

Chapitre 1

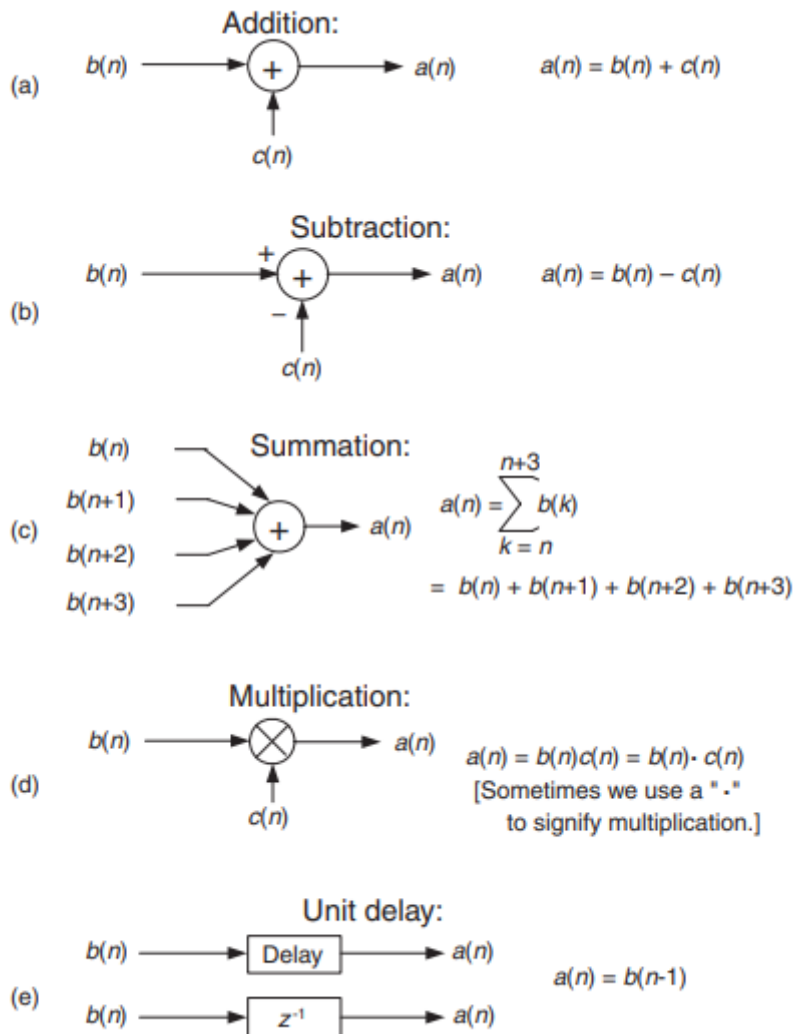
1.1

discrete-time signal : a signal whose independent time variable is quantized so that we know only the value of the signal at discrete instants in time.

Note

If a continuous sinewave represents a physical voltage, we could sample it once every t_s seconds and represent the sinewave as a sequence of discrete values.

1.3



1.4

| LTI : linear time-invariant

1.5

A linear system's output is the sum of the outputs of its parts. Where:

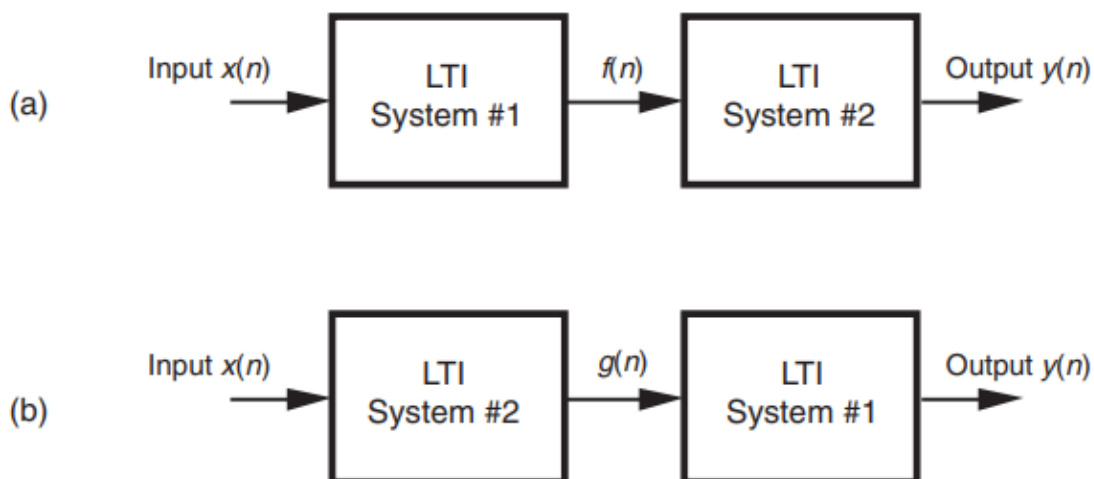
$$x(n)_1 + x(n)_2 \rightarrow y(n)_1 + y(n)_2$$

1.6

A time-invariant system is one where a time delay (or shift) in the input sequence causes an equivalent time delay in the system's output sequence. Keeping in mind that n is just an indexing variable we use to keep track of.

1.7

LTI systems have a useful commutative property by which their sequential order can be rearranged with no change in their final output



Chapitre 2

2.1

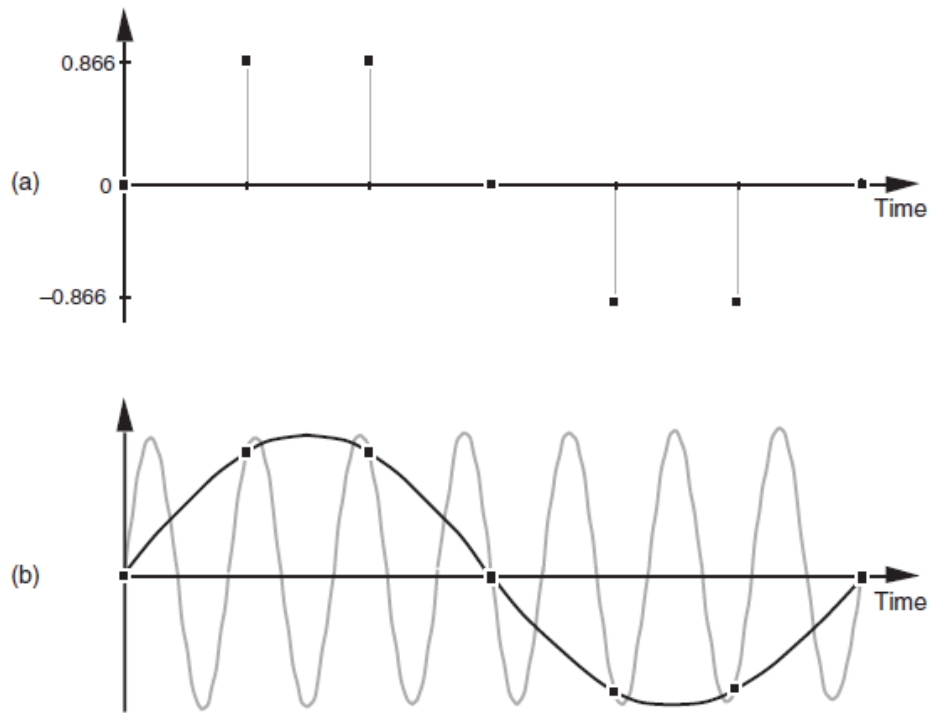


Figure 2-1 Frequency ambiguity: (a) discrete-time sequence of values; (b) two different sinewaves that pass through the points of the discrete sequence.

- An $x(n)$ sequence of digital sample values, representing a sinewave of f_0 Hz, also exactly represents sinewaves at other frequencies, namely, $f_0 + k f_s$.

So you can write:

$$x(n) = \sin(2\pi f_0 n t_0) = \sin(2\pi (f_0 + k F_s) n t_0) - \text{Eq. (2-5)}$$

🔍 Formally >

When sampling at a rate of f_s samples/second, if k is any positive or negative integer, we cannot distinguish between the sampled values of a sinewave of f_0 Hz and a sinewave of $(f_0 + k f_s)$ Hz.

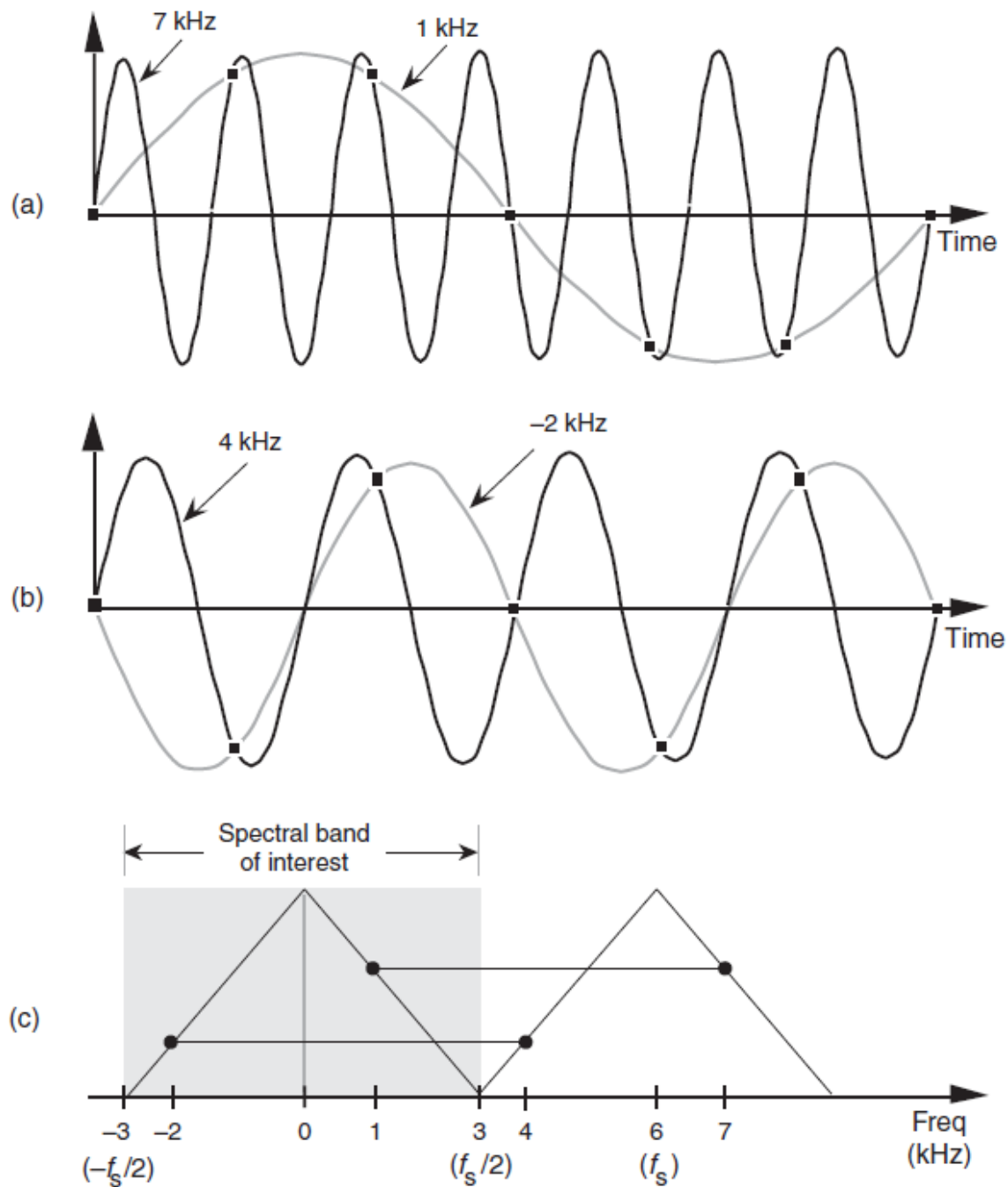


Figure 2-2 Frequency ambiguity effects of Eq. (2-5): (a) sampling a 7 kHz sinewave at a sample rate of 6 kHz; (b) sampling a 4 kHz sinewave at a sample rate of 6 kHz; (c) spectral relationships showing aliasing of the 7 and 4 kHz sinewaves.

-Figure 2-2(b) shows another example of frequency ambiguity that we'll call aliasing, where a 4 kHz sinewave could be mistaken for a -2 kHz sinewave. In Figure 2-2(b), $f_o = 4$ kHz, $f_s = 6$ kHz, and $k = -1$ in Eq. (2-5), so that $f_o + k f_s = [4 + (-1 \cdot 6)] = -2$ kHz

$\frac{f_s}{2}$ is an important quantity in sampling theory and is referred to by different names in the literature, such as critical Nyquist, half Nyquist, and folding frequency.

2.2

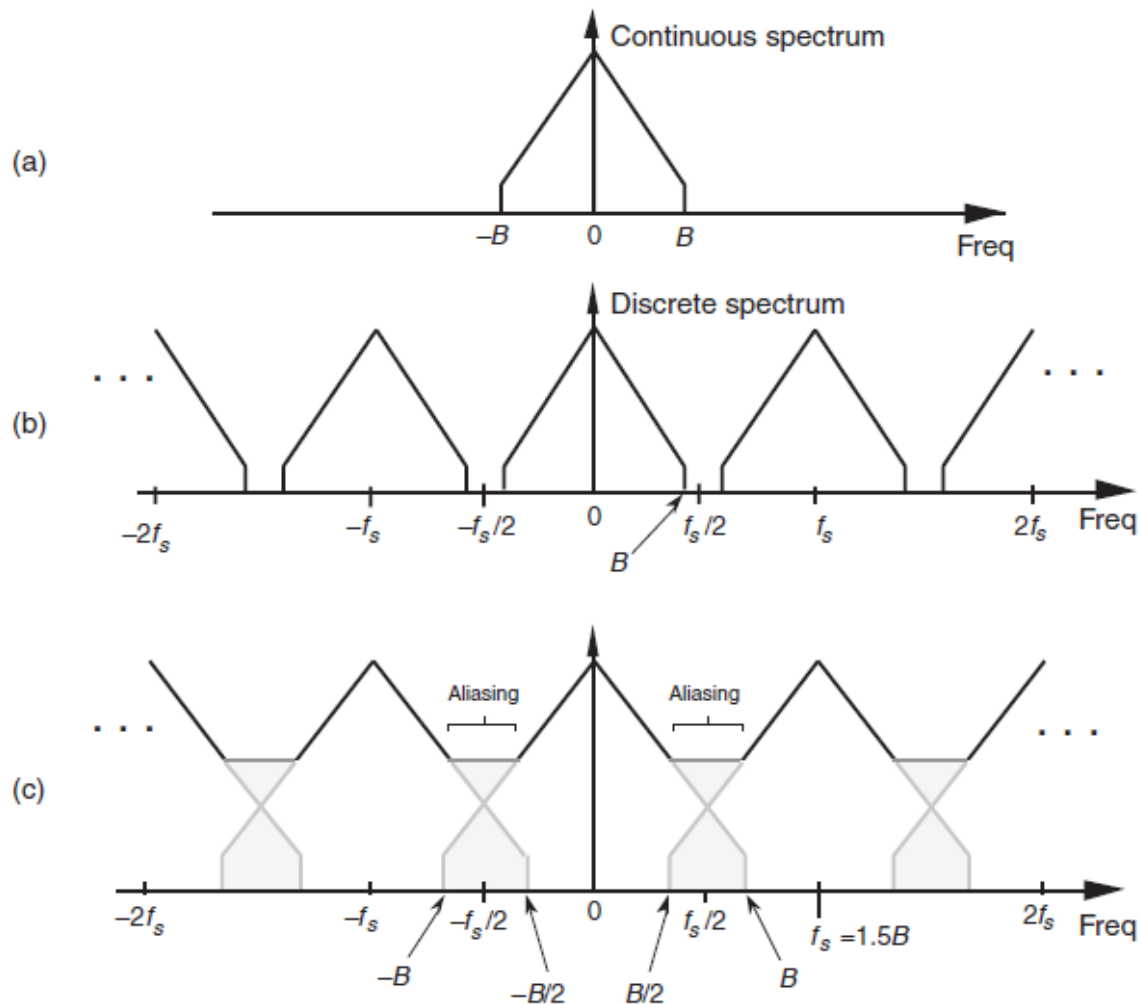


Figure 2-4 Spectral replications: (a) original continuous lowpass signal spectrum; (b) spectral replications of the sampled lowpass signal when $f_s/2 > B$; (c) frequency overlap and aliasing when the sampling rate is too low because $f_s/2 < B$.

-Given that the continuous $x(t)$ signal, whose spectrum is shown in Figure 2-4(a), is sampled at a rate of f_s samples/second, we can see the spectral replication effects of sampling in Figure 2-4(b) showing the original spectrum in addition to an infinite number of replications. To illustrate why the term folding frequency is used, let's lower our sampling frequency to $f_s = 1.5B$ Hz. The spectral result of this undersampling is illustrated in Figure 2-4(c)

Chapitre 3

The DFT is a mathematical procedure used to determine the harmonic, or frequency, content of a discrete signal sequence. The DFT is useful in analyzing any discrete sequence regardless of what that sequence actually represents.

-The DFT's origin, of course, is the continuous Fourier

transform $X(f)$ defined as :

$$X(f) = \int_{-\infty}^{\infty} x(t)e^{-j\pi ft} dt$$