3.6}$$

In Fig. 3.3, the input, an impulse function $\delta(t)$, applied at $t=0$, the block diagram representing the ideal low-pass filter and the impulse response are identified. The impulse response, a sinc function, peaks at $t=t_0$, giving a value of $(2f_cH_0)$ at this time. From the figure we can see that the response is nonzero for $t<0$. That is, there is output before the input is applied. The ideal low-pass filter is not causal and is physically unrealizable.

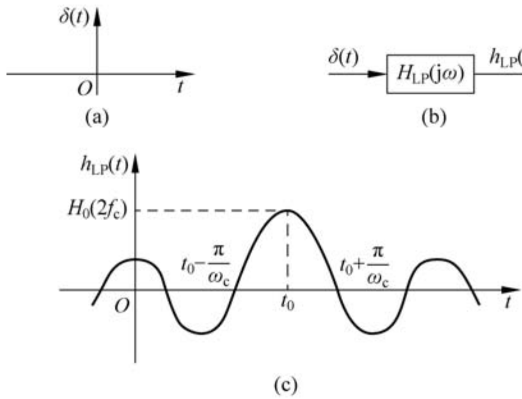

Figure 3.3 Amplitude and phase responses of an ideal low-pass filter to an impulse signal

For a causal system, the impulse response $h(t)=0$ for $t<0$, as it does not respond before the input is applied. The causality condition can be stated in terms of the transfer function $H(\mathrm{j}\omega)$. It is called the Paley-Wiener criterion[115] and is given in terms of the inequality

$$\int_{-\infty}^{\infty}\frac{|\ln|H(\mathrm{j}\omega)||}{(1+\omega^2)}\mathrm{d}\omega<\infty.$$

If $|H(\mathrm{j}\omega)|=0$ over a finite frequency band, then the above integral becomes infinite. $|H(\mathrm{j}\omega)|$ can be zero at isolated frequencies and still satisfy the criterion. The criterion describes the physical reliability conditions and is not of practical value.

That is, if $|H(\mathrm{j}\omega)|=0$ over any band of frequencies, the Paley-Wiener criterion states that the system is physically unrealizable. Ideal low-pass filter violates the condition. We can make a general statement that if the amplitude spectrum is a brick wall type function, the corresponding transfer function is physically unrealizable. Since the ideal low-pass filter function is physically unrealizable, the next best thing is to find a function that approximates the ideal filter characteristics. First, consider the simple RC circuit in Fig. 3.4. Its transfer function is

$$H_{\mathrm{LP}}(\mathrm{j}\omega)=\frac{1}{(1+\mathrm{j}\omega RC)}. \tag{3.7}$$

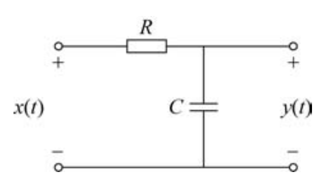

Figure 3.4 Simple RC circuit

The frequency amplitude characteristic is shown in Fig. 3. 5 with the cutoff frequency $f_c = f_{-3dB} = (1/2\pi RC)$. The input and the output transforms are related by $Y(j\omega) = H_{LP}(j\omega)X(j\omega)$.

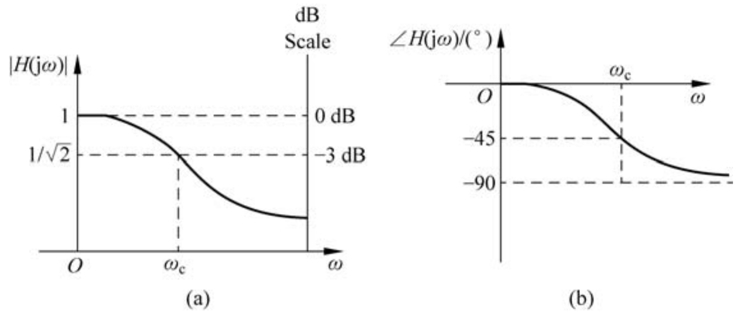

Figure 3. 5 (a) Amplitude response and (b) Phase response

■ Example 3. 1

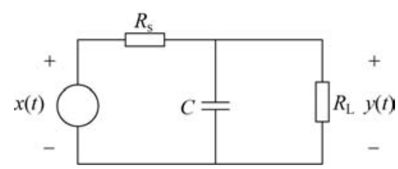

Figure 3. 6 Example 3. 1

Consider the RC circuit shown in Fig. 3. 6 with the source and the load resistors. Derive the transfer function and sketch the amplitude characteristic function for the two cases. a. $R_L = \infty$ and b. $R_L = R_s$.

Solution: The transfer functions are given by

$$\begin{cases} \left[\dfrac{Y(s)}{X(s)}\right] = H_{LP}(s) = \dfrac{Z_2}{Z_2 + Z_1}, \\ Z_2 = \dfrac{R_L/Cs}{R_L + \dfrac{1}{Cs}} = \dfrac{R_L}{1 + R_L Cs}, \quad Z_1 = R_s. \end{cases} \tag{3.8a}$$

$$\begin{cases} \left[\dfrac{Y(s)}{X(s)}\right] = \dfrac{\dfrac{R_L}{R_s R_L C}}{s + \left[\dfrac{R_L + R_s}{R_s R_L C}\right]} = \dfrac{K}{s + \omega_c}, \\ K = \dfrac{1}{R_s C}, \quad \omega_c = \dfrac{R_s + R_L}{R_s R_L C}. \end{cases} \tag{3.8b}$$

In case a, the load resistance is infinite, i. e. , the circuit is not loaded. In case b, $R_L = R_s$. For the two cases the corresponding transfer functions are

$$\begin{cases} \text{a. } H_{LP, R_L \to \infty}(s) = \dfrac{1/R_s C}{s + \dfrac{1}{R_s C}}, \\ \text{b. } H_{LP, R_L = R_s}(s) = \dfrac{1/R_s C}{s + \dfrac{2}{R_s C}}. \end{cases} \tag{3.9}$$

In both cases, the gain constant is the same. However, the cutoff frequency is increased in the case of a load resistance. Note that the peak value of the amplitude response function in case b. is 1/2. So, the 3 dB frequency corresponds to the value of the magnitude of the function equal to $(1/2)\times(1/\sqrt{2})$. At $\omega=0$ the filter circuit is transparent; at $\omega=\infty$ there is no signal transmission; in between these frequencies, the output signal amplitude attenuation is determined by the equation $|Y(j\omega)|=|H(j\omega)|\cdot|X(j\omega)|$. At the 3 dB frequency $\omega_{3\text{ dB}}$

$$|Y(j\omega)|_{\omega=\omega_{3\text{ dB}}}=\frac{1}{\sqrt{2}}|X(j\omega)|_{\omega=\omega_{3\text{ dB}}}.$$

Notes: For simplicity, generic functions for the input $x(t)$ and the output voltage $y(t)$ are used. Usually, $V_i(t)$ (or $V_s(t)$) for the input and $V_o(t)$ for the output are convenient.

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3.2.2 Simple high-pass filters

In the ideal low- and high-pass filter cases as shown in Figs.3.1 (a) and (b), it can be seen that

$$\begin{cases}|H_{LP}(j\omega)|_{\omega=0}=|H_{HP}(j\omega)|_{\omega=\infty}=1,\\ |H_{HP}(j\omega)|_{\omega=0}=|H_{LP}(j\omega)|_{\omega=\infty}=0.\end{cases}\tag{3.10}$$

In addition, the amplitudes of these functions transition at the frequency $\omega=\omega_c$ and the change in the amplitudes are as follows:

$$1\ (\text{low}-\text{pass})\rightarrow 0(\text{high}-\text{pass})]\quad\text{or}$$
$$0\ (\text{high}-\text{pass})\rightarrow 1(\text{low}-\text{pass}).$$

A logical conclusion is that $\omega\rightarrow 1/\omega$ (i.e., $s\rightarrow 1/s$) provides a transformation that gives a way to find a high-pass filter function from a low-pass filter function. Noting that the impedance of an inductor is $j\omega L$ and the impedance of a capacitor is $1/j\omega L$, a high-pass filter can be obtained from a low-pass filter by replacing a capacitor by an inductor. Since the impedance of the resistor R is independent of the frequency, no change is necessary in the case of resistors. Using the RC and the RL low-pass circuits. We have two simple high-pass filters as shown in Figs.3.7 (a) and (b), one is an RL and the other one is an RC circuit. The filter is a low-pass if the inductor is in the series arm and the capacitor in the shunt arm. Similarly the filter acts as a high-pass filter if the capacitor is in the series arm and the inductor is in the shunt arm. The transfer functions corresponding to the two circuits in Fig.3.7 are

$$\begin{cases}H_{Hpb}(s)=\dfrac{s}{s+\dfrac{R}{L}},\quad H_{Hpa}(s)=\dfrac{s}{s+\dfrac{1}{RC}};\\ H_{Hpb}(j\omega)=\dfrac{j\omega}{j\omega+\dfrac{R}{L}},\quad H_{Hpa}(j\omega)=\dfrac{j\omega}{j\omega+\dfrac{1}{RC}}.\end{cases}\tag{3.11}$$

The amplitude and the phase response characteristics of these are as follows:

$$|H_{\text{Hpa}}(j\omega)| = \frac{|\omega|}{\sqrt{\omega^2 + \left(\frac{1}{RC}\right)^2}},$$

$$\angle H_{\text{Hpa}}(j\omega) = 90^\circ - \arctan(\omega RC). \tag{3.12}$$

$$|H_{\text{Hpb}}(j\omega)| = \frac{|\omega|}{\sqrt{\omega^2 + \left(\frac{R}{L}\right)^2}},$$

$$\angle H_{\text{Hpb}}(j\omega) = 90^\circ - \arctan\left(\frac{\omega L}{R}\right). \tag{3.13}$$

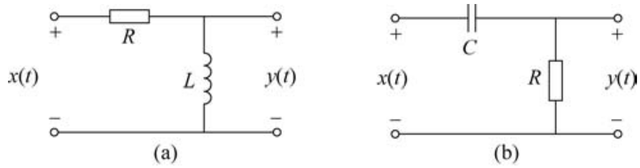

Figure 3.7 Simple high-pass filters

(a) RL circuit; (b) RC circuit

The amplitude and phase responses are shown in Fig. 3.8 for $\omega > 0$: the maximum value of the amplitude response is 1 or 0 dB. The 3 dB frequencies can be computed by equating the amplitude response function to $1/\sqrt{2}$ and solving for ω. That is

$$\frac{1}{\sqrt{2}} = \frac{|\omega_{\text{cb}}|}{\sqrt{\omega_{\text{cb}}^2 + \left(\frac{R}{L}\right)^2}}, \quad \frac{1}{\sqrt{2}} = \frac{|\omega_{\text{ca}}|}{\sqrt{\omega_{\text{ca}}^2 + \left(\frac{1}{RC}\right)^2}}. \tag{3.14}$$

$$\omega_{\text{cb}} = \frac{R}{L}, \quad \Rightarrow \omega_{\text{ca}} = \frac{1}{RC}, \tag{3.15}$$

As in the low-pass case, given the cutoff frequency ω_c, one of the reactive component (inductor or capacitor) values can be solved by selecting the resistor value. The high-pass filter is transparent from the input to the output at infinite frequency and no signal transmission at zero frequency. Note the low-frequency and high-frequency behaviors of the low-pass and the high-pass filter functions $H_{\text{LP}}(s)$ (or $H_{\text{HP}}(s)$) at $s = 0$ and $s = \infty$.

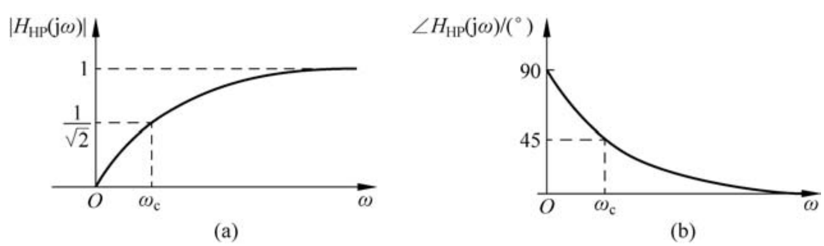

Figure 3.8 Simple high-pass filter responses

(a) amplitude and (b) phase

3.2.3 Simple band-pass filters

These filters pass a band of frequencies called the pass-band and attenuate or eliminate of the frequencies outside the pass-band, called the stop-band. The simplest band-pass filter has a second-order transfer function. The ideal band-pass filters have two cutoff (or 3 dB) frequencies ω_{low} and ω_{high}. These frequencies are defined as those for which the magnitude of the transfer function is equal to max $(1/\sqrt{2})\,|H_{\text{Bp}}(\mathrm{j}\omega)|$. In addition, a new frequency referred as the center or the resonant frequency ω_0, is of interest. It is defined as the frequency at which the transfer function of the circuit $H_{\text{Bp}}(\mathrm{j}\omega)$ is purely real. The center frequency is not in the middle of the pass-band, but the geometric center of the pass-band edges. It is related to the 3 dB frequencies by

$$\omega_0=\sqrt{\omega_{\text{low}}\omega_{\text{high}}}. \tag{3.16}$$

The second parameter of interest is the 3 dB bandwidth given by

$$\beta=\omega_{\text{high}}-\omega_{\text{low}}. \tag{3.17}$$

The third parameter, the quality factor, is the ratio of the center frequency to the 3 dB bandwidth. It is given by

$$Q=\frac{\omega_0}{\omega_{\text{high}}-\omega_{\text{low}}}. \tag{3.18}$$

A second-order function that has the band-pass characteristics is

$$H_{\text{Bp}}(s)=\frac{H_0 as}{s^2+as+b}=\frac{H_0\frac{\omega_0}{Q}s}{s^2+\frac{\omega_0}{Q}s+\omega_0^2}. \tag{3.19}$$

For simplicity the gain constant is assumed to be $H_0=1$ in the following. The transfer function has a zero at the origin ($s=0$) and the infinity ($s=\infty$), indicating that the function goes to zero at $\omega=0$ and at $\omega=\infty$. For $Q>\frac{1}{2}$, it has a pair of complex poles given by

$$s_1,\ s_2=-\frac{\omega_0}{2Q}\pm \mathrm{j}\omega_0\sqrt{1-\frac{1}{4Q^2}}. \tag{3.20}$$

The corresponding transfer function, the amplitude, and phase responses are given by

$$H_{\text{Bp}}(\mathrm{j}\omega)=\frac{\left(\frac{\omega_0}{Q}\right)\omega}{\omega_0^2-\omega^2+\mathrm{j}\left(\frac{\omega_0}{Q}\right)\omega}=\frac{1}{Q\left(\frac{\omega_0}{\omega}-\frac{\omega}{\omega_0}\right)+\mathrm{j}}, \tag{3.21}$$

$$|H_{\text{Bp}}(\mathrm{j}\omega)|=\frac{1}{\sqrt{1+Q^2\left(\frac{\omega}{\omega_0}-\frac{\omega_0}{\omega}\right)^2}}, \tag{3.22}$$

$$\angle H_{\mathrm{Bp}}(\mathrm{j}\omega)=\arctan Q\left(\frac{\omega}{\omega_0}-\frac{\omega_0}{\omega}\right). \tag{3.23}$$

The amplitude and the phase responses are sketched in Fig.3.9 for positive values of ω. From the amplitude response, the peak of the amplitude appears at the center frequency ω_0. The peak magnitude is 1 at $\omega=\omega_0$. The phase angle starts at $(\pi/2)$, crosses the frequency axis at $\omega=\omega_0$, and asymptotically reaches $(-\pi/2)$ as $\omega \to \infty$. Higher the value of Q, the more peaked the amplitude response, and steeper the phase response around $\omega=\omega_0$. The 3 dB or half-power bandwidth can be determined by assuming $\omega_{\text{low}}<\omega_0$ and $\omega_{\text{high}}>\omega_0$. These frequencies can be computed from

$$\left.|H_{\mathrm{Bp}}(\mathrm{j}\omega)|^2\right|_{\omega=\omega_{\text{low}},\omega_{\text{high}}}=\frac{1}{2}\to Q^2\left(\frac{\omega}{\omega_0}-\frac{\omega_0}{\omega}\right)^2=1$$

$$\Rightarrow \omega^2-\omega_0^2=\pm\frac{\omega_0\omega}{Q},\quad \omega^2\mp\frac{1}{Q}\omega\omega_0-\omega_0^2=0.$$

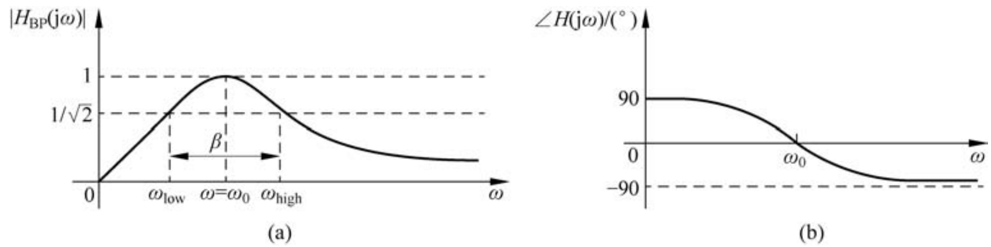

Figure 3.9 (a) Amplitude and (b) phase responses of a band-pass filter

There are four roots of this equation, two for positive and two for negative frequencies.

$$\omega_{\text{high}},\ -\omega_{\text{low}}=\frac{\omega_0}{2Q}\pm\frac{1}{2}\sqrt{\left(\frac{\omega_0}{Q}\right)^2+4\omega_0^2},$$

$$-\omega_{\text{high}},\ \omega_{\text{low}}=-\frac{\omega_0}{2Q}\mp\frac{1}{2}\sqrt{\left(\frac{\omega_0}{Q}\right)^2+4\omega_0^2}.$$

Assuming that $Q>1/2$, the positive roots are given by

$$\omega_{\text{low}},\ \omega_{\text{high}}=\omega_0\sqrt{1+\frac{1}{4Q^2}}\mp\frac{\omega_0}{2Q}. \tag{3.24}$$

The 3 dB bandwidth (B or β) and ω_0 are respectively given by

$$\beta=\omega_{\text{high}}-\omega_{\text{low}}=\frac{\omega_0}{Q},$$

$$\omega_0^2=\omega_{\text{low}}\omega_{\text{high}}\to\omega_0=\sqrt{\omega_{\text{low}}\omega_{\text{high}}}. \tag{3.25}$$

Clearly from Eq.(3.25) the bandwidth is inversely proportional to the value of Q. That is, the bandwidth decreases as Q increases and vice versa. The filter is assumed to be narrowband if ω_0 is very large compared to the bandwidth of the filter. As a rough measure, we assume the filter is a narrowband filter if