EMC Course Notes 2024

Signal Spectra

Time and Frequency Domains

Electrical signals have both time and frequency domain representations. In the time domain, voltage or current is expressed as a function of time as illustrated in Figure 4.1. Most people are relatively comfortable with time domain representations of signals. Signals measured on an oscilloscope are displayed in the time domain and digital information is often conveyed by a voltage as a function of time.

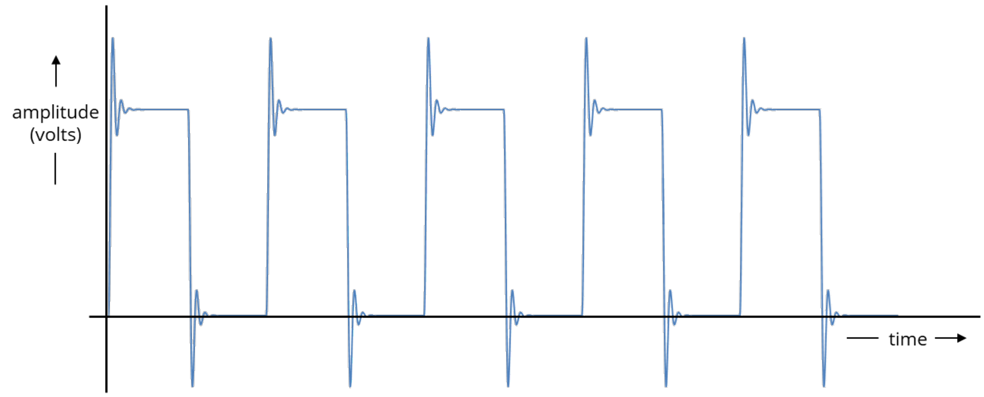


Figure 4.1. Time domain representation of an electrical signal.

Signals can also be represented by a magnitude and phase as a function of frequency. Signals that repeat periodically in time are represented by a line spectrum as illustrated in Figure 4.2. The line spectrum has a DC component at 0 Hz, a fundamental component at $1/T$, and harmonics at n/T (where n is an integer). This representation is also referred to as a *power spectrum* because the sum of the powers in each harmonic equals the time-average power in the time-domain signal.

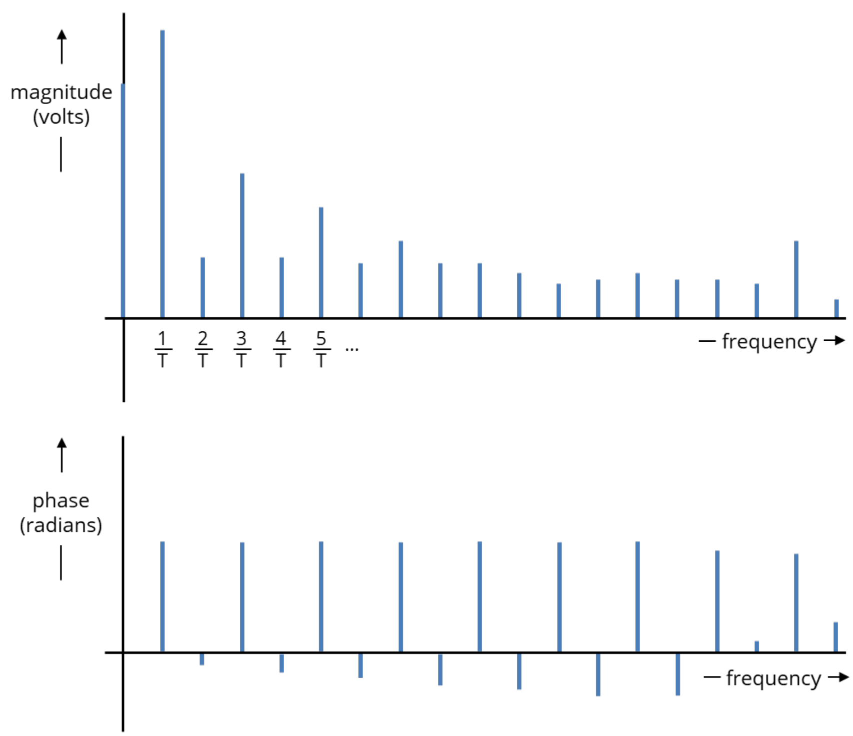


Figure 4.2. Frequency domain representation of a periodic signal.

Signals that are time limited (i.e., are only non-zero for a finite time) are represented by a continuous spectrum as illustrated in Figure 4.3. This representation is also referred to as an *energy spectrum* because the integral of the energy density in this waveform over frequency equals the total energy in the time-domain signal.

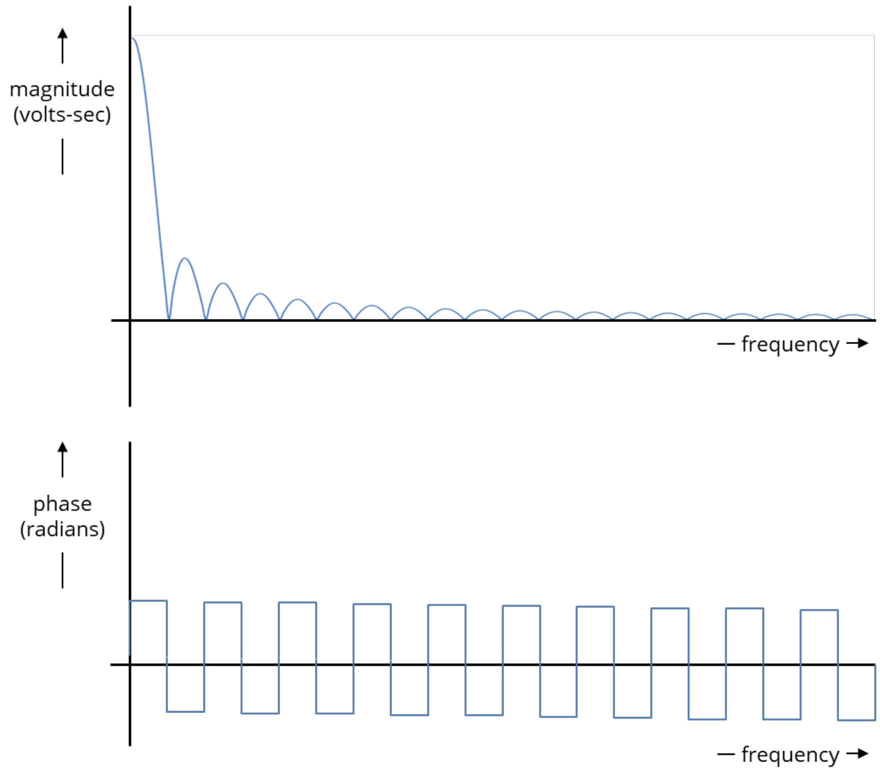


Figure 4.3. Frequency domain representation of a time-limited (transient) signal.

Frequency domain representations are particularly useful when analyzing linear systems. EMC and signal integrity engineers must be able to work with signals represented in both the time and frequency domains. Signal sources and interference are often defined in the time domain. However, system behavior and signal transformations are more convenient and intuitive when working in the frequency domain.

Linear Systems

Linear system theory plays a key role in the engineering analysis of electrical and mechanical systems. Engineers model a wide variety of things as linear transformations including circuit behavior, signal propagation, coupling and radiation. Therefore, it is important to review exactly what we mean by a *linear system* so that we recognize how and when to take advantage of the powerful linear system analysis tools available to us.

Figure 4.4 illustrates a system with one input, $x(t)$, and one output, $y(t)=H\{x(t)\}$. If an input, $x_1(t)$ produces an output $y_1(t)$, and an input $x_2(t)$ produces an output $y_2(t)$, then the system is linear if and only if,

$$ay_1(t)+by_2(t)=H\left\{ax_1(t)+bx_2(t)\right\} \tag{4.1}$$

where *a* and *b* are constants. In other words, scaling the input by a constant will produce an output scaled by the same constant, and combining (summing) two inputs will produce an output that is the sum of the outputs produced by the individual inputs.

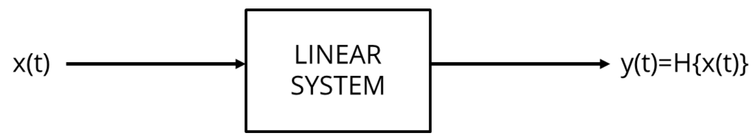


Figure 4.4. A linear system with input x(t) and output y(t).

Quiz Question

Which of the following equations describes the relationship between the output y(t) and the input x(t) of a linear system?

a.) $y=5x$

b.) $y(t)=0$

c.) $y=8x+3$

d.) $y=x^2$

e.) $y(t)=5t\,x(t)$

f.) $y=\sin x$

g.) $y(t)=5\frac{\partial}{\partial t}\left[x(t)\right]$

Of the choices above, only *a, b* and *g* are linear system transformations. *y(t)=0* is not a very interesting system, because its output is always zero, but it is linear. Simple derivative and integral operators are linear because they satisfy the conditions in Equation (4.1). The remaining choices are not linear operations. Note that *y=8x+3* is the equation of a straight line, but it does not describe a linear system because it has a non-zero output when there is no input.

At first, it may appear that very few real electrical or mechanical systems of interest behave this way. However, many non-linear systems can be approximated as linear over some small change of the input. Most engineering analysis depends on modeling real devices and circuits as linear systems.

Frequency Domain Analysis of Linear Systems

Linear systems have the unique property that any sinusoidal input will produce a sinusoidal output at exactly the same frequency. In other words, if the input is of the form,

$$x(t)=A_{in}\cos\left(\omega_0 t+\varphi_{in}\right), \tag{4.2}$$

then the output will have the form,

$$y(t) = A_{out} \cos\left(\omega_0 t + \varphi_{out}\right). \tag{4.3}$$

In general, the magnitude and phase of the sinusoidal signal may change but the frequency must be constant. This provides us with a very powerful analysis tool for analyzing linear systems. If we represent an input signal as the sum of its components in the frequency domain, then we can express the output as a simple scaling of the magnitudes and shifting of the phases of these components.

Phasor Notation

To facilitate the analysis of linear system responses to sinusoidal inputs, it is convenient to represent signals in an abbreviated form known as *phasor notation*. Consider an input of the form,

$$x(t) = A\cos\left(\omega t + \varphi\right). \tag{4.4}$$

This can be represented as,

$$\begin{aligned} x(t) &= \mathrm{Re}\left\{ A e^{j(\omega t + \varphi)} \right\} \\ &= A \cdot \mathrm{Re}\left\{ e^{j\omega t} e^{j\varphi} \right\} \end{aligned} \tag{4.5}$$

where $\mathrm{Re}\{\bullet\}$ indicates the real part of a complex quantity. Recognizing that the frequency ω will be the same throughout the system, we don't need to specifically write the term $e^{j\omega t}$ as long as we remember that it's there. The same applies to the $\mathrm{Re}\{\bullet\}$ notation. This allows us to express a sinusoidal signal simply in terms of its magnitude and phase as,

$$x = Ae^{j\phi} \quad or \quad A\angle\phi. \tag{4.6}$$

The expression in (4.6) is the signal in (4.4) expressed using phasor notation. Note that we must know the frequency of a signal in order to convert from phasor notation to the time domain representation.

Quiz Question

Write the following signals using phasor notation:

a.) $x(t) = 5\cos(\omega t)$ volts

b.) $y(t)=5\sin(\omega t)$ amps

c.) $z(t) = 5t\sin(\omega t)$ volts

The first signal expressed in phasor notation is simply $x = 5$ volts. To obtain the phasor notation for the second signal, we recognize that $\sin(\omega t) = \cos(\omega t + \pi/2)$ so $y = 5e^{j(\pi/2)}$.

The third signal is not a sinusoid and therefore cannot be expressed using phasor notation.

Fourier Series

Of course, many of the inputs to linear systems we would like to analyze are not sinusoidal. In this case, it is desirable to represent arbitrary signal waveforms as a sum of sinusoidal frequency components. In the frequency domain, each component can be analyzed individually. The frequency domain system outputs can then be summed and converted back to the time domain.

A periodic signal can be represented as a sum of its frequency components by calculating its *Fourier series* coefficients. A periodic signal with period T can be written,

$$x(t) = \sum_{n=-\infty}^{\infty} c_n e^{jn2\pi f_0 t} \tag{4.7a}$$

where

$$c_n = \frac{1}{T}\int_{t_0}^{t_0+T} x(t) e^{-jn2\pi f_0 t} dt\ . \tag{4.7b}$$

If $x(t)$ is a real time domain signal, the coefficients c_n and c_{-n} are complex conjugates (i.e., $c_{-n} = c_n^*$) and we can rewrite Eq. (4.7a) in the form,

$$\begin{aligned} x(t) &= c_0 + \sum_{n=1}^{\infty}\left(c_n e^{jn2\pi f_0 t} + c_n^* e^{-jn2\pi f_0 t}\right) \\ &= c_0 + \sum_{n=1}^{\infty}\left(\left|c_n\right| e^{jn2\pi f_0 t+\phi_n} + \left|c_n\right| e^{-\left(jn2\pi f_0 t+\phi_n\right)}\right) \\ &= c_0 + \sum_{n=1}^{\infty} 2\left|c_n\right| \cos\left(n2\pi f_0 t + \phi_n\right). \end{aligned} \tag{4.8}$$

In this form, we see that the Fourier series coefficients consist of a DC component, c_0, and positive harmonic frequencies, $n2\pi f_0$ (n = 1,2,3, …). This is the *one-sided Fourier series* and the coefficients, $2\left|c_n\right|$, represent the peak value of each harmonic. Dividing the peak value by $\sqrt{2}$ yields the root-mean-square (rms) value. Signal harmonics measured on a spectrum analyzer or EMI test receiver are the rms values of the one-sided Fourier Series coefficients. In other words, the amplitude of each measured harmonic is $\sqrt{2}\left|c_n\right|$.

The frequency domain representation of a periodic signal is a line spectrum. It can only have non-zero values at DC, the fundamental frequency, and harmonics of the fundamental. Because periodic signals have no beginning or end, non-zero periodic signals have infinite energy but finite power. The total power in the time domain signal,

$$P_{total} = \frac{1}{T}\int_{t_0}^{t_0+T} x^2(t)\, dt \tag{4.9}$$

is equal to the sum of the power in each frequency domain component,

$$P_{total} = \sum_{n=-\infty}^{\infty} \left| c_n \right|^2 . \tag{4.10}$$

A few periodic signals and their frequency domain representations are illustrated in Figure 4.5.

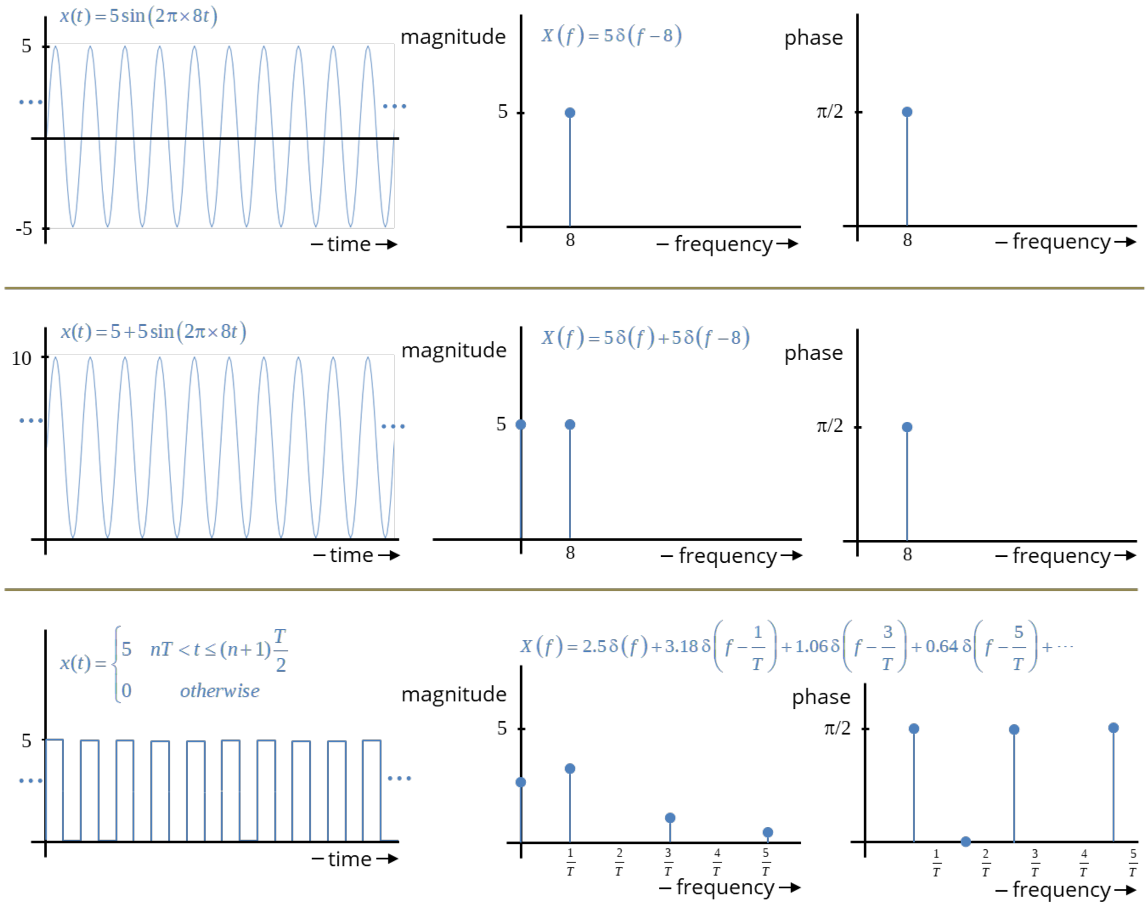
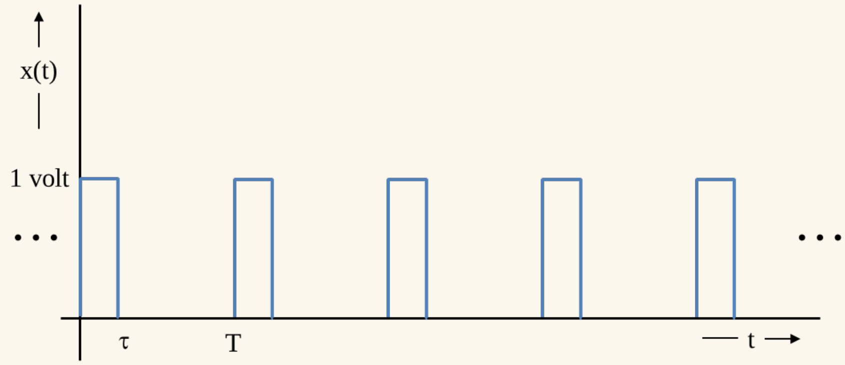


Figure 4.5. Periodic signals in the time and frequency domain.

Example 4-1: Frequency Domain Representation of a Pulse Train

Determine the frequency domain representation for the pulse train shown in the figure below.



In the time domain this signal is described by the following formula:

$$x(t) = \begin{cases} 1\text{ V} & nT < t < nT + \tau \\ 0 & otherwise \end{cases} \qquad n = \pm 1, \pm 2, \pm 3, \cdots.$$

The coefficients of the Fourier series are then calculated using Eq. (4.7b) as,

$$\begin{aligned} c_n &= \frac{1}{T}\int_0^T x(t)\, e^{-jn2\pi f_0 t} dt \\ &= \frac{1}{T}\int_0^\tau (A)\, e^{-jn2\pi t/T} dt \\ &= \frac{A}{T}\int_0^\tau e^{-jn2\pi t/T}\, dt \\ &= \frac{A\tau}{T}\left[\frac{\sin\left(n\pi\tau/T\right)}{\left(n\pi\tau/T\right)}\right] e^{-j\left(n\pi\tau/T\right)} \end{aligned}.$$

Note that as $\tau \to 0$, our time domain signal looks like an impulse train and the amplitudes of all the harmonics approach the same value. As $\tau \to T/2$, the signal becomes a square wave and the magnitude of the harmonics becomes,

$$c_n = \frac{A}{2}\left|\frac{\sin\left(n\pi/2\right)}{\left(n\pi/2\right)}\right|\left|e^{-j\left(n\pi/2\right)}\right| = \begin{cases} \dfrac{A}{n\pi} & n = \pm 1, \pm 3, \pm 5 \cdots \\ 0 & n = \pm 2, \pm 4, \pm 6 \cdots \end{cases}.$$

In this case, the amplitude of the even harmonics is zero and the odd harmonics decrease linearly with frequency (n).

Note that if we wanted to determine the amplitude of the harmonics as measured on a spectrum analyzer, we would calculate the rms amplitude of the one-sided Fourier Series coefficients,

$$\sqrt{2}\left|c_n\right| = \frac{\sqrt{2}A\tau}{T}\left[\frac{\sin\left(n\pi\tau/T\right)}{\left(n\pi\tau/T\right)}\right] \qquad n = 1, 2, 3, \cdots$$

Fourier Transform

Transient signals (i.e., signals that start and end at specific times) can also be represented in the frequency domain using the *Fourier transform*. The Fourier transform representation of a transient signal, *x(t)*, is given by,

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi ft} dt \, . \tag{4.11}$$

The inverse Fourier transform can be used to convert the frequency domain representation of a signal back to the time domain,

$$x(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} X(f) e^{j2\pi ft} df \, . \tag{4.12}$$

Two transient time domain signals and their Fourier transforms are illustrated in Figure 4.6.

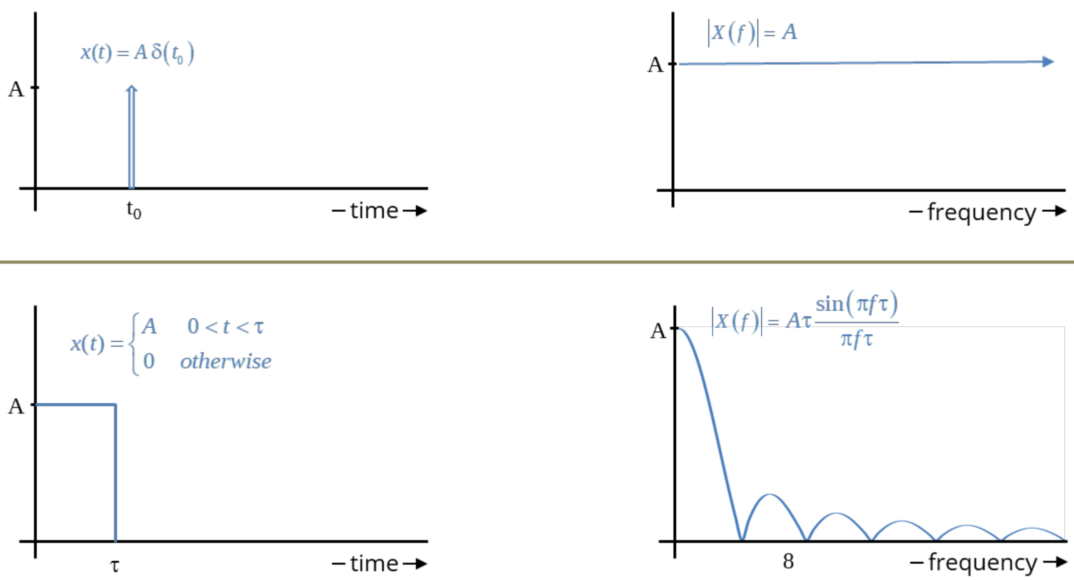


Figure 4.6. Transient signals in the time and frequency domain.

Note that transient signals have zero average power (when averaged over all time), but they have finite energy. The total energy in a transient time domain signal is given by,

$$\mathrm{E} = \int_{-\infty}^{\infty} x^2(t)\, dt \, . \tag{4.13}$$

This must equal the total energy in the frequency domain representation of the signal,

$$\mathrm{E} = \int_{-\infty}^{\infty} \left| X(f) \right|^2 df \, . \tag{4.14}$$

Frequency Domain Representation of a Trapezoidal Signal

Let's examine the frequency domain representation of the periodic trapezoidal waveform illustrated in Figure 4.7. Examining the behavior of this waveform helps us to gain insight into the relationship between time and frequency domain representations in general. Also, the similarity between the trapezoidal waveform and common digital signal waveforms will be useful when we investigate EMC or signal integrity problems with digital systems.

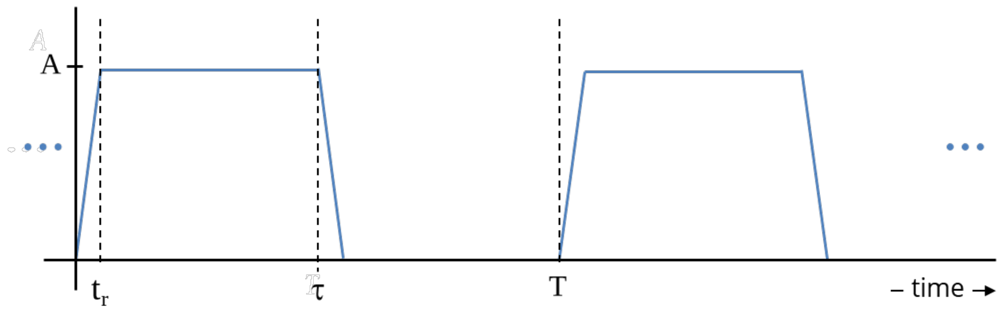


Figure 4.7. Trapezoidal waveform.

Using the one-sided Fourier series, Eq. (4.7b) and (4.8), we can represent this signal as the sum of its frequency components,

$$x(t) = c_0 + \sum_{n=1}^{\infty} 2\left|c_n\right| \cos\left(n2\pi f_0 t + \phi_n\right) \tag{4.15}$$

where

$$2\left|c_n\right| = \frac{2A\tau}{T}\left|\frac{\sin\left(n\pi\tau/T\right)}{\left(n\pi\tau/T\right)}\right|\left|\frac{\sin\left(n\pi t_r/T\right)}{\left(n\pi t_r/T\right)}\right|. \tag{4.16}$$

Equation (4.16) can be derived by noting that the trapezoidal waveform in Figure 4.7 can be obtained by convolving the pulse train in Example 4-1 with another pulse train whose pulses have a width, t_r, and an amplitude A/t_r. Convolution in the time domain is equivalent to multiplication in the frequency domain, so we can simply multiply the two frequency domain representations of these pulse trains to obtain Eq. (4.16).

Each term, $2|c_n|$, is the peak amplitude of the nth harmonic. If we assume that $t_r << T$, we note that the third term is approximately equal to $\frac{\sin(\text{small number})}{\text{small number}} \approx 1$ for the lower harmonics. If $\tau = T/2$ (i.e., a 50% duty cycle), then the numerator of the second term is 1 for the harmonics (n = 1,3,5…) and 0 for the even harmonics (n = 2,4,6…). The amplitude of the lower harmonics is then inversely proportional to n (i.e., the amplitude of the lower harmonics decreases proportional to the frequency). At higher harmonics, the third term also begins to decrease proportional to frequency, so the overall amplitude of the upper harmonics decreases on average at a rate proportional to the square of the frequency. This frequency representation of a trapezoidal signal $\left(\tau = T/2, t_r \ll T\right)$ and its envelope are illustrated in Figure 4.8. Note that small values of τ (short duty cycles) will extend the first knee frequency, which could cause the first several harmonics to have approximately the same amplitude.

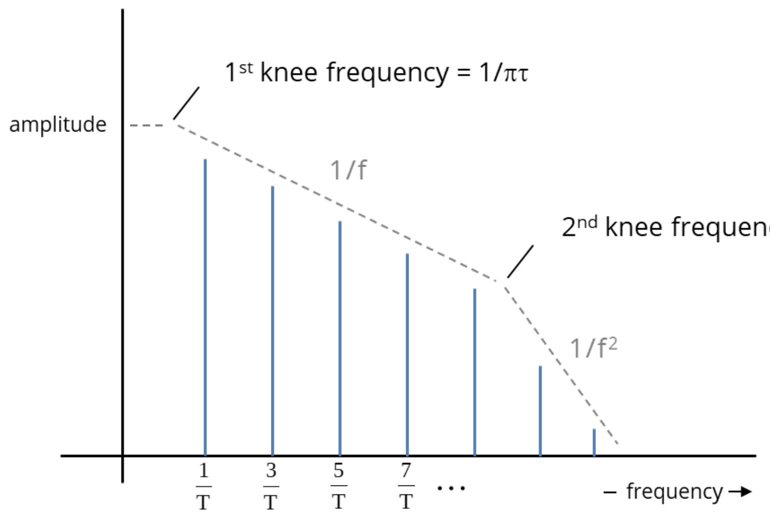


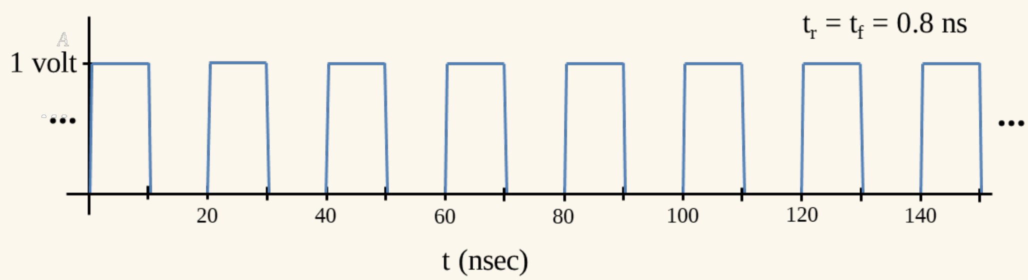
Figure 4.8. Frequency Domain representation of a trapezoidal signal $\left(\tau = {}^{T}\!/_{2}, t_r \ll T\right)$.

Example 4-2: Harmonics of a trapezoidal signal

The waveform shown below is measured on an oscilloscope in the lab. The rise and fall times are 0.8 ns.

a. *What is the fundamental frequency?*
b. *Calculate the amplitudes of the harmonics at 50 MHz, 150 MHz, 250 MHz, and 1.55 GHz that would be measured on a spectrum analyzer.*

If the rise and fall times are increased to 1.6 nanoseconds, then by how many dB will the harmonics at 50 MHz, 150 MHz, 250 MHz, and 550 MHz be reduced?



Noting that the period is 20 nsec, the fundamental frequency is easily determined to be,

$$f_0 = \frac{1}{T} = \frac{1}{2\times 10^{-8}} = 50\,\text{MHz}.$$

Therefore, we are being asked to determine the amplitudes of the 1st, 3rd, 5th, and 11th harmonics. Applying Equation (4.16) for n = 1, 3, 5 and 11, and dividing by $\sqrt{2}$ to get the rms value measured on a spectrum analyzer, yields the following harmonic amplitudes,

$$\sqrt{2}\left|c_1\right| = \frac{(1\text{ V})}{\sqrt{2}}\left|\frac{\sin\left(1\pi(10)/20\right)}{\left(1\pi(10)/20\right)}\right|\left|\frac{\sin\left(1\pi(0.8)/20\right)}{\left(1\pi(0.8)/20\right)}\right| = (0.707\text{ V})(0.637)(0.997) = 0.449\text{ V}$$

$$\sqrt{2}\left|c_3\right| = \frac{(1\text{ V})}{\sqrt{2}}\left|\frac{\sin\left(3\pi(10)/20\right)}{\left(3\pi(10)/20\right)}\right|\left|\frac{\sin\left(3\pi(0.8)/20\right)}{\left(3\pi(0.8)/20\right)}\right| = (0.707\text{ V})(0.212)(0.976) = 0.146\text{ V}$$

$$\sqrt{2}\left|c_5\right| = \frac{(1\text{ V})}{\sqrt{2}}\left|\frac{\sin\left(5\pi(10)/20\right)}{\left(5\pi(10)/20\right)}\right|\left|\frac{\sin\left(5\pi(0.8)/20\right)}{\left(5\pi(0.8)/20\right)}\right| = (0.707\text{ V})(0.127)(0.935) = 0.084\text{ V}$$

$$\sqrt{2}\left|c_{11}\right| = \frac{(1\text{ V})}{\sqrt{2}}\left|\frac{\sin\left(11\pi(10)/20\right)}{\left(11\pi(10)/20\right)}\right|\left|\frac{\sin\left(11\pi(0.8)/20\right)}{\left(11\pi(0.8)/20\right)}\right| = (0.707\text{ V})(0.058)(0.711) = 0.029\text{ V}.$$

None of these harmonics are significantly affected by the risetime. They have virtually the same amplitude that they would have had if the risetime had been zero. Increasing the risetime to 1.6 nsec, however, significantly affects the amplitude of the upper harmonics,

$$\sqrt{2}\left|c_1\right| = \frac{(1\text{ V})}{\sqrt{2}}\left|\frac{\sin\left(1\pi(10)/20\right)}{\left(1\pi(10)/20\right)}\right|\left|\frac{\sin\left(1\pi(1.6)/20\right)}{\left(1\pi(1.6)/20\right)}\right| = (0.707\text{ V})(0.637)(.990) = 0.446\text{ V}$$

$$\sqrt{2}\left|c_3\right| = \frac{(1\text{ V})}{\sqrt{2}}\left|\frac{\sin\left(3\pi(10)/20\right)}{\left(3\pi(10)/20\right)}\right|\left|\frac{\sin\left(3\pi(1.6)/20\right)}{\left(3\pi(1.6)/20\right)}\right| = (0.707\text{ V})(0.212)(0.908) = 0.136\text{ V}$$

$$\sqrt{2}\left|c_5\right| = \frac{(1\text{ V})}{\sqrt{2}}\left|\frac{\sin\left(5\pi(10)/20\right)}{\left(5\pi(10)/20\right)}\right|\left|\frac{\sin\left(5\pi(1.6)/20\right)}{\left(5\pi(1.6)/20\right)}\right| = (0.707\text{ V})(0.127)(0.757) = 0.068\text{ V}$$

$$\sqrt{2}\left|c_{11}\right| = \frac{(1\text{ V})}{\sqrt{2}}\left|\frac{\sin\left(11\pi(10)/20\right)}{\left(11\pi(10)/20\right)}\right|\left|\frac{\sin\left(11\pi(1.6)/20\right)}{\left(11\pi(1.6)/20\right)}\right| = (0.707\text{ V})(0.058)(0.133) = 0.005\text{ V}.$$

Doubling the risetime from 0.8 to 1.6 nsec reduces the first harmonic by only $20\log\left(\frac{0.449}{0.446}\right) = 0.06\text{ dB}$. The third harmonic is reduced by $20\log\left(\frac{0.146}{0.136}\right) = 0.62\text{ dB}$. The fifth harmonic is reduced by $20\log\left(\frac{0.084}{0.068}\right) = 1.8\text{ dB}$, while the eleventh harmonic is reduced by $20\log\left(\frac{0.029}{0.005}\right) = 15\text{ dB}$.

Note that changing the risetime can have a significant effect on the amplitude of the upper harmonics without changing the time domain representation of the signal significantly. Radiated EMI or crosstalk problems at the upper harmonic frequencies of a digital signal can often be solved by increasing the risetime of the digital signal waveform. Generally, a risetime that is equal to 10% of a bit length or more will still produce a very good digital signal while significantly limiting the amplitude of a signal at frequencies above the 10^{th} harmonic.

Spectrum Analysis

Spectrum Analysis – Using a Traditional Spectrum Analyzer

A traditional spectrum analyzer displays time-domain signals in the frequency domain. As illustrated in Figure 4.9, the input signal is mixed with (multiplied by) another sinusoidal signal generated by a voltage-controlled oscillator. The resulting signal has a frequency-domain representation that is the same as the original signal but shifted both up and down by a frequency equal to that of the voltage-controlled oscillator. A band-pass filter picks up the power in a given narrow band of frequencies as the voltage-controlled oscillator changes the amount of the shift with time. A plot of the power out of the band-pass filter vs the amount of frequency shift provides a frequency-domain representation of the original signal.

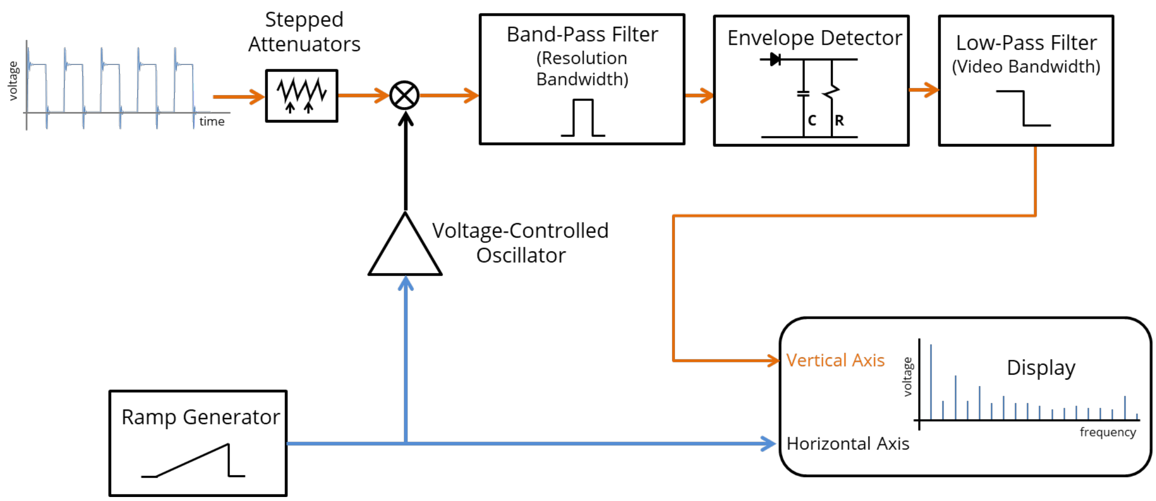


Figure 4.9. Basic operation of a traditional spectrum analyzer.

A key parameter of this measurement is the bandwidth of the band-pass filter, called the *resolution bandwidth*, which determines how finely individual frequencies can be resolved in the output.

Traditional spectrum analyzers are capable of making accurate measurements over a wide range of frequencies, but also have the ability to focus on specific narrow frequency bands to make high-resolution measurements.

Spectrum Analysis – Using a Real-Time Spectrum Analyzer

A real-time spectrum analyzer also typically employs a mixer to down-convert the received signal to a lower frequency. As illustrated in Figure 4.10, the converted signal is then sampled using an analog-to-digital converter, and a Fast Fourier Transform (FFT) algorithm is then used to convert the sampled time domain signal to a sampled frequency domain signal.

Overlapping time sequences are converted to the frequency domain, then digitally combined and displayed. The signal spectrum is continuously updated, allowing rapid changes in the signal spectrum to be monitored.

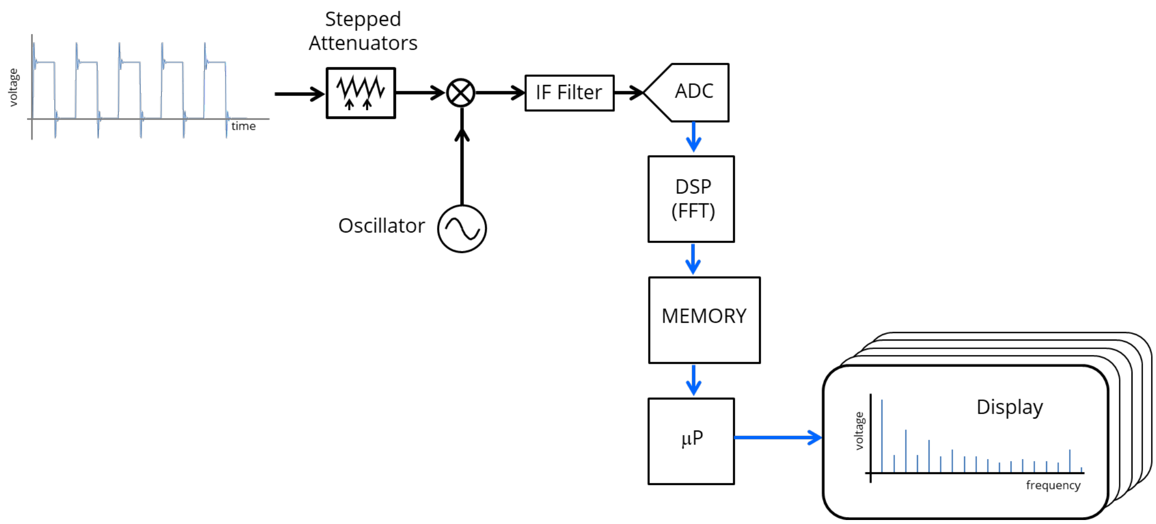


Figure 4.10. Basic operation of a real-time spectrum analyzer.

A significant advantage of time-domain spectrum analyzers is that they capture the entire signal content in a given frequency band all of the time. This allows the analyzer to capture brief transient events that might be missed by a traditional spectrum analyzer, which is only looking at one narrow frequency band at any given instant.

Spectrum Analysis – Using a Digital Oscilloscope

Digital oscilloscopes with an FFT function can also display signals in the frequency domain. Modern scopes with sophisticated signal processing algorithms can duplicate many of the functions typically associated with spectrum analyzers. As indicated in Figure 4.11, digital oscilloscopes do not down-convert the received signal and are slightly more limited in their ability to display results in specific frequency ranges. On the other hand, a significant advantage of most digital oscilloscopes is their ability to process two input signals on separate channels simultaneously. This provides an ability to add or subtract signals (e.g., to determine common-mode and differential-mode components), or to quantify the correlation between two signals.

Digital oscilloscopes tend to give the user a lot of control over the sampling parameters. It's important for the user to understand how changes in one parameter affect other important variables. Some of these relationships are listed in Table 4.1 below.

Table 4.1. Key Parameters for Time-Frequency Conversion using an FFT.

Time Domain	Frequency Domain
Sampling Rate: f_s in samples/second	Bandwidth (or frequency range): BW = $1/f_s$
Number of time-domain samples: N	Number of frequency-domain samples: N
Sample Period: T = N/f_s	Frequency resolution (Δf): BW/N = 1/T

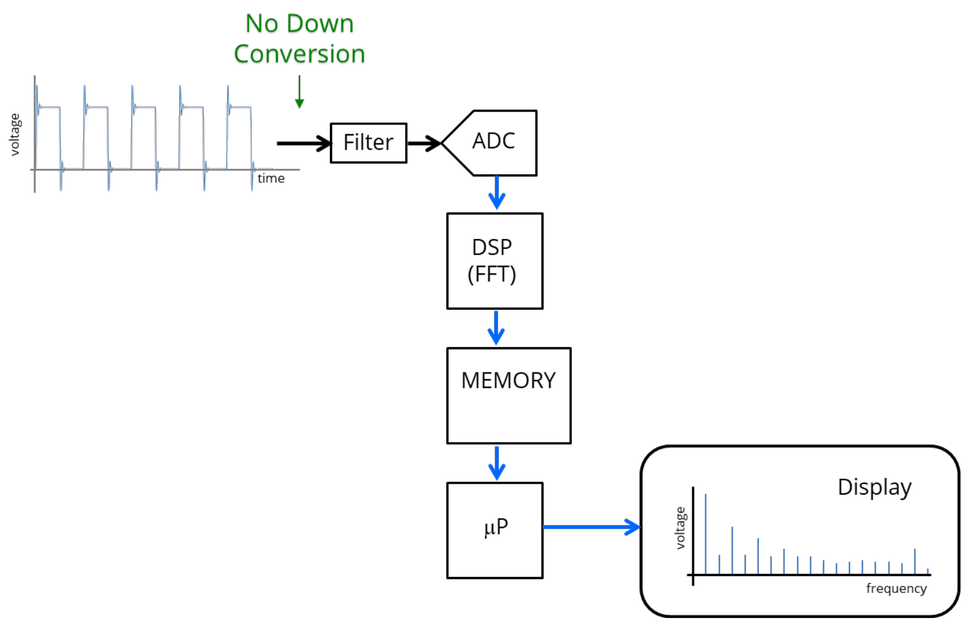


Figure 4.11. Basic operation of a digital oscilloscope with an FFT function.

Peak, Quasi-Peak and Average Measurements

A spectrum analyzer or EMI test receiver measures the power in a given resolution bandwidth as a function of frequency. This power is typically expressed as an rms voltage across the 50-Ω input resistance of the test equipment and compared to a limit specified in the test specification. However, depending the on the specification, the limit may apply to a measured peak value, a quasi-peak value or an average value. For example, the CISPR 32 conducted emissions specification, shown in Figure 4.12, places simultaneous limits on both the quasi-peak and the average values.

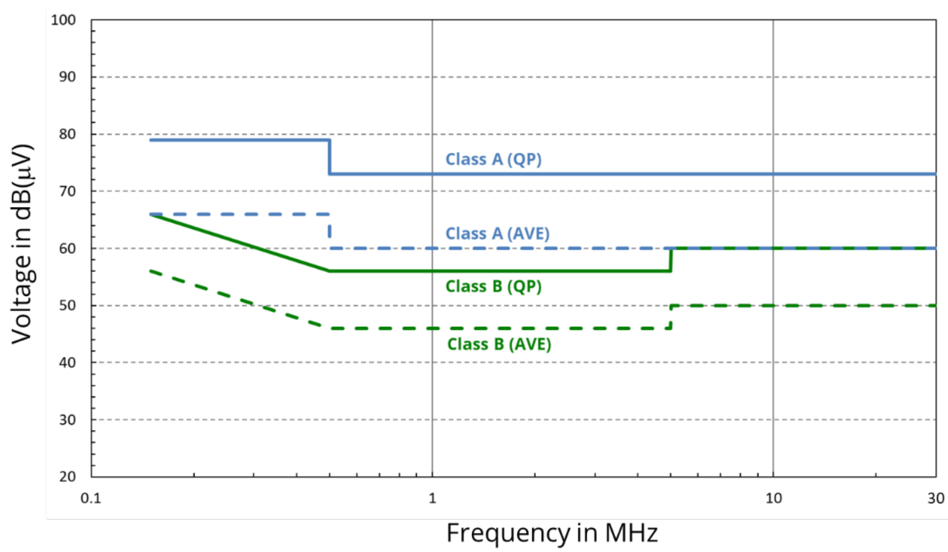


Figure 4.12. FCC and CISPR 32 conducted emissions limits.

Figure 4.13 illustrates how an intermittent signal results in different levels as detected by a peak, quasi-peak, and average detector.

Peak Value: The peak value is the highest value of average power measured in the given resolution bandwidth at the given frequency over the course of the measurement.

Average Value: The *average value* is the average power measured in the given resolution bandwidth at the given frequency over the course of the measurement. For example, if the signal is present 10% of the time and not present 90% of the time, the average value is equal to 10% of the peak value.

Quasi-peak Value: Quasi-peak detection was developed as a way to quantify the amount of annoyance caused by repetitive pulsed noise sources. The quasi-peak value is determined by looking at the detector output as a function of time. When the signal is present, the detector output ramps up to the peak signal power with a specified attack time constant. When the signal is missing or lower in amplitude, the detector output ramps down with a given decay time constant. The average value of the detector output is the quasi-peak value.

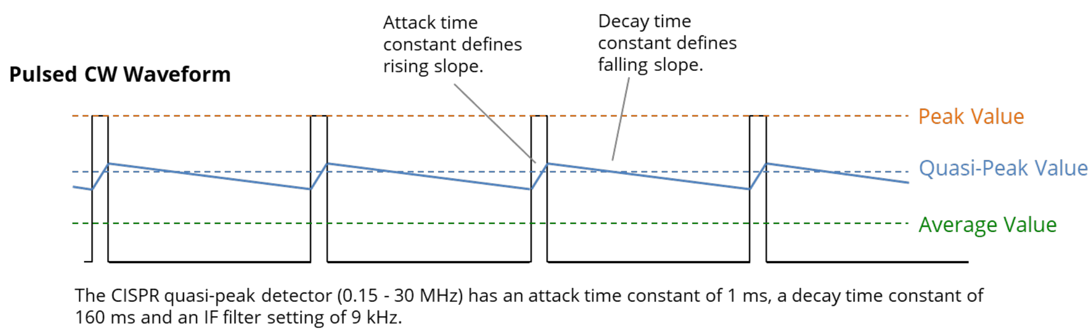


Figure 4.13. Response of peak, quasi-peak, and average detectors to an intermittent signal.

Peak values are always greater than or equal to quasi-peak values, which are always greater than or equal to average values. For a signal that is always present with constant amplitude, all three values are the same.