SFWR ENG 2MX3 Summary

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*Math objects made using* [*MathType*](http://www.dessci.com/en/products/mathtype/)*; graphs made using* [*Winplot*](http://math.exeter.edu/rparris/winplot.html)*.*

CTRL-F (?) to find locations which need to fixed

Bonus: Order of writing symbols, such as exponents: digits-i-variables

# Discrete Systems

If there’s no x in the equation, there must be a mistake because there’s no input.

## Deltas

2 types of deltas (δ):

**Kronecker Delta**: discrete time domain, 

**Dirac Delta**: continuous time domain, δ(t) = ∞, δ(else) = 0, ∫δ(0) = 1

Discrete manipulates functions of n.

Continuous manipulates functions of t.

y = output

**Causal system**: system = 0 before impulse

Only use cosine curves; represent sine curves with a phase shift.

y(n) and x(n) are not states

If you have a y(n−2) state, but not a y(n−1) state, include a y(n−1) state anyways.

**SISO**: single-input, single-output

**MIMO system**: Multiple Inputs Multiple Outputs.

All the systems we examine are “zero-state”. Initial state is zero, s(0) = 0



## Discrete Systems

* More precise, more theoretical
* Discrete input signals
* Easier to process because it’s digital

## Continuous System

* e.g. x(t) = 2cos(t)
* Analog

# Difference equation

## Difference equation to State Space equation

The first sum is manipulation of input signals and the second part is manipulation of output signals.

, which is interchangeable with:



Euler’s equations:



## [A, B, C, D] representation

(?)

**Linear Time Invariant** (LTI) System: you may flip order of system and delay (?)

If

Determine the equation of the next state and re-write the output in matrix form.

Next state:

Output:

## Compound Interest

*n*: billing period

*α*: interest

*x*: deposits



Similar to an impulse, let’s model someone putting money in a bank and leaving it there forever:





So you can interpolate: 

# Impulse Response

When the definition of impulse response has not been specified, it can be modeled by the [Kronecker delta function](#_Deltas).

## FIR

**Finite Impulse Response** (FIR) system: A system that has an impulse response that has a ﬁnite duration (is zero at a ﬁnite time). One way of seeing if it is finite is if it has no y’s in it,

### e.g.)

y(n) = x(n) + x(n−2)

How we model systems in this course is using a **state space table**. This table analyzes what the output of the given system is when you are given an impulse. The prof usually leaves the labels blank, but this time, I’ll leave them in there. Next time, however, they won’t be there:

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
| Time | 0 | 1 | 2 | 3 | 4 |
| Input [x(n)] | 1 | 0 | 0 | 0 | 0 |
| Output [y(n)] | 1 | 0 | 1 | 0 | 0 |

From this table, you can tell that it’s an FIR system, since the output is 0 after t = 2.

## IIR

**Inﬁnite Impulse Response** (IIR) system. A system that has an impulse response that has an inﬁnite duration (continues to respond indeﬁnitely). IIR systems generally have *y* on both sides of the equation

### e.g.)

y(n) = x(n) + 0.5y(n−1)

|  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- |
|  | 1 | 0 | 0 | 0 | 0 |
|  | 1 | ½ | ¼ | ⅛ | 1/16 |

Notice how the system doesn’t end? This is why it is called an IIR system. Also, notice how the labels are gone? Get used to it and imagine them there every time you make a table like this.

## Impulse Response

**Impulse Response** is notated as h(*n*). It is the output, y(n), when the input, x(n) is a combination of shifts of the Kronecker delta function.

Remember [ABCD representation](#_[A,_B,_C,)?



# Convolution



If something appears to have the convolution equation, it will be convolution and thus, you may use its properties.

## Impulse response to difference equation

Sometimes your impulse response may be specified. There are 2 ways of converting your impulse response into the [difference equation](#_Difference_equations):

* [Convolution Equation method](#_Equation_method)
* [Table method](#_Table_method)

### e.g.)

We’re going to use the same example to demonstrate both methods



x(n) = δ(n) + δ(n−2)

### Convolution Equation method



Let’s plug in some values of *n*:



However, since h(anything < 0) = 0, we only get output when i = 0:









We’re supposed to stop at this point for some reason.

### Table method

|  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- |
| x(n) 🡪 | 1 | 0 | 1 | 0 | 0 | 0 |
| h(0) | 1 | 2 | 3 | 0 | 0 | 0 |
| h(1) | 0 | 0 | 0 | 0 | 0 | 0 |
| h(2) | 0 | 0 | 1 | 2 | 3 | 0 |
| y(n) | 1 | 2 | 4 | 2 | 3 | 0 |

We’re supposed to stop at this point for some reason.

### Kemal’s Method

Someone else probably thought of this before me, but there’s a faster way of directly converting your impulse response to your difference equation:



# Frequency/Time Domain

Upper-case variable: frequency domain

Lower-case variable: time domain



Remember that *upper-case* variables (frequency-domain) are *multiplied*: , and *lower-case* variables (time-domain) are *convoluted*: 

 (?)

### e.g.

If you wanted to normalize  (should be divisible by 2)



## Frequency Response



e.g. y(n) + y(n−1) = 2x(n) − 5x(n−2)



Factor everything out to isolate H(ω):



Formula: 



Table: (the argument will always be negative because you can’t look ahead of time)

|  |  |  |  |
| --- | --- | --- | --- |
| ω | H(ω) | ||H(ω)|| |  |
| 0 | − 3/2 | 3/2 | −π |
| π/2 |  |  |  |
| π |  |  |  |





# Filter a signal



**Sawtooth signals** are periodic, so you have to use the Fourier Series on it.



# State Space

this vs ABCD (?)

S(n+1) = A∙s(n) + B∙x(n)

y(n) = C∙s(n) + D∙x(n)

e.g. y(n) = x(n) + 2x(n−2)

let S1(n) = x(n−2)

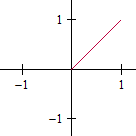
<insert matrices>

# Complex Numbers

Draw a graph where the x values represent the real dimension and the y values represent the complex dimension.

Refer to Euler’s equations a lot (in the difference equations section).

e.g. 



Magnitude ==

Divide by the magnitude on the unit circle. Thus, the point is: 

Know your special triangles.

= argument or degree

Now, put it into the form: 



= gain

= phase shift

# Impulse Response to Frequency Response

|  |  |  |
| --- | --- | --- |
|  | **FS (Fourier Series)** | **FT (Fourier Transform)** |
| DT (Discrete Time) |  | DTFT |
| CT (Continuous Time) |  | Summary:  CTFT |

**Forward transform**: time → frequency domain (look for negative exponents on e)

Transform−1: frequency → time domain (look for positive exponents on e)

For discrete frequency only: ω = ω0∙k

For continuous frequency only: ω = ω0∙t

If *h* is a function of time, then CTFT; if not, DTFT. i.e. h(t), do CTFT; h(n), do DTFT

The impulse response, *h*(*n*), of a system is the *y*(*n*) when *x*(*n*) is an impulse: *δ*(*n*).

# Step Response

Removal of frequency component at a point means H(point) = 0, so you just pretend the value is a root at that time. i.e. 

# DTFT versus DFT

The DFT is actually the most popular transform, because there's an efficient algorithm for it called the **Fast Fourier Transform** (FFT). It’s the same as the DTFS, but with different scaling. See the difference:

|  |  |
| --- | --- |
| **DTFS** | **DFT/FFT** |
|  |  |

Because the [forward] DFT doesn't divide out the number of points, *p*, using a higher sampling rate will make the DFT larger for the same input. This is why it's actually better to scale the DFT output by 1/p anyways when looking at it afterwards.

# Fourier Series

Midterm 1-Question 3:



Given . Your coefficient of k should be a multiple of 2π, so make p=4.

calculate your x(n) values before your \_\_ values.

x(n) = 2sin(π/2 n), so you get 0, 2, 0, and −2

Xk’s are the fourier coefficients.;jm

Alternative method:

X(t) = cos(t) + sin(2t) => convert to complex exponentials: , so

k = 1,−1 =>

Question 1, midterm 2

X(n) = 1, 1, 0, 0 →

N = p, so

Fourier Transform

Test 2, Question 3

CTFT:

Recall that integral of a dirac is 1.

# Time and Frequency Domain

[Gif demonstration of what the frequency domain is](https://upload.wikimedia.org/wikipedia/commons/7/72/Fourier_transform_time_and_frequency_domains_%28small%29.gif).



# Z and Laplace Transform

If you aren’t given a value for x, assume it’s a delta function.

e.g. Practice 3, question 3



1) Laplace:



2) Isolate for Y:



Poles of the transfer function are the coefficients of X(s) function.

Assume x(t) = δ(t), so X(s) = 1

3)



4) Quadratic formula OR factor to find the poles



This region of convergence **doesn’t** include the imaginary axis.

The region of convergence of *H*(*s*) must include the imaginary axis in order for the system to be stable.

*H*(*s*) is:

1. Called the **transfer function**, and is
2. The Laplace transform of the impulse response, h(t).

## Z-transform

Kind of like the laplace transform but for discrete functions.



# Block Diagrams

ALWAYS has 2 inputs, one output. If anything else, ignore. Output = sum of inputs.

When a wire splits up into 2, multiply outputs of the junction by the input.

When a wire encounters a delay or an amplifier or anything without a junction, multiply your current value by the object.

Feedback loop: jump ahead, then look back. Also,



D = delay = e−iω

*Bonus*: D is technically a function, so keep it on the left of things that depend on it, similar to d/dt