• Since the frequency response *H* is simply the frequency spectrum of the impulse response *h*, if *h* is *real*, then

$$|\mathcal{H}(\Omega)| = |\mathcal{H}(-\Omega)|$$
 and $\arg \mathcal{H}(\Omega) = -\arg \mathcal{H}(-\Omega)$

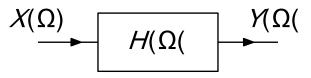
)i.e., the magnitude response $|\mathcal{H}(\Omega)|$ is *even* and the phase response arg $\mathcal{H}(\Omega)$ is *odd*.

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- Consider a ITI system with input X, output Y, and impulse response h, and let X, Y, and H denote the Fourier transforms of X, Y, and h, respectively.
- Often, it is convenient to represent such a system in block diagram form in the frequency domain as shown below.



• Since a ITI system is completely characterized by its frequency response, we typically label the system with this quantity.

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Representations of ITLS ystems Difference-Equation

- Many ITI systems of practical interest can be represented using an *Nth-order linear difference equation with constant coefficients*
- Consider a system with input X and output Y that is characterized by an equation of the form

$$\sum_{k=0}^{N} b_{k} y(n-k) = \sum_{k=0}^{M} a_{k} x(n-k)$$

- Let h denote the impulse response of the system, and let X, Y, and H denote the Fourier transforms of X, y, and h, respectively.
- One can show that $H(\Omega)$ is given by

$$\mathcal{H}(\Omega = \left(\frac{Y(\Omega)}{X(\Omega)} = \frac{\sum_{k=0}^{M} a_k(\theta^{j\Omega})^{-k}}{\sum_{k=0}^{N} b_k(\theta^{j\Omega})^{-k}} = \frac{\sum_{k=0}^{M} a_k e^{-jk\Omega}}{\sum_{k=0}^{N} b_k e^{-jk\Omega}}$$

• Each of the numerator and denominator of H is a *polynomial* in $e^{-j\Omega}$. • Thus, H is a *rational function* in the variable $e^{-j\Omega}$.

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Section 10.6

Application: Filtering

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- In many applications, we want to *modify the spectrum* of a signal by either amplifying or attenuating certain frequency components.
- This process of modifying the frequency spectrum of a signal is called filtering.
- A system that performs a filtering operation is called a filter.
- Many types of filters exist.
- Frequency selective filters pass some frequencies with little or no distortion, while significantly attenuating other frequencies.
- Several basic types of frequency-selective filters include: lowpass, highpass, and bandpass.

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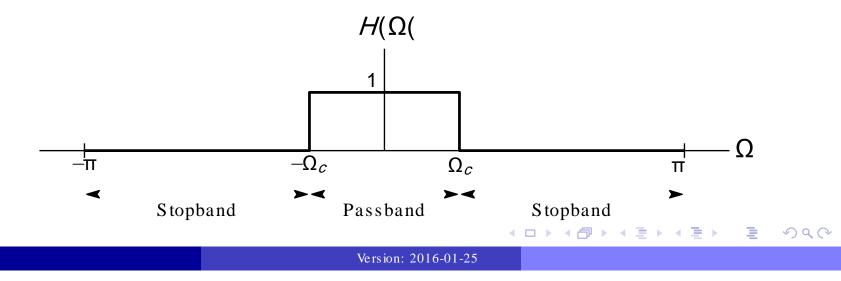
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- An ideal lowpass filter eliminates all baseband frequency components with a frequency whose magnitude is greater than some cutoff frequency, while leaving the remaining baseband frequency components unaffected.
- Such a filter has a *frequency response H* of the form

$$H(\Omega = \begin{pmatrix} 1 & \text{if } |\Omega| \le \Omega_{C} \\ 0 & \text{if } \Omega_{C} < |\Omega| \le \pi^{c} \end{pmatrix}$$

where Ω_c is the cutoff frequency.

• A plot of this frequency response is given below.

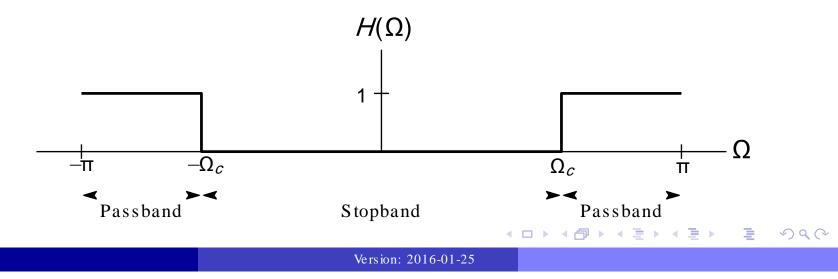


- An ideal highpass filter eliminates all baseband frequency components with a frequency whose magnitude is less than some cutoff frequency, while leaving the remaining baseband frequency components unaffected. Such a
- filter has a *frequency response H* of the form

$$\mathcal{H}(\Omega) = \begin{array}{cc} 1 & \text{if } \Omega_{\mathcal{C}} < |\Omega| \leq \pi \\ 0 & \text{if } |\Omega| \leq \Omega_{\mathcal{C}}, \end{array}$$

where Ω_c is the cutoff frequency.

• A plot of this frequency response is given below.

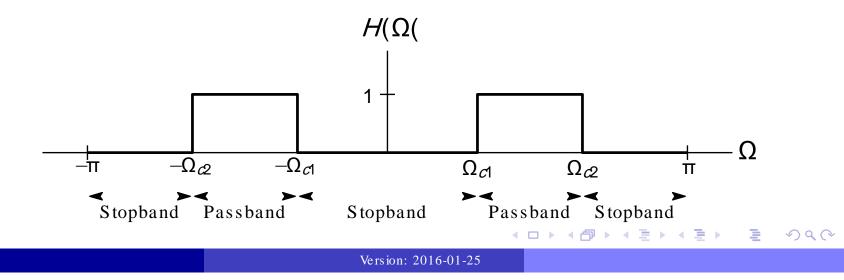


- An ideal bandpass filter eliminates all baseband frequency components with a frequency whose magnitude does not lie in a particular range, while leaving the remaining baseband frequency components unaffected.
- Such a filter has a *frequency response* H of the form

$$\mathcal{H}(\Omega) = \begin{array}{cc} 1 & \text{if } \Omega_{\mathcal{C}1} \leq |\Omega| \leq \Omega_{\mathcal{C}2} \\ 0 & \text{if } |\Omega| < \Omega_{\mathcal{C}1} \text{ or } \Omega_{\mathcal{C}2} < |\Omega| < \pi, \end{array}$$

where the limits of the passband are Ω_{C1} and Ω_{C2} .

• A plot of this frequency response is given below.



Part 11

Z Trans form (ZT)

Version: 2016-01-25

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- Another important mathematical tool in the study of signals and systems is known as the z transform.
- The z transform can be viewed as a *generalization of the Fourier transform*.
- Due to its more general nature, the z transform has a number of advantages over the Fourier transform.
- First, the z transform representation exists for some signals that do not have Fourier transform representations. So, we can handle a *larger class of signals* with the z transform.
- Second, since the z transform is a more general tool, it can provide additional insights beyond those facilitated by the Fourier transform.

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- Earlier, we saw that complex exponentials are eigensequences of ITI systems.
- In particular, for a ITI system H with impulse response h, we have that

$$H{z^n} = H(z)z^n$$
 where $H(z) = \sum_{n=1}^{\infty} h(n)z^{-n}$.

- Previously, we referred to H as the system function.
- As it turns out, H is the z transform of h.
- Since the z transform has already appeared earlier in the context of ITI systems, it is clearly a useful tool.
- Furthermore, as we will see, the z transform has many additional uses.

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Section 11.1

Z Trans form

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• The (bilateral) z transform of the sequence X, denoted $Z\{x\}$ or X, is defined as

$$X(z) = \sum_{n=1}^{\infty} x(n) z^{-n}$$

• The inverse z transform is then given by

$$X(n) = \frac{1}{2\pi j} \int_{\Gamma} X(z) z^{n-1} dz$$

where Γ is a counterclockwise closed circular contour centered at the origin and with radius r such that Γ is in the ROC of X.

• We refer to X and X as a z transform pair and denote this relationship as

$$X(n) \xleftarrow{ZT} X(Z(n))$$

• In practice, we do not usually compute the inverse z transform by directly using the formula from above. Instead, we resort to other means (to be discussed later.(・ロト ・ 四 ト ・ ヨ ト ・ ヨ ト

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- Two different versions of the z transform are commonly used:
 - bilateral (or *two-sided*) z transform; and
 - be *unilateral* (or *one-sided*) z transform.
- The unilateral z transform is most frequently used to solve systems of linear difference equations with nonzero initial conditions.
- As it turns out, the only difference between the definitions of the bilateral and unilateral z transforms is in the *lower limit of summation*.
- In the bilateral case, the lower limit is −∞, whereas in the unilateral case, the lower limit is 0.
- For the most part, we will focus our attention primarily on the bilateral z transform.
- We will, however, briefly introduce the unilateral z transform as a tool for solving difference equations.
- Unless otherwise noted, all subsequent references to the z transform should be understood to mean *bilateral* z transform.

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- Let X and X_F denote the z and (DT) Fourier transforms of X, respectively.
- The function X(z) evaluated at $z = \Theta^{\Omega}$ (where Ω is real) yields $X_{F}(\Omega)$. That is,

•Due to the preceding relationship of the Forth (Ω) transform of X is sometimes written as $X(\theta^{\Omega})$.

•The function X(Z) evaluated at an arbitrary complex value $Z = I \Theta^{\Omega}$ (where I = |Z| and $\Omega = \arg Z$) can also be expressed in terms of a Fourier transform involving X. In particular, we have

$$X(r\Theta^{\Omega}) = X_{F}(\Omega),$$

where $X_{\rm F}$ is the (DT) Fourier transform of $\dot{x}(n) = r^{-n} x(n)$.

- So, in general, the z transform of X is the Fourier transform of an exponentially-weighted version of X.
- Due to this weighting, the z transform of a sequence may exist when the Fourier transform of the same sequence does not.

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