

# ENGG7302: Advanced Computational Techniques in Engineering

## Lecture 5-6: Discrete-time Stochastic Processes

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# Overview of this lecture

- Correlation and Covariance
- Z-Transform and DTFT
- Difference equations
- PSD
- Filtering
- Sampling

## Correlation and covariance

- Most definitions of continuous-time stochastic process are applicable for discrete time
- Autocorrelation

$$R_X[n_1, n_2] = E[X[n_1]X^*[n_2]]$$

- Autocovariance

$$C_X[n_1, n_2] = E[\{X[n_1] - \mu_X[n_1]\}\{X[n_2] - \mu_X[n_2]\}^*]$$

- If  $X[n]$  is strict sense stationary then joint distributions are invariant to time-shift.

## Wide Sense stationary

- If  $X[n]$  is wide sense stationary then

$$R_X[n_1, n_2] = R_X[n_1 - n_2]$$

$$\mu_X[n] = \mu_X = \text{constant}$$

- In a discrete time white noise,  $X[n]$  are uncorrelated RV's.

## Example

- Given the stationary random sequence,

$x[n] = \{-1.6129, -1.2091, -0.4379, -2.0639, -0.6484\}$ , estimate the mean, the variance, the autocorrelation with a lag 0.

$$\begin{aligned}\mu &= \frac{-1.6129 - 1.2091 - 0.4379 - 2.0639 - 0.6484}{5} \\ &= -1.1944.\end{aligned}$$

## Example

- Recall, SD is how much a RV deviates from the mean. Variance is  $SD^2$ . First subtract the mean from each sample of the sequence and we get  $\{-0.4185, -0.0147, 0.7565, -0.8695, 0.5460\}$ . Then

$$\begin{aligned}\sigma^2 &= \frac{(-0.4185)^2 + (-0.0147)^2 + 0.7565^2 + (-0.8695)^2 + 0.5460^2}{5} \\ &= 0.3604.\end{aligned}$$

- Autocorrelation,

$$\begin{aligned}R_{XX}[0] &= \frac{1.6129^2 + 1.2091^2 + 0.4379^2 + 2.0639^2 + 0.6484^2}{5} \\ &= 1.787\end{aligned}$$

## Example: Autocovariance

- Autocorrelation lag of 1

$$\begin{aligned}R_{XX}[1] &= \frac{-1.6129(-1.2091) - 1.2091(-0.4379) - 0.4379(-2.0639) - 2.0639(-0.6484)}{4} \\ &= 1.1804\end{aligned}$$

- Autocovariance with a lag of 0

$$C_{XX}[0] = 0.3604$$

- Autocovariance with a lag of 1

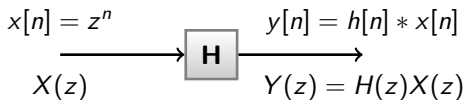
$$\begin{aligned}C_{XX}[1] &= \frac{-0.4185(-0.0147) - 0.0147(0.7565) + 0.7565(-0.8695) - 0.8695(0.5460)}{4} \\ &= -0.2844\end{aligned}$$

# Discrete-time signals and systems

- Recall that for continuous signals,  $\exp(j\omega t)$  was an eigen function of an LTI system.
- For discrete it is  $\exp(j\omega n)$ .

## Discrete-time signals and systems

- Recall that for continuous signals,  $\exp(j\omega t)$  was an eigen function of an LTI system.
- For discrete it is  $\exp(j\omega n)$ .
- Eigen functions of Discrete LTI systems have the form  $x[n] = z^n$ .



# Discrete-time Fourier Transform (DTFT) and Z

- Consider a signal  $x[n]$ . Its discrete time fourier transform is given by

$$X(w) = \sum_{n=-\infty}^{\infty} x[n] \exp(-jwn)$$

- But the Z transform of  $x[n]$  is given by:

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

- Therefore, at  $z = \exp(jw)$ , the z-transform of  $x[n]$  is the discrete time fourier transform.

## Relationship between DTFT and DFT

- OTH, a discrete fourier transform is a sampled version of DTFT

$$X(w) \Big|_{k\Delta w} = X[k] = \sum_{n=0}^{N-1} x[n] \exp\left(-\frac{jnk2\pi}{N}\right)$$

# Z-transform

- Impulse response and the Transfer Function are related via the z-transform i.e.

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

- Using the convolution property we can write

$$Y(z) = H(z)X(z)$$

$$H(z) = \frac{Y(z)}{X(z)}$$

- $H(z)$  is the transfer function. What does it tell about the digital system?

# Frequency Response

## Frequency Response

- So we had  $\exp(j\omega n)$  is the eigen function of the discrete system.
- Suppose  $x[n] = \exp(j\omega n)$ . Then

$$y[n] = x[n] * h[n]$$

$$y[n] = \sum_{k=-\infty}^{\infty} h[k] \exp\{j\omega(n-k)\}$$

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

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

$$y[n] = H(\exp(j\omega)) \exp(j\omega n)$$

# Frequency Response

- So for the eigen function  $\exp(j\omega n)$ , the eigen value is  $H(\exp(j\omega))$ .
- The eigen value is called the frequency response of the system

$$\begin{aligned}H(\exp(j\omega)) &= H_R(\exp(j\omega)) + jH_I(\exp(j\omega)) \\ &= |H(\exp(j\omega))| \exp(j\angle H(\exp(j\omega)))\end{aligned}$$

## Power Spectral Density

- Recall, for continuous processes PSD was given by:

$$\begin{aligned} S_x(f) &= \lim_{T \rightarrow \infty} \frac{1}{T} E \left[ \left| \int_{-T/2}^{T/2} X(t) \exp(-j2\pi ft) dt \right|^2 \right] \\ &= \lim_{T \rightarrow \infty} \frac{1}{T} E[|X(f)|^2] \end{aligned}$$

- For discrete, PSD can be defined in terms of the DTFT.

$$\begin{aligned} S_X(\exp(j\omega)) &= \lim_{N \rightarrow \infty} \frac{1}{2N+1} E[|X_N(\exp(j\omega))|^2] \\ X_N(\exp(j\omega)) &= \sum_{n=-\infty}^{\infty} x[n] \exp(-j\omega n) \end{aligned}$$

# Power Spectral Density

- For a WSS process

$$R_X[m] \stackrel{DTFT}{\leftrightarrow} S_X(\exp(j\omega))$$

- Inverse DTFT can be used to obtain  $R_X[m]$  from  $S_X(\exp(j\omega))$

$$R_X[m] = \int_{-\pi}^{\pi} S_X(\exp(j\omega)) \exp(j\omega m) d\omega$$

# Difference equations for LTI

## LTI systems governed by difference equations

- The inputs and outputs of a LTI system can be written via a difference equation

$$\sum_{k=0}^N a_k y[n-k] = \sum_{k=0}^M b_k x[n-k]$$

- Taking Z-transforms

$$Y(z) \sum_{k=0}^N a_k z^{-k} = X(z) \sum_{k=0}^M b_k z^{-k}$$
$$\frac{Y(z)}{X(z)} = \frac{\sum_{k=0}^M b_k z^{-k}}{\sum_{k=0}^N a_k z^{-k}}$$

- $H(z)$  becomes the transfer function.

## FIR filters

- Finite Impulse response filters
- Represented by the difference equation.

$$y[n] = b_0x[n] + b_1x[n-1] + \dots + b_kx[n-k]$$

$$y[n] = \sum_{k=0}^M b_kx[n-k]$$

- $b_k$  become the coefficients of the filters.
- Feedback stage absent
- Stable filters

## Transfer function: FIR filters

- We had

$$y[n] = \sum_{k=0}^M b_k x[n - k]$$

- Taking Z transforms we have

$$Y(z) = X(z) \left( \sum_{k=0}^M b_k z^{-k} \right)$$

$$H(z) = \frac{Y(z)}{X(z)} = \sum_{k=0}^M b_k z^{-k}$$

## Moving average process

- Impulse response of MA process is given by

$$h[n] = \frac{1}{M_2 + 1} \sum_{k=0}^{M_2} \delta[n - k]$$

- Impulse response is finite. Now, the output is  $h[n] * x[n]$ . Therefore

$$y[n] = \frac{1}{M_2 + 1} \sum_{k=0}^{M_2} x[n - k]$$

## IIR filters

- Infinite Impulse response filters
- Can be modelled using a difference equation given by

$$y[n] = \sum_{k=1}^N a_k y[n-k] + \sum_{k=0}^M b_k x[n-k]$$

$a_k$  and  $b_k$  become the coefficients of the filters.

- Feedback stage is present

## Transfer function: IIR filters

- We had

$$y[n] = \sum_{k=1}^N a_k y[n-k] + \sum_{k=0}^M b_k x[n-k]$$

- Taking Z transforms we have

$$Y(z) \left( 1 - \sum_{k=1}^N a_k z^{-k} \right) = X(z) \left( \sum_{k=0}^M b_k z^{-k} \right)$$

$$H(z) = \frac{Y(z)}{X(z)} = \frac{\sum_{k=0}^M b_k z^{-k}}{1 - \sum_{k=1}^N a_k z^{-k}}$$

# Accumulator

- Impulse response of accumulator is given by

$$h[n] = \sum_{k=-\infty}^n \delta[k]$$

- Impulse response is infinite. Now, the output is  $h[n] * x[n]$ . Therefore

$$y[n] = \sum_{k=-\infty}^n x[k]$$

## Accumulator: Difference equation

$$y[n] = \sum_{k=-\infty}^n x[k]$$

$$y[n-1] = \sum_{k=-\infty}^{n-1} x[k]$$

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

$$y[n] = x[n] + y[n-1]$$

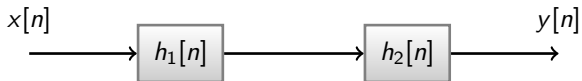
- This becomes an IIR filter of the form

$$y[n] = \sum_{k=1}^N a_k y[n-k] + \sum_{k=0}^M b_k x[n-k]$$

# Discrete time Processes

- If  $N = 0$ , and  $M > 0$  becomes a Moving Average process.
- If  $M = 0$ , and  $N > 0$  becomes an Auto regressive process.
- If  $M > 0$  and  $N > 0$ , it becomes an autoregressive moving average process ARMA(M,N).

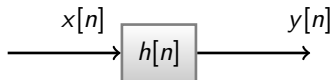
## Cascaded systems



- The Impulse response of the overall system is given by

$$h[n] = h_1[n] * h_2[n]$$

## Filtering Stochastic processes



- When the input to a LTI system is WSS, the autocorrelations and PSDs of input  $X[n]$  and output  $Y[n]$  are related:

$$R_Y[m] = h[m] * h^*[-m] * R_X[m]$$

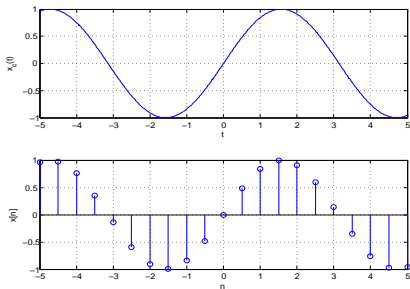
$$S_Y(\exp(j\omega)) = |H(\exp(j\omega))|^2 S_X(\exp(j\omega))$$

# Sampling

# Sampling

- An analogue signal  $x_c(t)$  is sampled to produce a digital or discrete-time signal  $x[n]$  such that

$$x[n] = x_c(nT_s)$$



## Nyquist criteria

- To avoid aliasing, a bandlimited signal of bandwidth  $w_B$  rad/s must be sampled at a frequency of  $2w_B$  rad/s That is  $w_s > 2w_B$ .
- This is basis of operation for A/D and D/A convertors.

## Sampling: Example

- For the signal  $X(w) = u(w) - u(w - w_0)$  calculate the sampling period.
- For the signal  $X(w) = u(w + w_0) - u(w - w_0)$  calculate the sampling period.