

Analog and Digital Signals

- A continuous-time signal that assumes a continuum of amplitude values between given maximum and minimum is called an **analog** signal
 - Most signals we encounter in the real world are analog in nature. Examples include speech, music, image, and video signals
- **Digital signals**, on the other hand, can change values at discrete instants of time, assuming one of a finite number of amplitude levels











Frequency Domain Representation

- Although electrical signals used in communication systems are functions of time, such as voltage and current, it is very useful to think of signals in terms of their frequency content
- Certain characteristics of signals are easier to analyze and measure in the frequency domain. In addition, the frequency domain analysis of many important operations on signals leads to unique and valuable insight towards understanding their effect
- That is why the frequency domain representation and analysis of signals and systems is an integral part of design tools for communication and control systems
- Figure shows the time domain representation of a 10 Hz sine wave embedded in noise

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Frequency Domain Representation (contd)

- Difficult to identify a 10 Hz tone in the presence of wideband ("white") noise on an oscilloscope display
- However, easy to identify 10 Hz tone in the frequency domain using a spectrum analyzer display. Note that the white noise forms the floor of the display
- In more complex situations, the composite signal may consist of hundreds of channels or carriers. An example is CATV system where several hundred channels or signals may be present
- Analyzing such a complex signal in time domain is not very useful. The frequency domain analysis, on the other hand, provides valuable insight into the effects of system impairments and noise when dealing with such signals

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Exponential Fourier Series (FS)

- The Fourier series can be used to represent **periodic** signals in the frequency domain
- A periodic function $x_p(t)$ with fundamental period T_o can be represented by an **exponential** Fourier series

$$c_p(t) = \sum_{n=-\infty}^{\infty} C_n e^{j2\pi n f_o t}$$

where $f_o = 1/T_o$ is called the **fundamental** frequency of the periodic signal $x_p(t)$

• The FS coefficients C_n are given by

$$C_n = \frac{1}{T_o} \int_T x_p(t) e^{-j2\pi n f_o t} dt$$

Observe that the FS expands a periodic function as an infinite sum of complex phasor signals

Exponential FS (contd)

• The term C_0 represents the **DC** component of the signal

$$C_0 = \frac{1}{T_o} \int_{T_o} x_p(t) dt$$

- n = ±1 FS coefficients represent the fundamental frequency (f_o) component in the periodic signal x_p(t)
- n = ±2, ±3,... FS coefficients represent the harmonic (nf_o) components in the periodic signal x_p(t)
- Each phasor term in FS can be written as

$$C_{n}e^{j2\pi nf_{o}t} = |C_{n}|e^{j(2\pi nf_{o}t+\Box C_{n})}$$

Plots of |C_n| and □ C_n versus *discrete* frequency values (n f_o, n = 0, ±1, ±2,....) are called the magnitude and the phase line spectra of the signal, respectively

Trigonometric Fourier Series

• For a real signal,

- $C_{-n} = \frac{1}{T_o} \int_{T_o} x_p(t) e^{j2\pi n f_o t} dt = \left(\frac{1}{T_o} \int_{T_o} x_p(t) e^{-j2\pi n f_o t} dt\right) = C_n^*$
- ⇒ the magnitude spectrum is an even function, and the phase spectrum is an odd function of frequency
- In this case, the exponential FS can be expressed as

$$x_{p}(t) = \sum_{n=-\infty}^{\infty} C_{n} e^{j2\pi n f_{n}^{t}} = C_{o} + \sum_{n=1}^{\infty} C_{-n} e^{-j2\pi n f_{n}^{t}} + \sum_{n=1}^{\infty} C_{n} e^{j2\pi n f_{n}^{t}}$$
$$= C_{0} + \sum_{n=1}^{\infty} |C_{n}| [e^{j(2\pi n f_{n}^{t} + \Box C_{n})} + e^{-j(2\pi n f_{n}^{t} + \Box C_{n})}]$$
$$= C_{0} + 2\sum_{n=1}^{\infty} |C_{n}| \cos(2\pi n f_{o}^{t} + \Box C_{n})$$
Trigonometric Fourier series









- Energy and power are useful parameters of a signal
- The normalized **energy** of a signal *x*(*t*) is defined as the energy dissipated by a voltage *x*(*t*) applied across a 1-ohm resistor (or a current *x*(*t*) passing though a 1-ohm resistor)

 $E_x \Box \int_{-\infty}^{\infty} |x(t)|^2 dt$

- The energy of a signal is meaningful only if the integral value is finite. Such signals are called **energy** signals
- **Example** Energy of a rectangular pulse $x(t) = A\Pi(t/T_b)$

$$\begin{aligned} \mathbf{x}(t) &= \begin{cases} A, & |t| \le I_b / 2\\ 0, & \text{otherwise} \end{cases} \\ E_x &= \int_{-\infty}^{\infty} |\mathbf{x}(t)|^2 dt = \int_{-T_b/2}^{T_b/2} A^2 dt = A^2 T_b \end{cases} \end{aligned}$$

Example: Energy of the Carrier Pulse $x(t) = \begin{cases} A\cos(2\pi f_c t) & |t| \le T_b / 2 \\ 0 & \text{otherwise} \end{cases}$ $E_x = \int_{-\infty}^{\infty} |x(t)|^2 dt = A^2 \int_{-T_b/2}^{T_b/2} \cos^2(2\pi f_c t) dt = \frac{A^2}{2} \int_{-T_b/2}^{T_b/2} [1 + \cos(4\pi f_c t)] dt$ $= \frac{A^2}{2} \int_{-T_b/2}^{T_b/2} dt + \frac{A^2}{2} \int_{-T_b/2}^{T_b/2} \cos(4\pi f_c t) dt = \frac{A^2 T_b}{2}$

- The second integral is zero because carrier frequency $f_c >> 1/T_b$ has been assumed true in practice
- The energy content of the signal becomes infinite in the limit as $T_b \rightarrow \infty$

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Power Signals

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• The normalized average **power** of a signal *x*(*t*) is defined as the power dissipated by a voltage *x*(*t*) applied across a 1-ohm resistor (or a current *x*(*t*) passing though a 1-ohm resistor)

$$P_x \Box \lim_{T \to \infty} \frac{1}{T} \int_{-T/2}^{T/2} |x(t)|^2 dt$$

- The normalized average power of a signal is meaningful only if the limit exists (that is, finite). Such signals are called **power** signals
- For a periodic signal $x_p(t)$ with period T_o , the expression for normalized power simplifies to

$$P \Box \frac{1}{T_o} \int_{-T_o/2}^{T_o/2} \left| x_p(t) \right|^2 dt$$

Example: Power of a Sinusoidal Signal

$$\begin{aligned} x_p(t) &= A\cos(2\pi f_o^t t + \phi) \\ P &= \frac{A^2}{T_o} \int_{-T_o/2}^{T_o/2} \cos^2(2\pi f_o^t t + \phi) dt \\ &= \frac{A^2}{2T_o} \Big[\int_{-T_o/2}^{T_o/2} dt + \int_{-T_o/2}^{T_o/2} \cos(4\pi f_c t + 2\phi) dt \Big] = \frac{A^2}{2} + 0 = \frac{A^2}{2} \end{aligned}$$

- The second integral is zero because it evaluates the integrand over two complete periods
- A signal cannot be both power- and energy-type, because for energy signals $P_x = 0$ and $E_x = \infty$ for power signals
 - · A signal may be neither energy-type nor power-type

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Bandwidth of a Signal

- The **bandwidth** of a signal is a measure of the frequency range that contains **significant** energy of the signal
- The term significant here implies inclusion of those
- frequencies that represent the signal with acceptable distortionThe latter is determined by the relevance in a given application.
- If the significant energy of the signal lies in the range of frequencies f₁ < f < f₂, the bandwidth would be f₂ f₁
- There are many definitions of bandwidth depending on how frequencies f_1 and f_2 are chosen
- For example, if the frequencies f_1 and f_2 are chosen so that 99% of the power resides in the frequency band $f_1 < f < f_2$, the quantity $f_2 f_1$ is called the **99% power bandwidth**

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• For example, power at DC frequency	n	C_n	Accumulated Power up to and including $f = nf_o$
De nequency	0	0.5	0.25
$\left C_o \right ^2 = 0.25 \left \operatorname{sinc}(0) \right ^2 = 0.25$	1	0.6366	0.4526
	3	-0.212	0.4752
• As the table shows, we	5	0.1273	0.4833
need to include 21 FS	7	-0.091	0.4874
power in the signal	9	0.0707	0.4899
Since each exected	11	-0.058	0.4916
• Since each spectral	13	0.0490	0.4928
by 1 MHz the 99%	15	-0.0424	0.4937
power bandwidth of the	17	0.0374	0.4944
periodic pulse train is	19	-0.0335	0.4949
~21 MHz.	21	0.0303	0.4954

99% Power Bandwidth Rectangular Pulse Train

• We will assume $T_o = 1 \ \mu \text{sec}$ and $\tau T_o = 0.5$

$$f_o = \frac{1}{T} = 1 \text{ MHz}$$

- The FS coefficients of a rectangular pulse train are given by $C_n = \frac{A\tau}{T} \operatorname{sinc}(nf_o \tau) = 0.5 \operatorname{sinc}(0.5n)$
- The normalized average power P is

$$P = \frac{1}{T_o} \int_{-T_o/2}^{T_o/2} \left| x_p(t) \right|^2 dt = \frac{1}{T_o} \int_{-\tau/2}^{\tau/2} A^2 dt = \frac{A^2 \tau}{T_o} = 0.5$$

• The power in various frequency components is given by $|C_n|^2 = 0.25 |\operatorname{sinc}(0.5n)|^2$

Fourier Transform

• Any continuous-time signal *x*(*t*) that has finite "energy", i.e.,

$\int |x(t)|^2 dt < \infty$

can be represented in the frequency domain via the Fourier transform (FT)

$$X(f) = \int_{-\infty}^{\infty} x(t) e^{-j2\pi f t} dt$$

• In general, *X*(*f*) is a complex function of frequency *f* and can be written as

$$X(f) = |X(f)| e^{j \Box X(f)}$$

where |X(f)| and $\Box X(f)$ are, respectively, called the **magnitude** and the **phase spectrum** of the signal x(t)

Fourier Transform (contd)

• The signal *x*(*t*) can be recovered from its FT *X*(*f*) using the **inverse Fourier transform** formula

 $x(t) = \int_{-\infty}^{\infty} X(f) e^{j2\pi f t} df$

- Note that *X*(*f*) is a continuous spectrum vs the line spectrum produced by FS coefficients *C_n* for a periodic signal
- Notation

 $\begin{aligned} x(t) &\stackrel{3}{\longrightarrow} X(f) \text{ or } X(f) = \Im\{x(t)\} \text{ FT operation} \\ X(f) &\stackrel{3^{-1}}{\longrightarrow} x(t) \text{ or } x(t) = \Im^{-1}\{X(f)\} \text{ Inverse FT operation} \\ x(t) &\stackrel{3}{\longleftrightarrow} X(f) \text{ FT or Inverse FT operation} \end{aligned}$











Time Function x(t)	Fourier Transform $X(f)$	
DC signal A Rectangular pulse $\Pi(N\pi)$	$A\delta(f)$ $\tau sinc(f\tau)$	
Triangular pulse $\Lambda(\theta'\tau)$	$\frac{\tau}{2} \operatorname{sinc}^2\left(\frac{f\tau}{2}\right)$	
Decaying exponential $e^{-\alpha t}u(t)$	$\frac{1}{\alpha + j2\pi f}$	
e-41	$\frac{2\alpha}{\alpha^2 + (2\pi f)^2}$	
Unit impulse $\delta(t)$	1	
$\delta(t - t_0)$	$e^{-j2\pi\beta_s}$	
Sinc pulse sinc(2Wr)	$\frac{1}{2W}\Pi(f/2W)$	
Complex sinusoid $e^{j2\pi f_s^2}$	$\delta(f - f_c)$	
Sinusoid $sin(2\pi f_e t)$	$\frac{1}{2j} \left[\delta(f - f_i) - \delta(f + f_i)\right]$	
Sinusoid $\cos(2\pi f_t t)$	$\frac{1}{2} \left[\delta(f - f_l) + \delta(f + f_l) \right]$	
Gaussian pulse $e^{-\alpha t^2}$	e^{-af^2}	
$\operatorname{sgn}(t)$	$\frac{1}{j\pi f}$	
Unit step u(t)	$\frac{1}{2}\delta(f) + \frac{1}{j2\pi f}$	
$\frac{1}{\pi t}$	-jsgn(f)	
800	2=1	







Properties of FT: Convolution

 $x(t) \otimes y(t) \xleftarrow{\mathfrak{I}} X(f) Y(f)$

• From the definition of convolution operation

$$x(t) \otimes y(t) = \int_{-\infty}^{\infty} x(\tau) y(t-\tau) d\tau$$

• If we take the Fourier transform of the right hand side, and exchange the order of integration, we get

$$\int_{-\infty}^{\infty} \left[\int_{-\infty}^{\infty} x(\tau) y(t-\tau) d\tau \right] e^{-j2\pi\beta} dt = \int_{-\infty}^{\infty} d\tau x(\tau) \left[\int_{-\infty}^{\infty} y(t-\tau) e^{-j2\pi\beta} dt \right]$$
$$= Y(f) \int_{-\infty}^{\infty} x(\tau) e^{-j2\pi/\tau} d\tau = Y(f) X(f)$$

 ⇒ the convolution operation in time-domain is equivalent to multiplication in the frequency domain

Properties of FT: Multiplication Property

 This property is the dual of the convolution property. The multiplication of two signals results in the convolution of their spectra

 $x(t)y(t) \xleftarrow{3} X(f) \otimes Y(f)$

Modulation involves multiplication of a signal x(t) by a high-frequency sinusoidal waveform. That is,
 y(t) = cos(2\pi f_ct)x(t)

• Applying the multiplication property

$$Y(f) = \left(\frac{1}{2}\delta(f - f_c) + \frac{1}{2}\delta(f + f_c)\right) \otimes X(f)$$

$$= \frac{1}{2}X(f - f_c) + \frac{1}{2}X(f + f_c)$$
Spectrum X(f) shifted
by carrier frequency f_c ³⁶

Properties of FT: Time/Frequency Scaling

$$\begin{aligned}
x(at) &\leftarrow \frac{3}{|a|} X\left(\frac{f}{a}\right) \\
\text{o. Assume } a > 0. \text{ Using the transformation of variables, } u = at, \\
we have
$$x(at) &\leftarrow \frac{3}{-\infty} = \int_{-\infty}^{\infty} x(at)e^{-j2\pi/t}dt = \frac{1}{a}\int_{-\infty}^{\infty} x(u)e^{-j2\pi(f/a)u}du = \frac{1}{a}X\left(\frac{f}{a}\right) \\
\text{o. Now with } a < 0, \text{ substituting } u = -|a|t \text{ yields} \\
x(at) &\leftarrow \frac{3}{-\infty} = \int_{-\infty}^{\infty} x(-|a|t)e^{-j2\pi/t}dt = \frac{1}{|a|}\int_{-\infty}^{\infty} x(u)e^{j2\pi(f/|e|)u}du \\
&= \frac{1}{|a|}X\left(-\frac{f}{a}\right) = \frac{1}{|a|}X\left(\frac{f}{a}\right)
\end{aligned}$$$$

Properties of Fourier Transform (contd)

- The function *x*(*at*), for *a* > 0, is a time compressed (by a factor *a*) version of *x*(*t*). On the other hand, a function *X*(*f*)*a*) represents a function *X*(*f*) expanded by the same factor *a*.
- The scaling property therefore states that compressing a signal in time domain will stretch its Fourier transform. Similarly stretching a time signal will compress its Fourier transform.
- The result is intuitively satisfying since compression in time by the factor *a* > 0 means that the function is varying rapidly in time by the same amount
- Consequently, the frequencies of its components will be increased by the factor *a*. The converse can also be justified by a similar argument.









Energy Spectral Density

- Equation (*) states that the energy of a signal is given by the area under the curve $|X(f)|^2$
 - $|X(f)|^2$ is called the **energy spectral density** of x(t)
- Note that the quantity $|X(f_o)|^2 \Delta f$ represents the energy contained in 2 spectral bands of Δf Hz centered at frequencies $\pm f_o$
- Thus $|X(f)|^2$ may be interpreted as the energy contained in the spectral components of x(t) centered at frequency f per Hz of bandwidth
- $|X(f)|^2$ thus represents distribution of energy as a function of frequency for a signal. So when integrated, it provides the energy of the signal
- The energy spectral density of a signal is specified in units of Watts-sec/Hz

Fourier Transform Properties

Property	Time Function x(t) y(t)	Fourier Transform X(f) Y(f)
Linearity	$\alpha x(t) + \beta y(t)$	$\alpha X(f) + \beta Y(f)$
Time-shifting	$x(t - t_0)$	$X(f)e^{-j2\pi ft_o}$
Frequency translation	$x(t)e^{j 2\pi f_c t}$	$X(f - f_c)$
Convolution	$x(t) \otimes y(t)$	X(f)Y(f)
Multiplication	x(t)y(t)	$X(f) \otimes Y(f)$
Time/Frequency scaling	x(at)	$\frac{1}{ a }X\left(\frac{f}{a}\right)$
Duality	X(t)	x(-f)
Differentiation in time	$\frac{d}{dt}x(t)$	$j2\pi f X(f)$
Differentiation in frequenc	y = tx(t)	$\frac{j}{2\pi} \frac{d}{df} X(f)$
Integration	$\int_{-\infty}^{t} x(\tau) d\tau$	$\frac{X(f)}{j2\pi f} + \frac{1}{2}X(0)\delta(f)$
Parseval's relation	$\int_{-\infty}^{\infty} x(t)y^{*}(t)dt = \int_{-\infty}^{\infty} X(f)Y^{*}(f)df$	

Fourier Transforms of Periodic Signals

- The Fourier transform is strictly defined for finite energy signals. However, we can extend its scope by allowing the FT to include delta functions
- Since a periodic signal can be expanded into exponential FS, its FT can be obtained by taking the FT of the FS term by term
- The FT expansion of a periodic function $x_p(t)$ is obtained as

$$\begin{split} X_p(f) &= \Im \left\{ \sum_{n=-\infty}^{\infty} C_n e^{j2\pi n f_n t} \right\} = \sum_{n=-\infty}^{\infty} C_n \Im \left\{ e^{j2\pi n f_n t} \right\} \\ &= \sum_{n=-\infty}^{\infty} C_n \mathcal{S} \left(f - n f_n \right) \end{split}$$

 ⇒ the FT of a periodic signal consists of impulses located at harmonic frequencies of the signal. The weight of each impulse equals FS coefficient, i.e., X_p(nf_o) = C_n









- However, a signal can be "approximately" time-limited and band-limited
 - That is, there exist numbers B > 0 and T > 0, such that |x(t)| is small for $|t| \ge T$ and |X(t)| is small for $|t| \ge B$
- The product of a signal's duration and its bandwidth is constant.
 - Duration × Frequency Bandwidth = k
- The constant k is determined by the precise definitions of **duration** in the time domain and **bandwidth** in the frequency domain
 - For a Gaussian pulse, if we use the RMS definitions of duration and bandwidth of a signal, it can be shown that

 $\Delta f_{rms} \Delta t_{rms} = \frac{1}{4\pi}$

Transmission of Signals Through LTI Systems

• An Linear Time-invariant (LTI) system is completely characterized in the time domain by its impulse response *h*(*t*)

 $y(t) = x(t) \otimes h(t) = \int_{-\infty}^{\infty} x(\tau)h(t-\tau)d\tau$

x(t) LTI system

• Applying the FT to both sides and using convolution property

Y(f) = X(f)H(f)

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• ⇒ that the output of the system in the frequency domain is given by multiplying the Fourier transform of the input by the system frequency response *H*(*f*)

X(f) $h(t) \xleftarrow{3} H(f)$ Y(f) = X(f)H(f)

 $y(t) = x(t) \otimes h(t)$





Distortionless Transmission

• An LTI system is termed **distortionless** if it introduces the same attenuation to all spectral components and offers linear phase response over the frequency band of interest, that is,

$$H_{ideal}(f) = \begin{cases} H_o e^{-j2\pi f f_o} & f_1 \le f \le f_2 \\ 0 & \text{otherwise} \end{cases}$$

• Substituting yields

$$Y(f) = X(f)H_{ideal}(f) = H_o X(f)e^{-j2\pi f t_o}$$

• Taking the FT of both sides, the output of a distortionless LTI system due to an arbitrary input signal *x*(*t*) is given by

delayed and scaled replica of the input

 $y(t) = H_o x(t - t_o)$

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Group Delay

- The group delay of an LTI system is defined as $\tau_g(f) \Box - \frac{1}{2\pi} \frac{d\Box H(f)}{df}$
 - Represents time delay incurred by a spectral component at frequency *f* as it passes through the LTI system
- The phase response of an ideal filter is linear function of frequency as given by
- $\Box H_{ideal}(f) = -2\pi f t_o, \qquad f_1 \le f \le f_2, \ t_o = \text{constant}$ • For a linear phase LTI system, we obtain

$$\tau_g(f) = -\frac{1}{2\pi} \frac{-2\pi f t_o}{df} = t_o = \text{constant}$$

• \Rightarrow all frequency components of the input signal undergo the same time delay through the system \Rightarrow *no distortion*

Ideal Filters

- One key application of LTI systems is processing of signals in order to enhance certain frequency components and to reject certain others
 - For example, if a signal consists of a low-frequency information-bearing message waveform and high-frequency noise, we can employ a filter to reject the high frequencies and thus remove the noise
- An ideal filter designed to pass certain frequency components should have a magnitude response that is constant and phase response that is linear over these frequencies- called the **passband** of the filter
- The magnitude response of the ideal filter is zero over the range of frequencies blocked by the filter called the **stopband** of the filter





Ideal Highpass Filter

• The magnitude response of an ideal highpass (HP) filter is

$$H_{HP}(f) = \begin{cases} 0 & -W \le f \le W \\ A_o & \text{otherwise} \end{cases}$$

- The range of frequencies $f \le W$ is the stopband of the filter. The range of frequencies f > W is the passband of the filter
- The frequency response of an ideal HP filter can now be written as

 $H_{HP}(f) = A_o[1 - \Pi(f/2W)]e^{-j2\pi f t_o}$

• Taking the inverse FT yields $H_{HP}(f) = A_o [1 - \Pi(f / 2W)] e^{-j2\pi f_o}$

Ideal Bandpass Filter

• The amplitude response of an ideal bandpass (BP) filter is

$$H_{BP}(f) = \begin{cases} A_o & f_c - W \le |f| \le f_c + W \\ 0 & \text{otherwise} \end{cases}$$

- The range of frequencies $f_c W \le |f| \le f_c + W$ is the passband of the filter. The range of frequencies $|f| > f_c + W$ and $|f| < f_c W$ are the stopband regions of the filter
- The frequency response of an ideal BP filter can now be written as

$$H_{BP}(f) = H_o(f - f_c) + H_o(f + f_c)$$

where

$$H_o(f) = A_o \Pi(f/2W) e^{-j2\pi f t}$$



Ideal Bandstop Filter

• The amplitude response of an ideal *bandstop* (BS) filter is defined as

$$|H_{BS}(f)| = \begin{cases} A_o & \text{otherwise} \\ 0 & f_c - W \le |f| \le f_c + W \end{cases}$$

• The range of frequencies $f_c - W \le |f| \le f_c + W$ is the stopband of the filter. The range of frequencies $|f| > f_c + W$ and $|f| < f_c - W$ are the passband regions of the filter.

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Power Density Spectrum

- In the design of communication systems, we are interested in power distribution of a power signal in the frequency domain
- The problem in dealing with power signals in the frequency domain is that their Fourier transform may not exist as they have infinite energy
- To overcome this problem, we define a new function $x_T(t)$ by truncating x(t) outside the interval |t| > T/2

$$x_T(t) = \begin{cases} x(t) & -T/2 \le t \le T/2 \\ 0 & \text{otherwise} \end{cases}$$

• $x_T(t)$ has finite energy as long as T is finite. Using Parseval's relation

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$$E_{x_T} = \int_{-\infty}^{\infty} |x_T(t)|^2 dt = \int_{-\infty}^{\infty} |X_T(f)|^2 df$$

Power Density Spectrum (contd)
• Since

$$\int_{-T/2}^{T/2} |x(t)|^2 dt = \int_{-\infty}^{\infty} |x_T(t)|^2 dt = \int_{-\infty}^{\infty} |X_T(f)|^2 df$$
the average power of signal can be expressed as

$$P_x = \lim_{T \to \infty} \frac{1}{T} \int_{-T/2}^{T/2} |x(t)|^2 dt = \lim_{T \to \infty} \frac{1}{T} \int_{-\infty}^{\infty} |X_T(f)|^2 df$$
• Since $x(t)$ is a power signal, the integral on the right hand side
exists in the limit as $T \to \infty$. Therefore, we can change the
order of integration and limit yielding

$$P_x = \lim_{T \to \infty} \int_{-\infty}^{\infty} \frac{|X_T(f)|^2}{T} df = \int_{-\infty}^{\infty} \lim_{T \to \infty} \frac{|X_T(f)|^2}{T} df$$

PSD of Periodic Signals

• For a periodic signal $x_p(t)$, the normalized average power is given by

$$P_x = \sum_{n=-\infty}^{\infty} |C_n|^2$$

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Since |C_n|² is power contained in the spectral component at f = ηf_o, the PSD G_k(f) of a periodic signal can be expressed as

$$\boldsymbol{\mathcal{G}}_{\boldsymbol{x}}(f) = \sum_{n=-\infty}^{\infty} \left| C_n \right|^2 \delta(f - nf_o)$$

• Since FS coefficient $C_n = X_p(nf_o)$, we can express the PSD as

$$\boldsymbol{\mathcal{G}}_{x}(f) = \sum_{n=-\infty}^{\infty} \left| \boldsymbol{X}_{p}(nf_{o}) \right|^{2} \delta(f - nf_{o})$$



For a linear system with transfer function H(f), the output y(t) in response to a deterministic input signal x(t) is given by
 Y(f) = X(f)H(f)

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$$\boldsymbol{\mathcal{G}}_{\boldsymbol{x}}(f) \xrightarrow{\text{LTI System}} \boldsymbol{\mathcal{G}}_{\boldsymbol{y}}(f) = |H(f)|^2 \boldsymbol{\mathcal{G}}_{\boldsymbol{x}}(f)$$

$$\overrightarrow{\boldsymbol{X}(f)} \xrightarrow{\boldsymbol{X}(f)} H(f) \xrightarrow{\boldsymbol{3}} H(f) \quad Y(f) = H(f)X(f)$$

• The PSD of a power signal y(t) can be written as $\boldsymbol{G}_{y}(f) = \lim_{T \to \infty} \frac{|Y_{T}(f)|^{2}}{T}$

• Now

$$Y_T(f) = X_T(f)H(f)$$

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Response of LTI System (contd)

• Substituting yields

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$$\boldsymbol{\mathcal{G}}_{\mathcal{Y}}(f) = \lim_{T \to \infty} \frac{|X_{\tau}(f)H(f)|^2}{T} = |H(f)|^2 \lim_{T \to \infty} \frac{|X_{\tau}(f)|^2}{T}$$
$$= |H(f)|^2 \boldsymbol{\mathcal{G}}_{\tau}(f)$$

- \Rightarrow that the output signal PSD in an LTI system depends on the magnitude of H(f), and is given by $|H(f)|^2$ times the input PSD
- Taking inverse FT of both sides and applying the convolution property, we obtain

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 $\boldsymbol{\mathcal{R}}_{\boldsymbol{y}}(\tau) = h(\tau) \otimes h^{*}(-\tau) \otimes \boldsymbol{\mathcal{R}}_{\boldsymbol{x}}(\tau)$