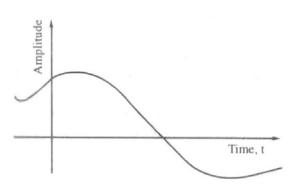
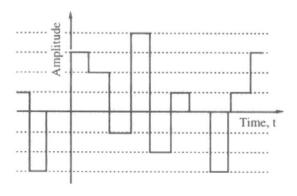
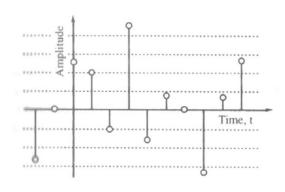
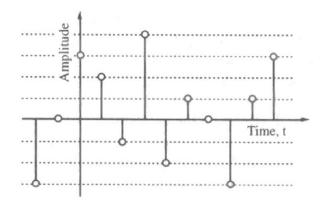
- A continuous-time signal with a continuous amplitude is usually called an analog signal.
 - A speech signal is an example of an analog signal.
- A continuous-time signal with discrete valued amplitudes has been referred to as a quantized boxcar signal.
 - This type of signal occurs in digital electronic circuits where the signal is kept at fixed level (usually one of two values) between two instants of clocking.





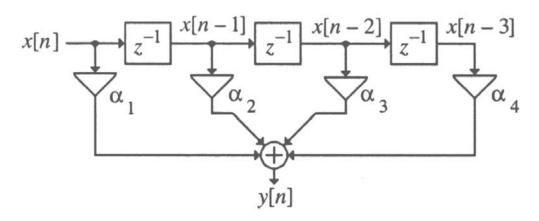
- A discrete time signal with continuous valued amplitudes is called a sampled-data signal.
 - The amplitude of the signal may be any value.
- A discrete time signal with discrete valued amplitudes represented by a finite number of digits is referred to as a digital signal.
 - A digital signal is thus a quantized sampled-data signal.





Combinations of Basic Operations

• Example:



$$y[n] = \alpha_1 x[n] + \alpha_2 x[n-1] + \alpha_3 x[n-2] + \alpha_4 x[n-3]$$

Digital Signal Processing by Yu Yajun @ SUSTech

-

Classification of Sequences

- Based on Symmetry
 - Conjugate-symmetric: $x[n] = x^*[-n]$
 - Conjugate-antisymmetric: $x[n] = -x^*[-n]$
- Base on Periodicity
 - A sequence $\tilde{x}[n]$ satisfying

$$\tilde{x}[n] = \tilde{x}[n+kN]$$
 for all n

is called a periodic sequence with a period *N*.

Digital Signal Processing by Yu Yajun @ SUSTech

4

Basic Sequences

• Unit impulse
$$\delta[n] = \begin{cases} 1 & n = 0 \\ 0 & n \neq 0 \end{cases}$$

• Unit Step
$$\mu[n] = \begin{cases} 1 & n \ge 0 \\ 0 & n < 0 \end{cases}$$

• Exponential
$$x[n] = \begin{cases} A\alpha^n & n \ge 0 \\ 0 & n < 0 \end{cases}$$

• An arbitrary sequency
$$x[n] = \sum_{k=-\infty}^{\infty} x[k]\delta[n-k]$$

- $x[n] = A\cos(\omega_0 n + \varphi)$ Discrete Sinusoids
 - Periodicity with respect to time *n*
 - Periodicity with respect to frequency ω_0

Digital Signal Processing by Yu Yajun @ SUSTech

Discrete Time System

Accumulator

•
$$y[n] = \sum_{l=-\infty}^{n} x[l] = y[n-1] + x[n] = y[-1] + \sum_{l=0}^{n} x[l]$$

• *M*-point Moving-Average Filter

•
$$y[n] = \frac{1}{M} \sum_{l=0}^{M-1} x[n-l] = y[n-1] + \frac{1}{M} (x[n] - x[n-M])$$

• Linear Interpolator (factor of 2)

•
$$y[n] = x_u[n] + \frac{1}{2}(x_u[n-1] + x_u[n+1])$$

Median Filter

•
$$y[n]=med\{x[n-k], ..., x[n-1], x[n], x[n+1], ..., x[n+k]\}$$

Compressor

•
$$y[n] = x[Mn]$$
 for $M > 1$

Time domain representation of Discrete time System

• Input – output relation

Properties of Discrete time System

- Linearity
- Causality
- Memoryless
- Time-invariance
- BIBO-stability
- Passive and Lossless properties

Digital Signal Processing by Yu Yajun @ SUSTech

_

Linear and Time-invariant (LTI) System

- What is impulse response
- How to find the impulse response of a given system
- Why impulse response can fully characterize an LTI system
- How to compute convolution
- How to compute the output of an LTI system, given the input sequence and impulse response
- Why only LTI system can use convolution to compute the output sequence
- BIBO and causal properties of LTI system
- General difference equation

Properties of LTI System

• An LTI system is BIBO stable iff h[n] is absolutely summable

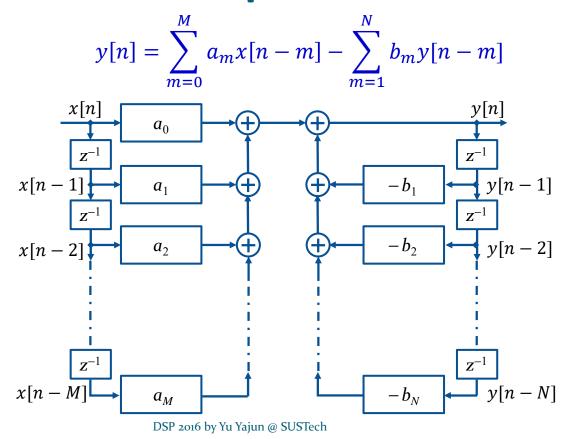
$$\sum_{k=-\infty}^{\infty} |h(k)| < \infty$$

An LTI system is causal iff

$$h[k] = 0$$
 for $k < 0$

Digital Signal Processing by Yu Yajun @ SUSTech

Difference Equation



.

Discrete-Time Fourier Transform (DTFT)

$$X(e^{j\omega}) = \sum_{n=-\infty}^{\infty} x[n]e^{-j\omega n}$$
 DTFT

$$x[n] = \frac{1}{2\pi} \int_{-\pi}^{\pi} X(e^{j\omega}) e^{j\omega n} d\omega$$
 Inverse DTFT

• $X(e^{j\omega})$ is a complex function of the real variable ω

$$X(e^{j\omega}) = X_{\rm re}(e^{j\omega}) + jX_{\rm im}(e^{j\omega});$$
 $X(e^{j\omega}) = |X(e^{j\omega})|e^{j\theta(\omega)}$

Symmetrical and periodic

$$\begin{aligned} |X(e^{j\omega})| &= |X(e^{-j\omega})| \text{ and } \theta(\omega) = -\theta(-\omega) \text{ for real } x[n] \\ X(e^{j\omega}) &= X(e^{j(\omega+2k\pi)}), \text{ i.e.,} \\ |X(e^{j\omega})|e^{j[\theta(\omega)+2k\pi]} &= |X(e^{j\omega})|e^{j\theta(\omega)} \end{aligned}$$

Digital Signal Processing by Yu Yajun @ SUSTech

11

12

Properties of the DTFT

Assume: $x[n] \leftrightarrow X(e^{j\omega}), \ h[n] \leftrightarrow H(e^{j\omega}), \ y[n] \leftrightarrow Y(e^{j\omega})$

- Time Reversal: $x[-n] \leftrightarrow X(e^{-j\omega})$
- Time and Frequency Shifting

$$x[n-n_d] \leftrightarrow e^{-j\omega n_d} X\left(e^{j\omega}\right); \qquad e^{j\omega_0 n} x[n] \leftrightarrow X\left(e^{j(\omega-\omega_0)}\right)$$

• Differentiation in frequency

$$nx[n] \leftrightarrow j \frac{dX(e^{j\omega})}{d\omega}$$

Convolution

$$y[n] = x[n] \otimes h[n] \Rightarrow Y(e^{j\omega}) = X(e^{j\omega})H(e^{j\omega})$$
$$y[n] = x[n]h[n] \Rightarrow Y(e^{j\omega}) = \frac{1}{2\pi} \int_{-\pi}^{\pi} X(e^{j\theta})H(e^{j(\omega-\theta)})d\theta$$

• Parseval's theory:
$$\sum_{n=-\infty}^{\infty} |x[n]|^2 = \frac{1}{2\pi} \int_{-\pi}^{\pi} |X(e^{j\omega})|^2 d\omega$$

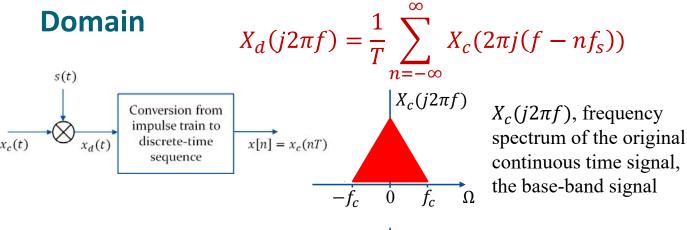
DTFT Convergence

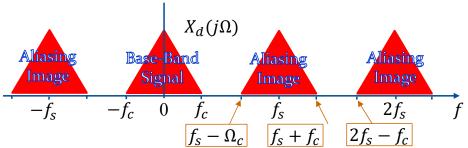
| Sequence | DTFT | | | |
|--|---|--|--|--|
| $\alpha^n \mu[n]$, ($ \alpha < 1$) Absolutely Summable | $ \frac{1}{1 - \alpha e^{-j\omega}} $ Exist for all ω | | | |
| $\mu[n]$ Neither absolutely summable, nor finite energy Not no | $\frac{1}{1 - e^{-j\omega}} + \sum_{k = -\infty}^{\infty} \pi \delta(\omega + 2\pi k)$ Not exist for $\omega = 0$ | | | |
| 1 (for all <i>n</i>) Neither absolutely summable, nor finite energy | $\sum_{k=-\infty}^{\infty} 2\pi \delta(\omega - 2\pi k)$ Exist for all ω | | | |
| $h_{LP}[n] = \frac{\sin \omega_C n}{\pi n}$ Finite energy | $H_{LP}(e^{j\omega}) = \begin{cases} 1, 0 \le \omega \le \omega_c \pi \\ 0, \omega_c \pi \le \omega \le \pi \end{cases}$ Exist for all ω | | | |

Digital Signal Processing by Yu Yajun @ SUSTech

13

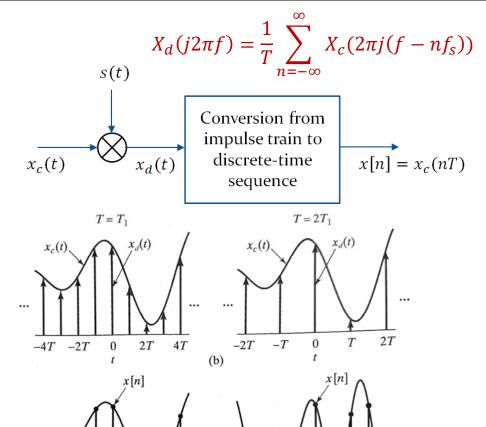
Effect of Time-Domain Sampling in Frequency





The condition to fully recover the continuous signal:

$$f_s \ge 2f_c$$



 $X(e^{j\omega})$ is a frequency scaled version of $X_d(j\Omega)$, $\omega = \Omega T$

A normalization of the frequency axis so that the frequency $\Omega = \Omega_s$ in $X_d(j\Omega)$ is normalized to $\omega = 2\pi$ for $X(e^{j\omega})$.

a result of the time normalization in the transformation from $x_d(t)$ to x[n].

15

Eigenfunction

-4 -3 -2 -1 0 1

$$x[n] \xrightarrow{x[n]} h[n] \qquad y[n]$$

$$x[n] = e^{j\omega n} = f(n) \qquad h[n] \qquad y[n] = C \cdot f(n)$$

$$y[n] = h[n] \otimes e^{j\omega n} = \left(\sum_{k=-\infty}^{\infty} h[k]e^{-j\omega k}\right) e^{j\omega n}$$

Tech

-4 -3 -2 -1 0

(c)

Let

$$H(e^{j\omega}) = \sum_{k=-\infty}^{\infty} h[k]e^{-j\omega k}$$

the **frequency response** of the LTI system with impulse response h[n]

Frequency-Domain Characterization of LTI System

$$y[n] = x[n] \otimes h[n]$$
 $x[n]$ $h[n]$ $y[n]$

$$Y(e^{j\omega}) = X(e^{j\omega})H(e^{j\omega}) \xrightarrow{X(e^{j\omega})} H(e^{j\omega}) \xrightarrow{Y(e^{j\omega})}$$

FIR system
$$H(e^{j\omega}) = \sum_{k=N_1}^{N_2} h[k]e^{-j\omega k}, \quad y[n] = \sum_{k=N_1}^{N_2} h[k]x[n-k],$$

IIR system
$$H(e^{j\omega}) = \frac{Y(e^{j\omega})}{X(e^{j\omega})} = \frac{\sum_{m=0}^{M} a_m e^{-j\omega m}}{\sum_{m=0}^{N} b_m e^{-j\omega m}}$$

$$\sum_{m=0}^{N} b_m y[n-m] = \sum_{m=0}^{M} a_m x[n-m]$$

Digital Signal Processing by Yu Yajun @ SUSTech

DFT & IDFT

DFT:

$$X[k] = \sum_{n=0}^{N-1} x[n] e^{-\frac{j2\pi kn}{N}} = X(e^{j\omega}) \Big|_{\omega = 2\pi k/N} = \sum_{n=0}^{N-1} x[n] W_N^{kn},$$

where,
$$W_N = e^{-j2\pi/N}$$
, for $k = 0, 1, ..., N - 1$

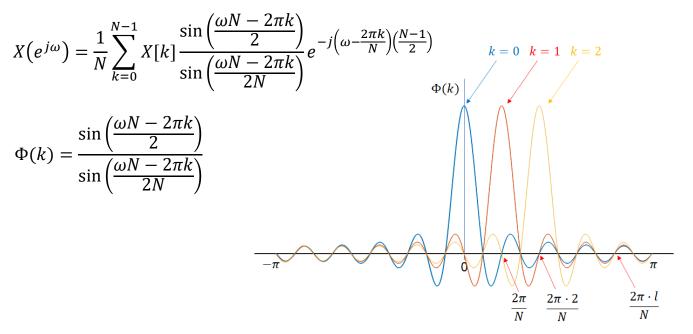
IDFT:

$$x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X[k] W_N^{-kn}, \qquad n = 0, 1, \dots, N-1.$$

DTFT vs. DFT

Q: Can we reconstruct the DTFT spectrum from the DFT?

$$x[n] \xrightarrow{DFT} X[k] \xrightarrow{?} X(e^{j\omega})$$



Digital Signal Processing by Yu Yajun @ SUSTech

19

Sampling the DTFT

 Consider a length M sequence x[n] (0 ≤ n ≤ M − 1) going through the following transforms and operations:

Sample N-point
$$x[n] \xrightarrow{\text{DTFT}} X(e^{j\omega}) \xrightarrow{N \text{ points}} Y[k] \xrightarrow{\text{IDFT}} y[n]$$

Find the relation between x[n] and y[n].

 $y[n] = \sum_{m=-\infty}^{\infty} x[n+mN], \qquad 0 \le n \le N-1$

Finite-Length Sequence

circular time-reversal

$$y[n] = x[\langle -n \rangle_N]$$
, for $0 \le n \le N-1$

• circular conjugate (anti-)symmetric sequence

$$x[n] = x^*[\langle -n \rangle_N] = x^*[\langle N - n \rangle_N]$$

$$x[n] = -x^*[\langle -n \rangle_N] = -x^*[\langle N - n \rangle_N]$$

• circular Shift

$$x_c[n] = x[\langle n - m \rangle_N]$$

circular convolution

$$y_c[n] = x[n] \otimes h[n] \triangleq \sum_{m=0}^{N-1} x[m] h[\langle n-m \rangle_N]$$

Digital Signal Processing by Yu Yajun @ SUSTech

21

22

Examples

- If $x[n] \leftrightarrow X[k]$, then $x^*[n] \leftrightarrow X^*[\langle -k \rangle_N]$
- If $x[n] \leftrightarrow X[k]$, then $X[\langle -k \rangle_N] = X[\langle N-k \rangle_N] = X^*[k]$
- If $x[n] \leftrightarrow X[k]$ for $0 \le n \le N-1$, and $g[n] = x[2n], h[n] = x[2n+1], g[n] \leftrightarrow G[k], h[n] \leftrightarrow H[k]$ for $0 \le n \le N/2-1$

then
$$X[k] = G\left[\langle k \rangle_{\frac{N}{2}}\right] + W_N^k H\left[\langle k \rangle_{\frac{N}{2}}\right], \ 0 \le k \le N-1$$

Linear Convolution using DFT

• Zero-pad x[n] by P-1 zeros

$$x_{zp}[n] = \begin{cases} x[n] & 0 \le n \le L - 1 \\ 0 & L \le n \le L + P - 2 \end{cases}$$

• Zero-pad h[n] by L-1 zeros

$$h_{zp}[n] = \begin{cases} h[n] & 0 \le n \le P - 1 \\ 0 & P \le n \le L + P - 2 \end{cases}$$

• compute the linear convolution using a circular one with length M = L+P-1

$$y[n] = h[n] * x[n] = x_{zp}[n] \otimes h_{zp}[n]$$

• implement a circular convolution using the DFT property:

$$\begin{split} h[n] * x[n] &= x_{zp}[n] \otimes h_{zp}[n] \\ &= \mathcal{IDFT} \{ \mathcal{DFT} \{ x_{zp}[n] \} \cdot \mathcal{DFT} \{ h_{zp}[n] \} \} \end{split}$$

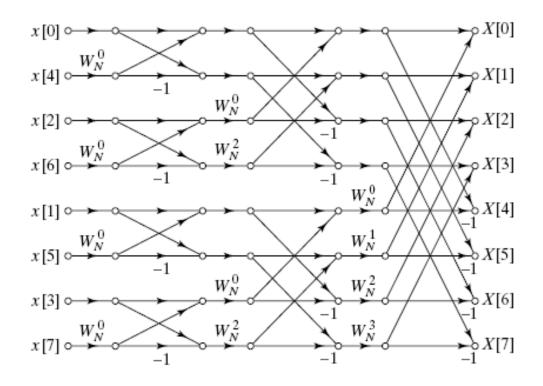
Digital Signal Processing by Yu Yajun @ SUSTech

23

Fourier-Domain Filtering

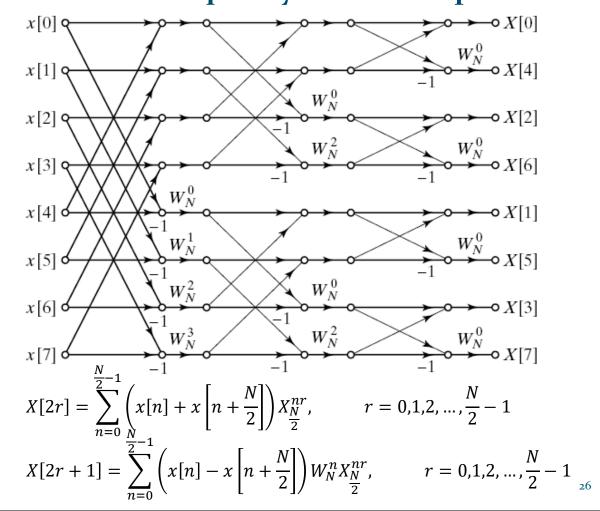
- Find the DTFT of the signal to get $X(e^{j\omega})$, multiply with $H(e^{j\omega})$ to obtain $Y(e^{j\omega})$, and find the IDTFT of $Y(e^{j\omega})$
- We can use DFT to compute $X(e^{j\omega})$ and $Y(e^{j\omega})$ at frequency values of $\omega=2\pi k/N$, for , k=0,1,...,N-1
- This approach is equivalent to the **circular convolution** of the finite-length signal x[n] and the finite-length ideal filter h[n].
- However, the ideal filter has an infinite length impulse response. Sampling the Fourier transform to create DFT samples leads to the time domain aliasing.

Decimation-in-time FFT of an 8-point DFT



$$\begin{split} X_0[k] &= X_{00} \big[\langle k \rangle_{N/4} \big] + W_{N/2}^k \, X_{01} \big[\langle k \rangle_{N/4} \big], 0 \leq k \leq N/2 - 1 \\ X_1[k] &= X_{10} \big[\langle k \rangle_{N/4} \big] + W_{N/2}^k X_{11} \big[\langle k \rangle_{N/4} \big], 0 \leq k \leq N/2 - 1 \\ & \text{Digital Signal Processing by Yu Yajun @ SUSTech} \end{split}$$

Decimation-in-frequency FFT of an 8-point DFT



Cost to compute an N-point DFT

- Using the simplified butterfly computation, the number of complex multiplications performed at each stage is reduced to N/2. Thus the total numbers become $Nv/2 = \frac{N}{2} \log_2 N$
- By excluding trivial complex multiplications with $W_N^0 = 1$ and $W_N^{N/2} = -1$, the exact count of nontrivial complex multiplications are even less, given by $\frac{N}{2}(\log_2 N 2) + 1$

Digital Signal Processing by Yu Yajun @ SUSTech

27

Z-Transform

$$\begin{array}{c}
z^n \\
\hline
 h[n] \\
\end{array}$$

$$y[n] = \sum_{k=-\infty}^{\infty} h[k]z^{n-k} = \left(\sum_{k=-\infty}^{\infty} h[k]z^{-k}\right)z^n = H(z)z^n$$

$$H(z) = Z\{h[n]\} = \sum_{n=-\infty}^{\infty} h[n]z^{-n}$$

$$\sum_{m=0}^{N} b_m y[n-m] = \sum_{m=0}^{M} a_m x[n-m], \qquad H(z) \triangleq \frac{Y(z)}{X(z)} = \frac{\sum_{m=0}^{M} a_m z^{-m}}{\sum_{m=0}^{N} b_m z^{-m}}$$

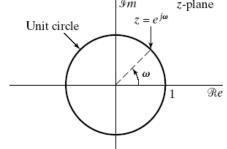
z-Transform

$$X(z) = Z\{x[n]\} = \sum_{n=-\infty}^{\infty} x[n]z^{-n}$$

• Let $z = re^{j\omega}$, then the expression reduces to

$$X(re^{j\omega}) = \sum_{n=-\infty}^{\infty} (x[n]r^{-n})e^{-j\omega n},$$

• If r = 1(i. e., |z| = 1), the z-transform reduces to DTFT.



Digital Signal Processing by Yu Yajun @ SUSTech

29

Rational z-transform

• LTI system with z-transforms represented as a rational function of z^{-1}

$$H(z) = \frac{P(z)}{D(z)} = \frac{p_0 + p_1 z^{-1} + \dots + p_{M-1} z^{-(M-1)} + p_M z^{-M}}{d_0 + d_1 z^{-1} + \dots + d_{N-1} z^{-(N-1)} + d_N z^{-N}}$$

$$H(z) = 6 \frac{\left(z - \frac{1}{3}\right)\left(z - \frac{1}{2}\right)}{z^2}$$

Region of Convergence

• Example 1: Right-sided sequence $x[n] = a^n \mu[n]$

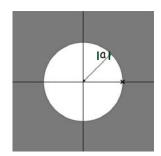
$$X(z) = \frac{1}{1 - az^{-1}}, \quad \text{for } |az^{-1}| < 1$$

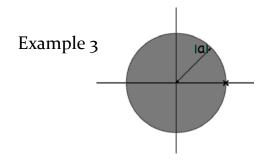
- ROC = $\{z: |z| > |a|\}$
- Example 3: Left sided sequence $x[n] = -a^n \mu[-n-1]$

$$X(z) = \frac{1}{1 - az^{-1}}, \quad \text{for } |a^{-1}z| < 1$$

• ROC = $\{|z| < |a|\}$

Example 1





Digital Signal Processing by Yu Yajun @ SUSTech

31

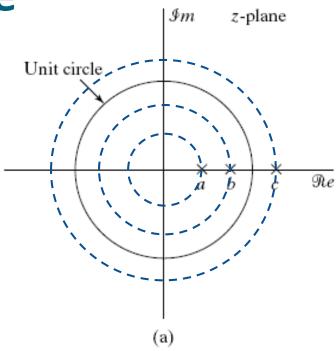
Properties of ROC

• In general, ROC of a z-transform is in a form:

$$R_{\chi^-} < |z| < R_{\chi^+}$$
, an annular region

- For right-sided sequences: $|z| > R_{\text{maxpole}}$
- For left-sided: $|z| < R_{\text{minpole}}$
- For two-sided: $R_{x^-} < |z| < R_{x^+}$ or does not exist.
- For finite duration sequences: the entire z-plane, except possibly z=0, $z=\infty$

Example



(a) A system with three poles.

Digital Signal Processing by Yu Yajun @ SUSTech

33

ROC for LTI System

- Consider the transfer function H(z) of a linear system:
 - If the system is stable, ROC must include the unit circle.
 - If the system is causal, $|z| > R_{\text{maxpole}}$
 - Therefore, a stable causal LTI system has all poles inside unit circle.

Inverse z-transform

By Inspection

$$X(z) = \frac{1}{1 - \alpha z^{-1}}, |z| < |\alpha| \to x[n] = \alpha^n \mu[n]$$

$$|z| > |\alpha| \to x[n] = -\alpha^n \mu[-n - 1]$$

Partial fraction expansion

$$X(z) = \frac{\sum_{i=0}^{M} p_i z^{-i}}{\sum_{i=0}^{N} d_i z^{-i}} = \sum_{l=0}^{M-N} \eta_l z^{-l} + \frac{P_1(z)}{D(z)} = \sum_{l=1}^{N} \left(\frac{\rho_l}{1 - \lambda_l z^{-1}}\right)$$
$$\rho_l = (1 - \lambda_l z^{-1}) X(z) \Big|_{z=\lambda_l}$$

Power series expansion

$$X(z) = \sum_{n=-\infty}^{\infty} x[n]z^{-n} = \dots + x[-2]z^2 + x[-1]z + x[0] + x[1]z^{-1} + \dots$$

Digital Signal Processing by Yu Yajun @ SUSTech

35

Frequency Response from Transfer Function

• If the ROC of H(z) includes the unit circle

$$H(e^{j\omega}) = H(z)\Big|_{z=e^{j\omega}}$$

For a real coefficient transfer function

$$|H(e^{j\omega})|^2 = H(e^{j\omega})H^*(e^{j\omega}) = H(e^{j\omega})H(e^{-j\omega})$$
$$= H(z)H(z^{-1})\Big|_{z=e^{j\omega}}$$

 A stable causal LTI system has all poles inside unit circle.

Linear Phase FIR Filter

- Its frequency response is given by $e^{-j\left(\frac{N-1}{2}\omega-\beta\right)}R(\omega)$, where $R(\omega)$ is a real function.
- The group delay is $-d\{\theta(\omega)\} = \frac{N-1}{2} = \alpha$.
- Simple example

$$H(z) = \frac{1+z^{-1}}{2} \leftrightarrow \{h[n]\} = \left\{\frac{1}{2}, \frac{1}{2}\right\} \leftrightarrow e^{-j\omega/2} \cos(\omega/2)$$

$$\uparrow$$

$$H(z) = \frac{1-z^{-1}}{2} \leftrightarrow \{h[n]\} = \{\frac{1}{2}, -\frac{1}{2}\} \leftrightarrow e^{j(\pi/2 - \omega/2)} \sin(\omega/2)$$

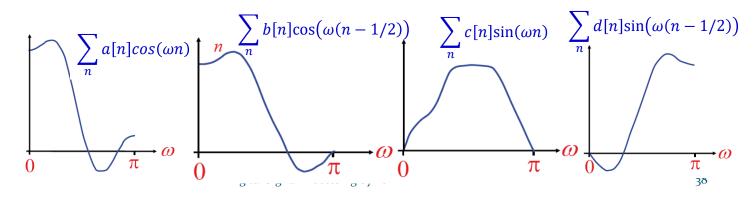
$$\uparrow$$

Digital Signal Processing by Yu Yajun @ SUSTech

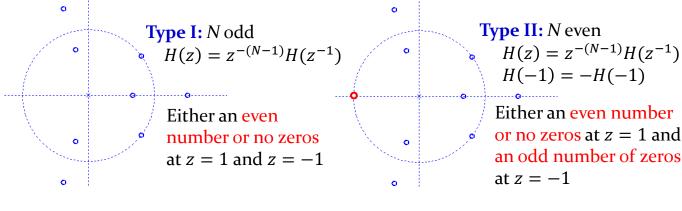
37

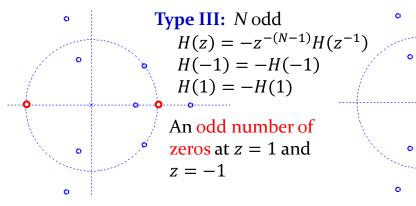
Four type linear phase FIR filters

- Classified based on the symmetry of impulse response and parity of filter length
 - Type I: Odd length, symmetric impulse response
 - Type II: Even length, symmetric impulse response
 - Type III: Odd length, anti-symmetric impulse response
 - Type IV: Even length, anti-symmetric impulse response
- Frequency response of the 4 type FIR filters



Zero-Locations of FIR Filters





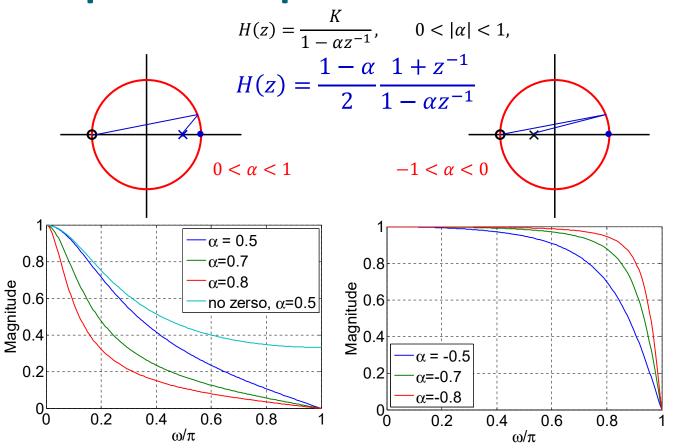
Type IV: *N* even $H(z) = -z^{-(N-1)}H(z^{-1})$ H(1) = -H(1)

An odd number of zeros at z = 1 and either an even number or no zeros at z = -1

Digital Signal Processing by Yu Yajun @ SUSTech

39

Simple and improved IIR filters

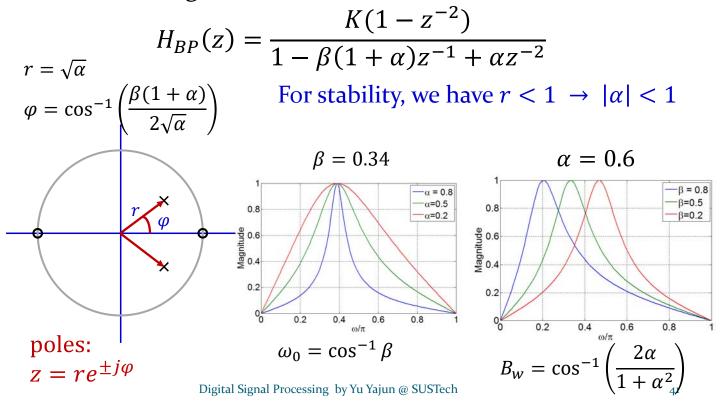


Digital Signal Processing by Yu Yajun @ SUSTech

40

Bandpass IIR Digital Filter

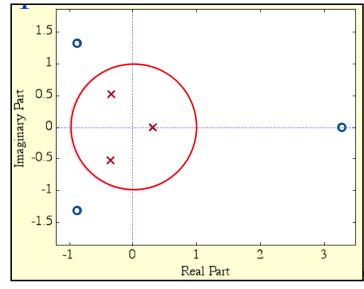
A 2nd-order general form



Allpass Filter

 An M-th order causal real-coefficient allpass transfer function is of form

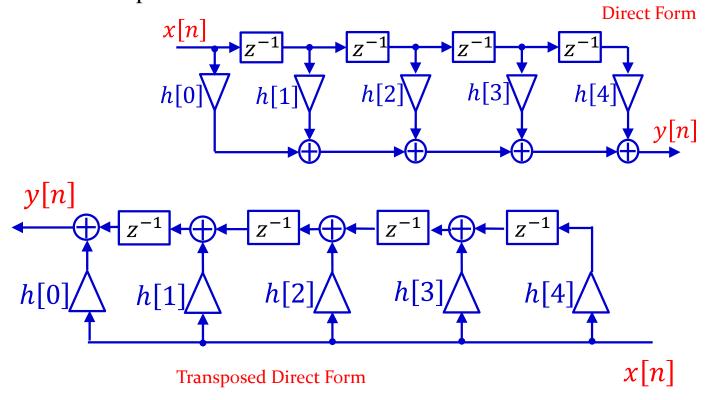
$$A_M(z) = \frac{d_M + d_{M-1}z^{-1} + \dots + d_1z^{-M+1} + z^{-M}}{1 + d_1z^{-1} + \dots + d_{M-1}z^{-M+1} + d_Mz^{-M}} = \frac{z^{-M}D_M(z^{-1})}{D_M(z)}$$

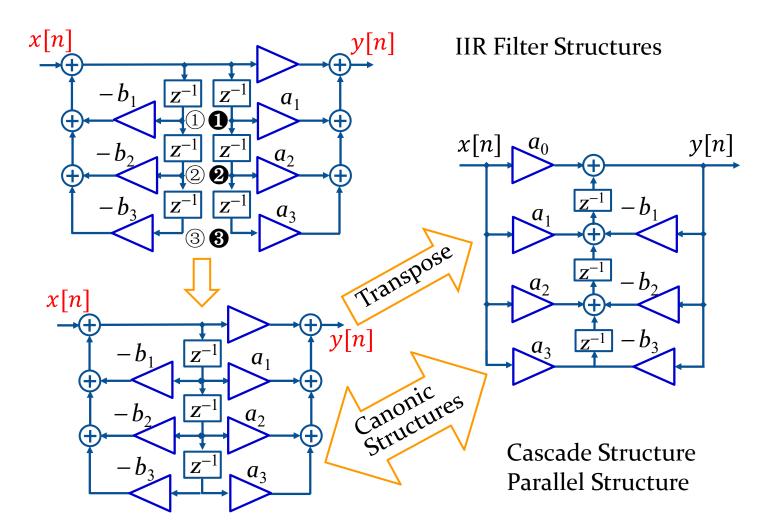


$$|A(e^{j\omega})|^2 = 1$$
, for all ω

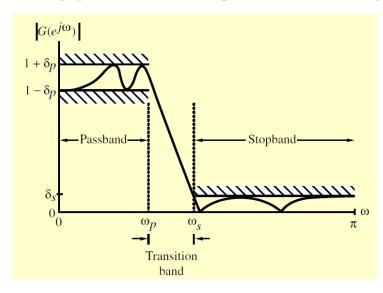
The poles of a causal stable transfer function must lie inside the unit circle in *z*-plane

- FIR Filter Structures
 - Direct form and
- $Y(z) = H(z)X(z) = \sum_{k=0}^{N-1} h[k]z^{-k}X(z)$
- Transposed direct form





Typical magnitude Specifications



• Passband:

$$\omega \le \omega_p$$
,
$$1 - \delta_p \le \left| G(e^{j\omega}) \right| \le 1 + \delta_p$$

• Stopband:

$$\omega_s \leq \omega \leq \pi$$
, $|G(e^{j\omega})| \leq \delta_s$

• Transition band:

$$\omega_p < \omega < \omega_s$$
, arbitrary response

Digital Signal Processing by Yu Yajun @ SUSTech

45

46

IIR Filter Design

Transform H(S) into the desired digital transfer function G(z)

- ✓ Imaginary ($j\Omega$) axis in the s-plane be mapped onto the unit circle of the z-plane
- ✓ Left-half of the *s*-plane be mapped inside the unit circle

Bilinear Transformation

$$s = k \left(\frac{1 - z^{-1}}{1 + z^{-1}} \right), k > 0,$$
 or $z = \frac{k + s}{k - s}$

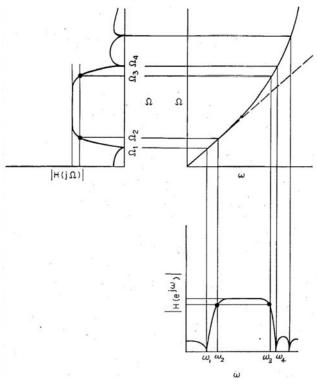
• Thus, relation between G(z) and H(s) is then given by

$$G(z) = H(s)\Big|_{s=k\left(\frac{1-z^{-1}}{1+z^{-1}}\right)}$$

Frequency Warping

- Nonlinear mapping introduces a distortion in the frequency axis called frequency warping
- Effect of warping

$$\omega = 2 \tan^{-1} \frac{\Omega}{k}$$
, or $\Omega = k \tan \left(\frac{\omega}{2}\right)$



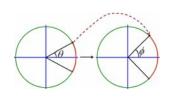
Digital Signal Processing by Yu Yajun @ SUSTech

47

Spectral Transformation

Lowpass to lowpass transformation:

$$z^{-1} = \frac{\hat{z}^{-1} - \alpha}{1 - \alpha \hat{z}^{-1}} \iff \hat{z}^{-1} = \frac{z^{-1} + \alpha}{1 + \alpha z^{-1}}$$

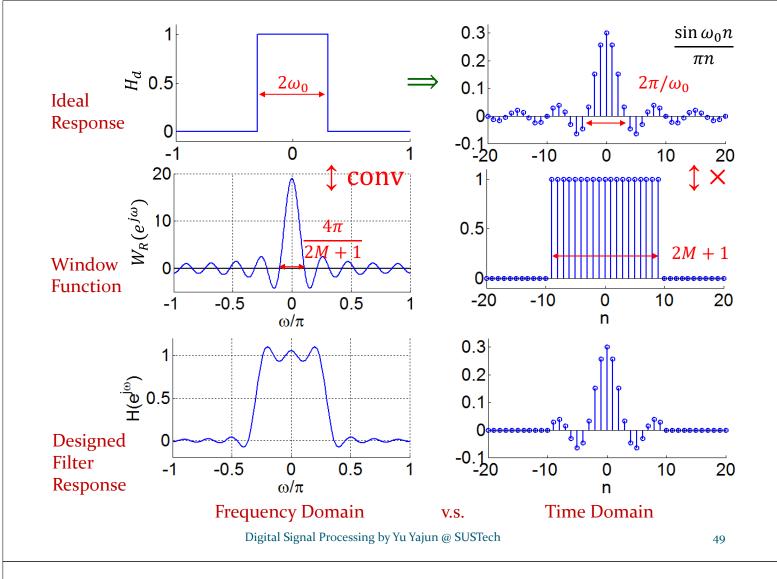


•
$$z = e^{j\omega} \implies |\hat{z}| = 1$$
; $\tan\left(\frac{\omega}{2}\right) = \left(\frac{1+\alpha}{1-\alpha}\right) \tan\left(\frac{\widehat{\omega}}{2}\right)$; $\alpha = \frac{\sin\left(\frac{\omega-\widehat{\omega}}{2}\right)}{\sin\left(\frac{\omega+\widehat{\omega}}{2}\right)}$

Lowpass to highpass transformation

$$z^{-1} = -\frac{\hat{z}^{-1} + \alpha}{1 + \alpha \hat{z}^{-1}} \iff \hat{z}^{-1} = -\frac{z^{-1} + \alpha}{1 + \alpha z^{-1}}$$

•
$$z = e^{j\omega} \implies |\hat{z}| = 1$$
; $\operatorname{ctan}\left(\frac{\omega}{2}\right) = \left(\frac{-1+\alpha}{1+\alpha}\right) \operatorname{tan}\left(\frac{\widehat{\omega}}{2}\right)$; $\alpha = -\frac{\cos\left(\frac{\omega-\widehat{\omega}}{2}\right)}{\cos\left(\frac{\omega+\widehat{\omega}}{2}\right)}$



Properties of Fixed Windows

| Type of Window | Window function | | Resultant Filter | |
|---------------------|------------------------------------|--|--|---|
| | Main Lobe Width Δ _{ML} | Relative Side- lobe Level <i>A_{sl}</i> | Minimum Stop-band Attenuation δ | Transition Bandwidth $\Delta \omega$ |
| Rectangular | $\frac{4\pi}{2M+1}$ | 13.3dB | 20.9dB | $\frac{0.92\pi}{M}$ |
| Hanning | $\frac{8\pi}{2M+1}$ | 31.5dB | 43.9dB | $\frac{3.11\pi}{M}$ |
| Hamming | $\frac{8\pi}{2M+1}$ | 42.7dB | 54.5dB | $\frac{3.32\pi}{M}$ |
| Blackman- Harris | $\frac{12\pi}{2M+1}$ | 58.1dB | 75.3dB | $\frac{5.56\pi}{M}$ |

- δ is independent from M, or ω_c , and is essentially constant.
- $\Delta \omega = \frac{c}{M}$

Course Learning Outcomes

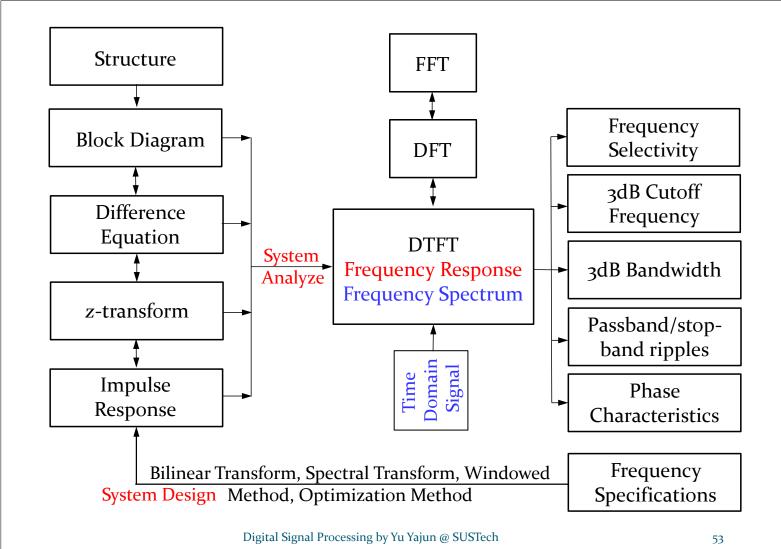
- CLO 1: I have an ability to represent discrete time signals and systems in time and frequency domain;
- CLO 2: I have an ability to understand, represent, and analyse linear time invariant discrete time systems in transformed domain by applying mathematics principles, such as differential calculus, complex variables.
- CLO 3: I have an ability to analyse digital filters and design digital filters to meet given specifications.
- CLO 4: I have an ability to use a programming language to conduct analysis and design of discrete-time signal processing systems to process discrete-time signals.

Digital Signal Processing by Yu Yajun @ SUSTech

51

Exam coverage

- Convolution, circular convolution, DTFT, DFT, IDTFT, IDFT.
- Difference equation ↔ z-transform↔ impulse response ↔ frequency response ↔ structure block diagram
- z-transform and LTI system, BIBO stable, zeros and poles, ROC.
- Frequency response: magnitude response, bandwidth, 3-db cutoff frequency.
- FIR filter, simple IIR filter, allpass filter
- minimum phase, maximum phase _______ spectrum transform.
- Bilinear transformation for the design of IIR filter
- Window method for the design of FIR filter



•
$$e^x = \cos x + j \sin x$$
, $\cos x = \frac{e^x + e^{-x}}{2}$, $\sin x = \frac{e^x - e^{-x}}{2i}$

•
$$|H(e^{j\omega})|^2 = H(e^{j\omega})H^*(e^{j\omega}) = H(z)H^*(z)|_{z=e^{j\omega}}$$

real impulse response
$$= H(e^{j\omega})H(e^{-j\omega})$$

$$= H(z)H(z^{-1})|_{z=e^{j\omega}}$$

Revisit trigonometric formulas

•
$$\Omega = 2\pi f \xrightarrow{\text{analog}} f \text{ in Hz} \xrightarrow{\text{digital}} \omega = 2\pi \frac{f}{f_s}$$

- When compute sin, cos values using calculator, note it is for degree or for radian.
- Filter length *N* or filter order *N*.
- Skip difficult questions and answer as many questions as possible in the first round answering.

Reminder:

You should manually practice the tutorial questions, instead of only reading or looking at the solutions.

DSP 2016 by Yu Yajun @ SUSTech

- 2. 微信端: 通过微信进入"南方科技大学"微信企业号--教学质量管理平台。
- 3. 在"我的任务-待评"中填写并提交本学期所选课程的所有听课评教表。操作 指南请见附件(或扫描以下二维码获取)。特别提醒:因本学期考试课程考查申请 也使用该系统,请选择待评任务中首行显示教师姓名的任务进行评价,首行显示课 程名称的任务无需填写评价(如下图所示)。



56

