Review of Discrete-Time System

Electrical & Computer Engineering University of Maryland, College Park

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Contact: minwu@umd.edu. Updated: August 28, 2012.

- Discrete-time signals: $\delta(n)$, u(n), exponentials, sinusoids
- Transforms: ZT, FT
- Discrete-time system: LTI, causality, stability, FIR & IIR system
- Sampling of a continuous-time signal
- Discrete-time filters: magnitude response, linear phase
- Time-frequency relations: FS; FT; DTFT; DFT

Homework: Pick up a DSP text and review.

0.1 Basic Discrete-Time Signals

unit pulse (unit sample)

$$\delta[n] = \begin{cases} 1 & n = 0 \\ 0 & otherwise \end{cases}$$



2 unit step
$$u[n] = egin{cases} 1 & n \geq 0 \ 0 & otherwise \end{cases}$$



Questions:

- What is the relation between $\delta[n]$ and u[n]?
- How to express any x[n] using unit pulses? $x[n] = \sum_{k=-\infty}^{\infty} x[k]\delta[n-k]$

$\S0.1$ Basic Discrete-Time Signals

Sinusoids and complex exponentials $x_1[n] = A \cos(\omega_0 n + \theta)$ $x_2[n] = a e^{j\omega_0 n}$ •

 x₂[n] has real and imaginary parts; known as a single-frequency signal.

Exponentials



Questions:

Is $x_1[n]$ a single-frequency signal? Are $x_1[n]$ and $x_2[n]$ periodic?

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ENEE630 Lecture Part-0 DSP Review

The **Z-transform** of a sequence x[n] is defined as

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

In general, the region of convergence (ROC) takes the form of $R_1 < |z| < R_2$.

E.g.:
$$x[n] = a^n u[n]$$
: $\mathbb{X}(z) = \frac{1}{1-az^{-1}}$, ROC is $|z| > |a|$.

The same $\mathbb{X}(z)$ with a different ROC |z| < |a| will be the ZT of a different $x[n] = -a^n u[-n-1]$.

The **Fourier transform** of a discrete-time signal x[n]

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

- Often known as the Discrete-Time Fourier Transform (DTFT)
- If the ROC of $\mathbb{X}(z)$ includes the unit circle, we evaluate $\mathbb{X}(z)$ with $z = e^{j\omega}$, we call $\mathbb{X}(e^{j\omega})$ the Fourier Transform of x[n]
- The unit of frequency variable ω is radians
- $\mathbb{X}(\omega)$ is periodic with period 2π
- The inverse transform is $x[n] = \frac{1}{2\pi} \int_0^{2\pi} \mathbb{X}(\omega) e^{j\omega n} d\omega$

Question: What is the FT of a single-frequency signal $e^{j\omega_0 n}$?

- Since the ZT of a^n does not converge anywhere except for a = 0, the FT for $x[n] = e^{j\omega_0 n}$ does not exist in the usual sense.
- But we can unite its FT as $2\pi\delta_a(\omega \omega_0)$ for ω in the range between $0 < \omega < 2\pi$ and periodically repeating, by using a Dirac delta function $\delta_a(\cdot)$.

$\S0.2$ (3) Parseval's Relation

Let $\mathbb{X}(\omega)$ and $\mathbb{Y}(\omega)$ be the FT of x[n] and y[n], then

$$\sum_{n=-\infty}^{\infty} x[n] y^*[n] = \frac{1}{2\pi} \int_0^{2\pi} \mathbb{X}(\omega) \mathbb{Y}^*(\omega) d\omega.$$

i.e., the inner product is preserved (except a multiplicative factor):

$$<$$
 x[n], y[n] >=< X(ω), Y(ω) > $\cdot rac{1}{2\pi}$

• If x[n] = y[n], we have $\sum_{n=-\infty}^{\infty} |x[n]|^2 = \frac{1}{2\pi} \int_0^{2\pi} |\mathbb{X}(\omega)|^2 d\omega$

Parseval's Relation suggests that the energy of x[n] is conserved after FT and provides us two ways to express the energy.

Question: Prove the Parseval's Relation.

(Hint: start with applying the definition of inverse DTFT for x[n] to LHS)

Question 1: How to characterize a general system?

$\S0.3$ (2) Linear Time-Invariant Systems



Linearity

(input)
$$a_1x_1[n] + a_2x_2[n] \rightarrow$$
 (output) $a_1y_1[n] + a_2y_2[n]$

If the output in response to the input $a_1x_1[n] + a_2x_2[n]$ equals to $a_1y_1[n] + a_2y_2[n]$ for every pair of constants a_1 and a_2 and every possible $x_1[n]$ and $x_2[n]$, we say the system is linear.

Shift-Invariance (Time-Invariance)

(input) $x_1[n - N] \rightarrow$ (output) $y_1[n - N]$

i.e., The output in response to the shifted input $x_1[n - N]$ equals to $y_1[n - N]$ for all integers N and all possible $x_1[n]$.

An LTI system is both linear and shift-invariant. Such a system can be completely characterized by its impulse response h[n]:

(input) $\delta[n] \rightarrow$ (output) h[n]

Recall all x[n] can be represented as $x[n] = \sum_{m=-\infty}^{\infty} x[m]\delta[n-m]$ \Rightarrow By LTI property:

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

$\S0.3$ (4) Input-Output Relation of LTI Systems

The input-output relation of an LTI system is given by a convolution summation:

$$\underbrace{y[n]}_{\text{output}} = h[n] * \underbrace{x[n]}_{\text{input}} = \sum_{m=-\infty}^{\infty} x[m]h[n-m] = \sum_{m=-\infty}^{\infty} h[m]x[n-m]$$

• The transfer-domain representation is $\mathbb{Y}(z) = H(z)\mathbb{X}(z)$, where

$$H(z) = \frac{\mathbb{Y}(z)}{\mathbb{X}(z)} = \sum_{n=-\infty}^{\infty} h[n] z^{-n}$$

is called the transfer function of the LTI system.

A major class of transfer functions we are interested in is the rational transfer function:

$$H(z) = \frac{B(z)}{A(z)} = \frac{\sum_{k=0}^{N} b_k z^{-k}}{\sum_{m=0}^{N} a_m z^{-m}}$$

- $\{a_n\}$ and $\{b_n\}$ are finite and possibly complex.
- *N* is the order of the system if B(z)/A(z) is irreducible.

$\S0.3$ (6) Causality

The output doesn't depend on future values of the input sequence. (important for processing a data stream in real-time with low delay)

An LTI system is causal iff $h[n] = 0 \forall n < 0$.

Question: What property does H(z) have for a causal system?

Pitfalls: note the spelling of words "casual" vs. "causal".

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- A causal *N*-th order finite impulse response (FIR) system can have its transfer function written as $H(z) = \sum_{n=0}^{N} h[n]z^{-n}$
- A causal LTI system that is not FIR is said to be IIR (infinite impulse response).

e.g. exponential signal
$$h[n] = a^n u[n]$$
:
its corresponding $H(z) = \frac{1}{1-az^{-1}}$.

BIBO: bounded-input bounded-output

An LTI system is BIBO stable iff $\sum_{n=-\infty}^{\infty} |h[n]| < \infty$

i.e. its impulse response is absolutely summable.

This sufficient and necessary condition means that ROC of H(z) includes unit circle: $\therefore |H(z)|_{z=e^{j\omega}} \leq \sum_n |h[n]| \times 1 < \infty$

If H(z) is rational and h[n] is causal (s.t. ROC takes the form |z| > r), the system is stable **iff** all poles are inside the unit circle (such that the ROC includes the unit circle).

We use the subscript "a" to denote continuous-time (analog) signal and drop the subscript if the context is clear.

The Fourier Transform of a continuous-time signal
$$x_a(t)$$

$$\begin{cases}
\mathbb{X}_a(\Omega) \triangleq \int_{-\infty}^{\infty} x_a(t) e^{-j\Omega t} dt & \text{``projection''} \\
x_a(t) = \frac{1}{2\pi} \int_{-\infty}^{\infty} \mathbb{X}_a(\Omega) e^{j\Omega t} d\Omega & \text{``reconstruction''}
\end{cases}$$

- $\Omega = 2\pi f$ and is in radian per second
- f is in Hz (i.e., cycles per second)

$\S0.4(2)$ Sampling

Consider a sampled signal $x[n] \triangleq x_a(nT)$.

• T > 0: sampling period; $2\pi/T$: sampling (radian) frequency

The Discrete Time Fourier Transform of x[n] and the Fourier Transform of $x_a(t)$ have the following relation:

$$\mathbb{X}(\omega) = \frac{1}{T} \sum_{k=-\infty}^{\infty} \mathbb{X}_{a}(\Omega - \frac{2\pi k}{T})|_{\Omega = \frac{\omega}{T}}$$



$\S0.4(3)$ Aliasing

If X_a(Ω) = 0 for |Ω| ≥ π/T (i.e., band limited), there is no overlap between X_a(Ω) and its shifted replicas.

Can recover $x_a(t)$ from the sampled version x[n] by retaining only one copy of $\mathbb{X}_a(\Omega)$. This can be accomplished by interpolation/filtering.

• Otherwise, overlap occurs. This is called aliasing.



Reference: Chapter 7 "Sampling" in Oppenheim et al. Signals and Systems Book

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$\S0.4$ (4) Sampling Theorem

• Let $x_a(t)$ be a band-limited signal with $\mathbb{X}_a(\Omega) = 0$ for $|\Omega| \ge \sigma$, then $x_a(t)$ is uniquely determined by its samples $x_a(nT)$, $n \in \mathbb{Z}$,

if the sampling frequency $\Omega_s \triangleq 2\pi/T$ satisfies $\Omega_s \ge 2\sigma$.

• In the ω domain, 2π is the (normalized) sampling rate for any sampling period T.

Thus the signal bandwidth can at most be π to avoid aliasing.



§0.5 Discrete-Time Filters

• A Digital Filter is an LTI system with rational transfer function. The frequency response $H(e^{j\omega})$ specifies the properties of a filter:

 $H(\omega) = |H(\omega)|e^{j\phi(\omega)}$

 $|H(\omega)|$: magnitude response $\phi(\omega)$: phase response

2 Magnitude response determines the type of filters:



③ Linear-phase filter: phase response $\phi(\omega)$ is linear in ω .

- Linear phase is usually the minimal phase distortion we can expect.
- A real-valued linear-phase FIR filter of length N normally is either symmetric h[n] = h[N n] or anti-symmetric h[n] = -h[N n].

§0.6 Relations of Several Transforms (answer)

TRANSFORM	TIME-DOMAIN (Analysis)	FREQUENCY-DOMAIN (Synthesis)
Fourier Series (<i>FS</i>)		
Fourier Transform (FT)		
Discrete-Time Fourier Transform (DTFT)		
Discrete Fourier Transform (DFT)		

0.6 Relations of Several Transforms

TRANSFORM	TIME-DOMAIN (Analysis)	FREQUENCY-DOMAIN (Synthesis)
Fourier Series (<i>FS</i>)	$x(t) \text{ continuous periodic}$ $X_n = \frac{1}{T} \int_{-\frac{T}{2}}^{+\frac{T}{2}} x(t) e^{-j2\pi nt/T} dt$	$X_n \text{ discrete aperiodic} \\ x(t) = \sum_{n=-\infty}^{+\infty} X_n e^{j2\pi nt/T}$
Fourier Transform (FT)	$egin{aligned} & x(t) ext{ continuous aperiodic} \ & X(\Omega) = \int_{-\infty}^{+\infty} x(t) e^{-j\Omega t} dt \ & (ext{ or in } f ext{ where } \Omega = 2\pi f) \end{aligned}$	$X(\Omega) \text{ continuous aperiodic} x(t) = \frac{1}{2\pi} \int_{-\infty}^{+\infty} X(\Omega) e^{j\Omega t} d\Omega$
Discrete-Time Fourier Transform (DTFT)	x[n] discrete aperiodic $X(\omega) = \sum_{n=-\infty}^{+\infty} x[n]e^{-j\omega n}$	$X(\omega)$ continuous periodic $x[n] = rac{1}{2\pi} \int_{-\pi}^{+\pi} X(\omega) e^{j\omega n} d\omega$
Discrete Fourier Transform (DFT)	$\begin{split} x[n] & \text{discrete periodic} \\ X[k] &= \sum_{n=0}^{N-1} x[n] W_N^{kn} \\ (\text{where } W_N^{kn} = e^{-j2\pi kn/N}) \end{split}$	$X[k] \text{ discrete periodic}$ $x[n] = \frac{1}{N} \sum_{k=0}^{N-1} X[k] W_N^{-kn}$

Question 1: How to characterize a general system?

Ans: by its input-output response (which may require us to enumerate all possible inputs, and observe and record the corresponding outputs)

Question 2: Why are we interested in LTI systems?

Ans: They can be completely characterized by just one response - the response to impulse input