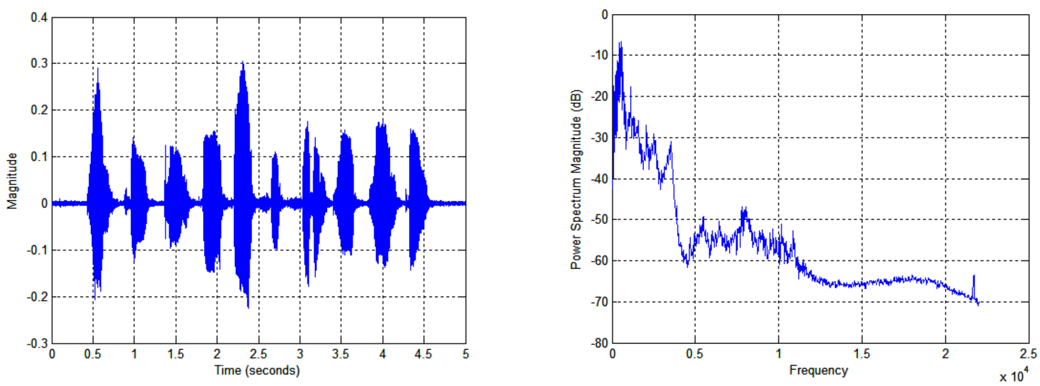
Baseband vs. Passband Communication Systems

Communication systems can be classified into two groups depending on the range of frequencies they use to transmit information. These communication systems are classified into BASEBAND or PASSBAND system. Baseband transmission sends the information signal as it is without modulation (without frequency shifting) while passband transmission shifts the signal to be transmitted in frequency to a higher frequency and then transmits it, where at the receiver the signal is shifted back to its original frequency.

Almost all sources of information generate baseband signals. Baseband signals are those that have frequencies relatively close to zero such as the human voice (20 Hz – 5 kHz) and the video signal from a TV camera (0 Hz – 5.5 MHz). A plot of an audio signal and its frequency spectrum are shown below, where it is seen that the most of the power of the audio signal is concentrated in the frequency range from (0 – 4 kHz). The telephone system used for homes and offices, for example, may transmit the baseband audio signal as it is when the call is local (from your home to your neighbor's home). However, when the telephone call is a long–distance call that is transmitted via microwave or satellite links, the baseband audio signal becomes unsuitable for transmission and the communication system becomes a passband system. Similarly, transmitting the video signal from your camera to your TV using a wire represents a baseband communication while transmitting that video signal via satellites passband transmission. Therefore, baseband transmission, which is easier than passband transmission, is usually used when communicating over wires, while over–the–air transmission requires passband transmission. Notice that even over wires, the transmission may be passband transmission in specific applications.



(a) (b)

An audio signal in (a) time–domain, and (b) in frequency–domain.

The process of shifting the baseband signal to passband range for transmission is known as MODULATION and the process of shifting the passband signal to baseband frequency range at the receiver is known as DEMODULATION. In modulation, one characteristic or more of a signal (generally a sinusoidal wave) known as the carrier is changed based on the information signal that we wish to transmit. The characteristics of the carrier signal that can be changed are the amplitude, phase, or frequency, which result in Amplitude modulation, Phase modulation, or Frequency modulation.

Types of Amplitude Modulation (AM)

AM is itself divided into different types:

1. **Double Sideband with carrier (we will call it AM)**: This is the most widely used type of AM modulation. In fact, all radio channels in the AM band use this type of modulation.

2. **Double Sideband Suppressed Carrier (DSBSC):** This is the same as the AM modulation above but without the carrier.

3. **Single Sideband (SSB):** In this modulation, only half of the signal of the DSBSC is used.

4. **Vestigial Sideband (VSB):** This is a modification of the SSB to ease the generation and reception of the signal.

Double Sideband Suppressed Carrier (DSBSC)

Assume that we have a message signal $m(t)$ with bandwidth (BW) $2\pi B$ rad/s (or B Hz) that has a FT

$$m(t) \Leftrightarrow M(\omega).$$

Let the signal $c(t)$ be a carrier signal (itself carrying no information at all) that is given by

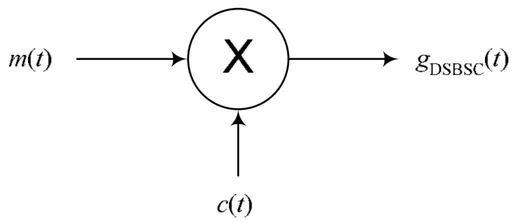
$$c(t) = \cos(\omega_c t),$$

such that the frequency of the carrier ω_c is much larger than the highest frequency in the information signal (we set the amplitude of the carrier to be 1, but it can be any value).

DSBSC Modulation

The DSBSC signal is simply obtained by multiplying the information signal with the carrier signal as shown in the modulator (or transmitter) block diagram shown below

$$g_{\text{DSBSC}}(t) = m(t)\cdot\cos(\omega_c t) \Leftrightarrow (1/2)\,[M(\omega - \omega_c) + M(\omega + \omega_c)].$$

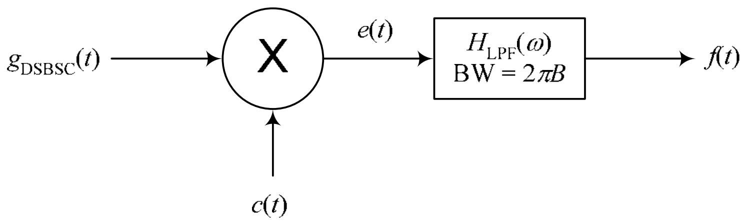


DSBSC Modulator (transmitter)

This signal $g_{\mathrm{DSBSC}}(t)$ is a modulated signal that has its spectrum centered around ω_c and $-\omega_c$. Therefore, this signal becomes a passband signal with frequency that is much larger than the maximum frequency in $m(t)$ and can be transmitted using a relatively short antenna. Also, other similar information signals can be modulated using cosine functions with different frequencies from ω_c and therefore, will not overlap or interfere with this modulated signal when transmitted over the same channel like a air or a coaxial cable.

DSBSC Demodulation

The demodulation process of a DSBSC signal involves obtaining the original information signal or scaled version of it from the modulated signal. This can be done by multiplying the modulated signal with another carrier signal that has EXACTLY the same frequency and phase as the carrier signal in the modulator block as seen in the demodulator block diagram shown below. The amplitude of the two carrier signals in the modulator and demodulator are not important since they just affect the magnitude of the different intermediate signals and final output signal of the demodulator.



DSBSC Demodulator (receiver)

The signal labeled $e(t)$ in the demodulator becomes

$$\begin{aligned} e(t) &= g_{\mathrm{DSBSC}}(t)\cdot\cos(\omega_c t) = m(t)\cdot\cos^2(\omega_c t) = (1/2)\, m(t)\,[1+\cos(2\omega_c t)] \\ &= (1/2)\, m(t) + (1/2)\, m(t)\cos(2\omega_c t) \end{aligned}$$

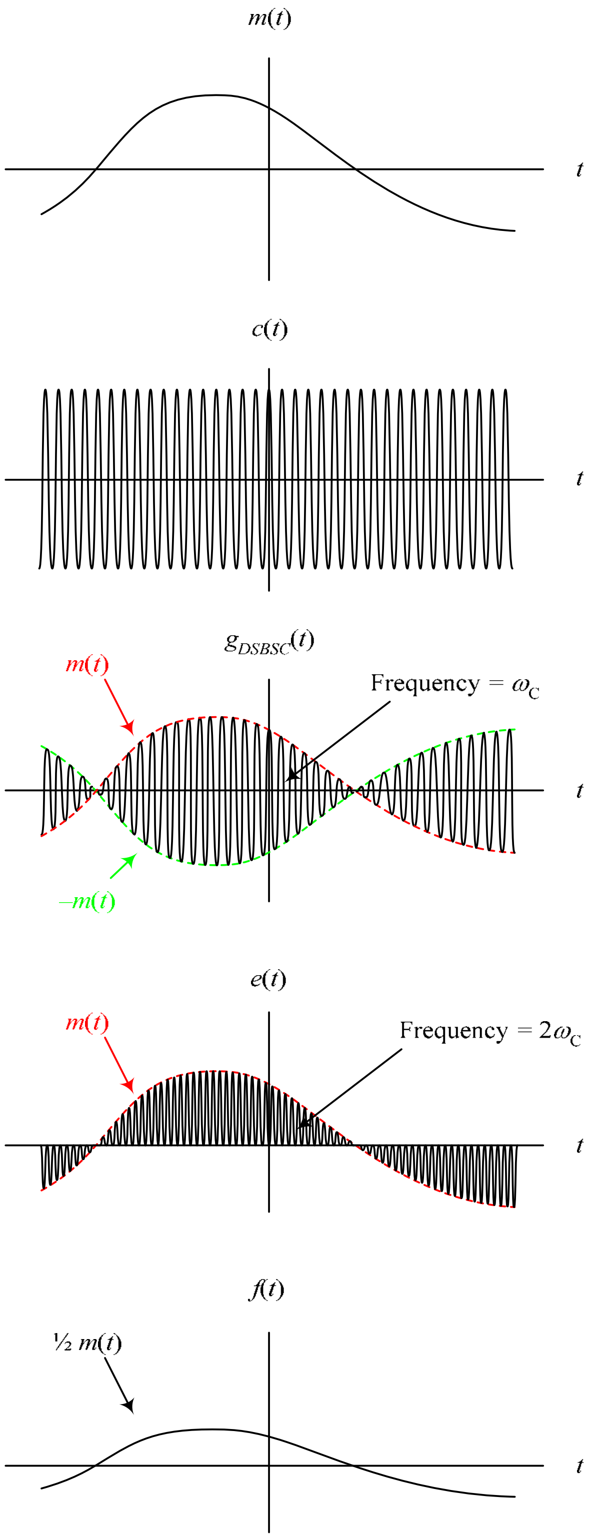
$$\Leftrightarrow\ (1/2)\, M(\omega) + (1/2)\,[M(\omega - 2\omega_c) + M(\omega + 2\omega_c)].$$

However, as seen in the FT of $e(t)$, the original message signal (scaled by 1/2) is present but also other components with frequencies centered around $2\omega_c$ and $-2\omega_c$. These components are undesired and must be removed fop us to get the message signal. This can be done using a LPF (a filter centered around zero frequency that permits low frequencies to pass and rejects high frequencies). The BW of the filter must be $2\pi B$ rad/s (or B Hz) or possibly slightly higher (but not much higher that it will allow the high-frequency components around $2\omega_c$ and $-2\omega_c$ to partially or completely pass).

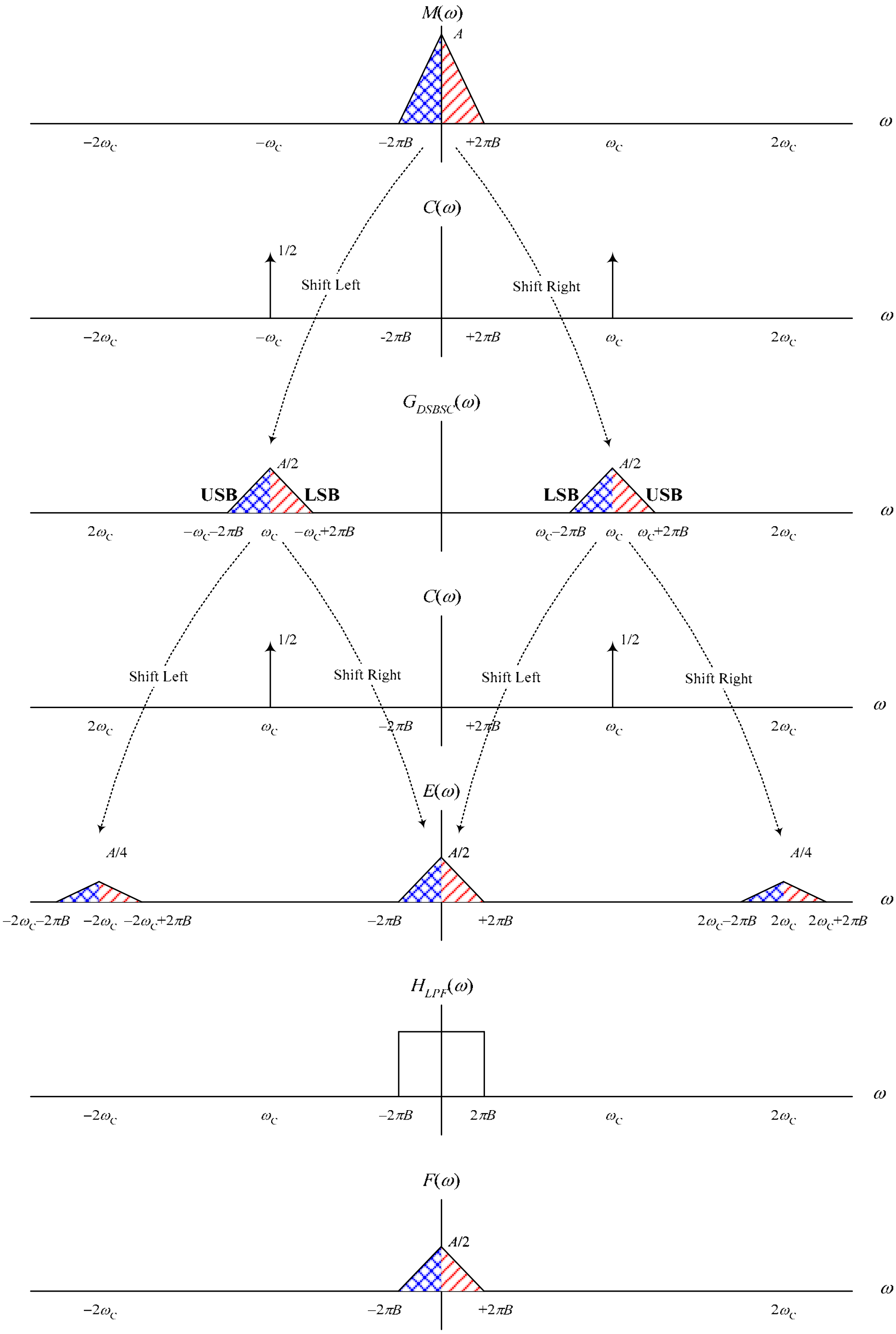
Therefore, the output signal $f(t)$ of the LPF will be

$$e(t) = (1/2)\, m(t) \quad \Leftrightarrow \quad (1/2)\, M(\omega).$$

This is simply a scaled version of the original transmitted signal that can be easily amplified to obtain the original signal exactly.



Time–domain representation of the different signals obtained in the DSBSC modulation–demodulation process.



Frequency–domain representation of the different signals obtained in the DSBSC modulation–demodulation process.