

## Section A

2022 1(a)

A discrete time signal is a signal defined only at discrete points in time.

These signals are obtained by sampling a continuous-time signal at uniform time intervals.

A digital signal is a discrete-time signal with both time and amplitude quantized. It is the result of sampling and quantization.

The frequency of two sinusoidal signals

$$f_1 = 10 \text{ Hz}$$

$$f_2 = 50 \text{ Hz}$$

$$F_s = 40 \text{ Hz}$$

To find the aliased frequency  
The signal

$$x_1(t) = \cos 2\pi(10)t$$

$$x_2(t) = \cos 2\pi(50)t$$

The corresponding discrete time signal are:

$$x_1(n) = \cos 2\pi \left(\frac{10}{40}\right)n = \cos \frac{\pi}{2} n$$

$$x_2(n) = \cos 2\pi \left(\frac{50}{40}\right)n = \cos \frac{5\pi}{2} n$$

$$\cos \frac{5\pi}{2} n = \cos \left(2\pi f + \frac{\pi n}{2}\right) = \cos \frac{\pi n}{2}$$

Hence  $x_2(n) = x_1(n)$ . The sinusoidal signal signals are identical and consequently indistinguishable.

We say that the frequency  $f_2 = 50\text{Hz}$  is an alias of the frequency  $f_1 = 10\text{Hz}$  at the sampling rate of  $40\text{Hz}$ .

Aliasing occurs when a signal is sampled at a rate lower than twice the highest frequency component. Hence  $\frac{f_s}{2} = 20\text{Hz}$

$$f_1 = 10\text{Hz} \quad f_2 = 50\text{Hz}$$

$$(f_2) > \frac{f_s}{2} \geq f_1$$

WEEK 1 (b)

Linear Time Invariant Systems  
are a special and very  
important class of systems  
in signal processing and control  
theory. The properties are

1. Linearity

2. Time Invariance

3. Stability

4. Causality

5. Linearity

An LTI system obeys the principle of Additivity and

homogeneity.

Additivity:

If the system's response to  $x_1(t)$  is  $y_1(t)$  and  $x_2(t)$  is  $y_2(t)$

then the response to  $x_1(t) + x_2(t)$  is

$$y(t) = y_1(t) + y_2(t)$$

Homogeneity:

If the response to  $x(t)$  is  $y(t)$  then the response to  $a \cdot x(t)$  is  $a \cdot y(t)$

### 2. Time Invariance:

A system is time-invariant if its behaviour does not change over time.

If the input is delayed by  $t_0$ , the output is also delayed by  $t_0$

$$x(t) \rightarrow y(t) \Rightarrow x(t-t_0) \rightarrow y(t-t_0)$$

### 3. Stability:

If  $|x(t)| < M$  for all  $t$ , then  $|y(t)| < N$  for some constant  $N$ .

Condition:

$$\sum_{n=-\infty}^{\infty} |h(n)| < \infty \quad (\text{discrete time})$$

$$\int_{-\infty}^{\infty} |h(t)| dt < \infty \quad (\text{continuous time})$$

IT is more difficult

## 9: causality:

A system is causal if the output at any time depends only on present and past inputs.

For an LTI system, this means:

$$y(t) = 0 \text{ for } t < 0$$

Given system

$$y(n) = n x(n) \quad [\text{from Q2.5 example}]$$

Two input sequence  $x_1(n)$  and  $x_2(n)$

$$y_1(n) = n x_1(n)$$

$$y_2(n) = n x_2(n)$$

$$\begin{aligned} \cancel{y_3(n)} &= \cancel{n} [a_1 x_1(n) + a_2 x_2(n)] \\ &= n [a_1 x_1(n) + a_2 x_2(n)] \\ &= n \cdot a_1 x_1(n) + n \cdot a_2 x_2(n) \end{aligned}$$

$$a_1 y_1(n) + a_2 y_2(n) = a_1 n x_1(n) + a_2 n x_2(n)$$

The system is linear. That's why the system is an LTI.

2022 - I (c)

$$x(n) = 2^n + 6, \quad 0 \leq n \leq 6$$

$$x(0) = 1 \quad x(1) = 2 \quad x(2) = 4 \quad x(3) = 8$$

$$x(4) = 16 \quad x(5) = 32 \quad x(6) = 64$$

i)  $y(n) = x(2n)$

$$y(0) = x(2 \cdot 0) = x(0) = 1$$

$$y(1) = x(2 \cdot 1) = x(2) = 4$$

$$y(2) = x(2 \cdot 2) = x(4) = 16$$

$$y(3) = x(2 \cdot 3) = x(6) = 64$$

$$y(n) = \{1, 4, 16, 64\}$$

ii)  $y(n) = x(3-n)$

$$y(0) = x(3-0) = x(3) = 8$$

$$y(1) = x(3-1) = x(2) = 4$$

$$y(2) = x(3-2) = x(1) = 2$$

$$y(3) = x(3-3) = x(0) = 1$$

$$y(n) = \{8, 4, 2, 1\}$$

2022-2 (a)

### Statement of Sampling Theorem:

If a continuous time signal  $x(t)$  has a maximum frequency component  $f_{\max}$ , then it can be perfectly reconstructed from its samples  $x[n] = x(nT_s)$  taken at a uniform sampling rate  $f_s = \frac{1}{T_s}$ , provided that the sampling rate  $f_s$  is greater than twice the maximum frequency component of the signal.

$$f_s \geq 2f_{\max}$$

The minimum sampling rate  $2f_{\max}$  is called the Nyquist rate.

In simpler terms, when converting an analog signal into a digital signal (discrete), we must take samples at regular intervals. The Sampling theorem tells us the minimum sampling rate needed to perfectly reconstruct the original analog signal from its samples.

If the sampling is done too slowly we get a phenomenon called aliasing where different signals become indistinguishable when sampled.

Aliasing is a distortion that occurs when a continuous signal is sampled at a rate lower than the Nyquist rate. Due to under sampling, different frequency components in the original signal become

indistinguishable, resulting a mis-

representation of signal.

I have a sinusoidal signal  
 $f = 3 \text{ kHz}$  and sampling rate

$$f_s = 6 \text{ kHz}$$

$$\text{Nyquist rate} = 2 \times 3 \text{ kHz} = 6 \text{ kHz}$$

$$f_s > 2f$$

No aliasing occurs.

$$\text{Sampling rate} = 14 \text{ kHz}$$

$$f_s < 2f$$

Aliasing is occurring.

Antialiasing filter before sampling

a low pass filter that removes

frequencies higher than the

half Sampling rate. It insures

that only frequency components

that can be correctly

sampled remain after filtering

aliasing will not occur

Real world analog signals often contain frequency components higher than the Nyquist frequency. If we sample such a signal directly, these high-frequency components will cause aliasing, leading to incorrect or distorted digital representation. That's why we need anti-aliasing filters.

Difference equations provide a time-domain recursive relationship between the input and output of a LTI discrete-time system. They express the current output sample as a linear combination of past output samples and current and past input samples.

Here how they characterize the response:

### Input Output Relationship:

A difference equation directly defines how the system transforms an input sequence  $x[n]$ , given into an output sequence  $y[n]$ .

### Implementation:

Difference equations provide a direct blueprint for implementing LTI systems in software or hardware using delays, multipliers and adder.

### Analysis:

They are a starting point for analyzing system properties such as the system.

finding the system's transfer function in the z-domain by taking z transform of the difference equation.

$$(B)x + (A)y = (P)x \Rightarrow Y(z) = Pz^{-1}X(z)$$

The difference equation for a 3-point moving averaging filter is:

$$y[n] = \frac{1}{3}[x[n] + x[n-1] + x[n-2]]$$

Given input sequence

$$x[n] = [2, 3, 1, 5, 0, 4, 2]$$

$$\begin{aligned} y[0] &= \frac{1}{3}[x[0] + x[-1] + x[-2]] \\ &= \frac{1}{3}[2 + 0 + 0] = \frac{2}{3} \end{aligned}$$

$$\begin{aligned} y[1] &= \frac{1}{3}[x[1] + x[0] + x[-1]] \\ &= \frac{1}{3}[3 + 2 + 0] = \frac{5}{3} \end{aligned}$$

$$\begin{aligned} y[2] &= \frac{1}{3}[x[2] + x[1] + x[0]] \\ &= \frac{1}{3}[2 + 3 + 0] = \frac{5}{3} \end{aligned}$$

$$\begin{aligned} y[3] &= \frac{1}{3}[x[3] + x[2] + x[1]] \\ &= \frac{1}{3}[0 + 2 + 3] = \frac{5}{3} \end{aligned}$$

$$y[3] = \frac{1}{3} (x[3] + x[2] + x[1])$$

$$= \frac{1}{3} (0 + 5 + 1 + 3)$$

$$= 3$$

$$y[4] = \frac{1}{3} (x[4] + x[3] + x[2])$$

$$= \frac{1}{3} (0 + 5 + 1)$$

$$= 2$$

$$y[5] = \frac{1}{3} (x[5] + x[4] + x[3])$$

$$= \frac{1}{3} (4 + 0 + 5)$$

$$= 3$$

$$y[6] = \frac{1}{3} (x[6] + x[5] + x[4])$$

$$= \frac{1}{3} (2 + 4 + 0)$$

$$= [2 + 4 + 0] / 3 = 2$$

$$y[7] = \frac{1}{3} (x[7] + x[6] + x[5])$$

$$= [0 + 2 + 4] / 3 = 2$$

$$y[8] = \frac{1}{3} (x[8] + x[7] + x[6])$$

$$= \frac{1}{3} (0 + 2 + 4) = \frac{2}{3}$$

THE OUTPUT SIGNAL

$$y[n] = \left\{ \dots, 0, \frac{2}{3}, \frac{5}{3}, 2, 3, 2, 3, 2, 2, \dots \right\}$$

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we use the Fourier Transform  
to convert a signal from the  
time domain to the frequency  
domain; This helps us

- i) Analyze the frequency components of signals
- ii) Simplify the analysis of linear time invariant systems
- iii) Solve differential equation more easily.
- iv) Understand signal behaviour, filtering, and modulation in the frequency spectrum

## Fourier Transform $\delta(t) \cdot u(t)$

$\delta(t)$  is the Dirac delta function

$u(t)$  is the unit step function.

$\delta(t)$  is nonzero only at  $t=0$

$u(t) = 1$  for  $t > 0$  and 0 for  $t < 0$

$$\delta(t) \cdot u(t) = \delta(t) \text{ for } t=0$$

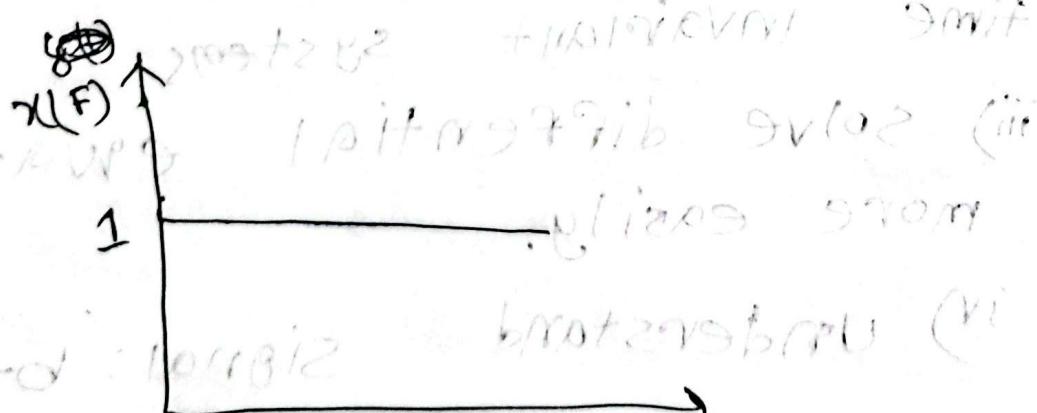
$$X(f) = \int_{-\infty}^{\infty} u(t) e^{-j2\pi ft} dt$$

$$= \int_{-\infty}^{\infty} \delta(t) e^{-j2\pi ft} dt$$

when  $t=0$  it is 1

$$F(\delta(t)) = e^{-j2\pi f \cdot 0} = e^{0} = 1$$

$$\int_{-\infty}^{\infty} \delta(t) dt = 1$$



it is 1 because it is 1

unit impulse function

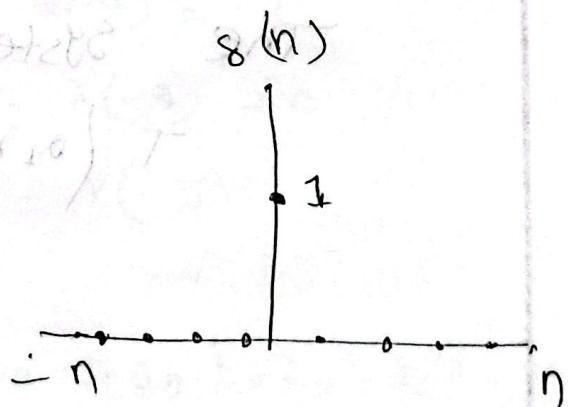
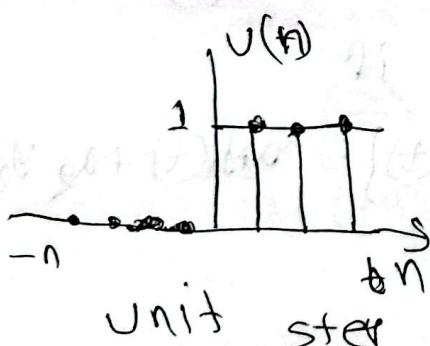
2020 - 3(a)

The unit sample sequence is a discrete time signal with a value of 1 at  $n=0$  & for all other values of  $n$ .

$$s[n] = \begin{cases} 1, & \text{if } n=0 \\ 0, & \text{if } n \neq 0 \end{cases}$$

Unit step sequence is a discrete time signal that remains constant at a value of 1 for all time samples greater than or equal to 0 and 0 for all negative time samples.

$$u(n) = \begin{cases} 1, & n \geq 0 \\ 0, & n < 0 \end{cases}$$



~~3. (b)~~ The linear property refers to the behavior of a system, not a signal itself. A system is said to be linear if it satisfies the following two properties:

### 1. Additivity:

If a system  $T$  satisfies

$$T[x_1(t)] = y_1(t) \quad T[x_2(t)] = y_2(t)$$

$$T[x_1(t) + x_2(t)] = y_1(t) + y_2(t)$$

### 2. Homogeneity (Scaling):

$$T[x(t)] = y(t)$$

$$T[a \cdot x(t)] = a \cdot y(t)$$

The system is linear if

$$T[\alpha_1 x_1(t) + \alpha_2 x_2(t)] = \alpha_1 y_1(t) + \alpha_2 y_2(t)$$

2020

$$3.c) x(n) = \begin{cases} n & -3 \leq n \leq 4 \\ 0 & \text{otherwise} \end{cases}$$

$$i) y(n) = x(n-1)$$

$$y(0) = x(0-1) = \cancel{x(-1)} \quad x(-1) = -1$$

$$y(1) = x(1-1) = x(0) = 0$$

$$y(2) = x(2-1) = x(1) = 1$$

$$y(3) = x(3-1) = x(2) = 2$$

$$y(4) = x(4-1) = x(3) = 3$$

$$y(5) = x(5-1) = x(4) = 4$$

$$y(6) = x(6-1) = x(5) = 0$$

$$y(7) = x(7-1) = x(6) = 0$$

⋮

⋮

$$y(-1) = x(-1-1) = x(-2) = -2$$

$$y(-2) = x(-2-1) = x(-3) = -3$$

$$y(-3) = x(-3-1) = x(-4) = 0$$

$$y(-4) = x(-4-1) = x(-5) = 0$$

$$y(n) = \left\{ \dots, 0, -3, -2, -1, 0, 1, 2, 3, 4, 0, \dots \right\}$$

$$ii) y(n) = 3 \cdot x(n+1)$$

$$y(-5) = 3 \cdot x(-5+1) = 3 \cdot x(-4) = 3 \cdot 0 = 0$$

$$y(-4) = 3 \cdot x(-4+1) = 3 \cdot x(-3) = 3 \cdot 0 = 0$$

$$y(-3) = 3 \cdot x(-3+1) = 3 \cdot x(-2) = 3 \cdot -2 = -6$$

$$y(-2) = 3 \cdot x(-2+1) = 3 \cdot x(-1) = 3 \cdot -1 = -3$$

$$y(-1) = 3 \cdot x(-1+1) = 3 \cdot x(0) = 3 \cdot 0 = 0$$

$$y(0) = 3 \cdot x(0+1) = 3 \cdot x(1) = 3 \cdot 1 = 3$$

$$y(1) = 3 \cdot x(1+1) = 3 \cdot x(2) = 3 \cdot 2 = 6$$

$$y(2) = 3 \cdot x(2+1) = 3 \cdot x(3) = 3 \cdot 3 = 9$$

$$y(3) = 3 \cdot x(3+1) = 3 \cdot x(4) = 3 \cdot 4 = 12$$

$$y(4) = 3 \cdot x(4+1) = 3 \cdot x(5) = 3 \cdot 0 = 0$$

$$y(n) = \{ \dots, 0, -6, -3, 0, 3, 6, 9, 12, 0, \dots \}$$

$$\underline{2020 - 9(a)}$$

An system is a system that satisfies two key properties:

Linearity:

A system is linear if it satisfies

$$T\{x_1(t) + x_2(t)\} = T\{x_1(t)\} + T\{x_2(t)\}$$

$$T\{a \cdot x(t)\} = a T\{x(t)\}$$

Together

$$T(a_1 x_1(t) + a_2 x_2(t)) = a_1 T\{x_1(t)\} + a_2 T\{x_2(t)\}$$

## 2. Time Invariance:

A system is time invariant if a time shift  $t$  in the input causes the same time shift in the output.

$$T\{x(t)\} = y(t)$$

Then any shift  $t_0$ :

$$T\{x(t-t_0)\} = y(t-t_0)$$

Q.

2020-4(b)

Let  $h[n]$  be the impulse response of a discrete time LTI system.

That is if input is  $s[n]$  then output is

$$T\{s[n]\} = h[n]$$

Any discrete time signal

$$x[n] = \sum_{k=-\infty}^{\infty} x[k] \cdot s(n-k)$$

This is called shifting property of the delta function.

Due to linearity,

$$T\{x[n]\} = T\left\{\sum_{k=-\infty}^{\infty} x[k] \cdot s(n-k)\right\}$$

$$= \sum_{k=-\infty}^{\infty} x[k] + \{s \cdot (n-k)\}$$

Due to time invariance:

$$T\{s[n-k]\} = h[n-k]$$

$$y[n] = T\{x[n]\}$$

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

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

This is the convolution sum  
in discrete time.

$$y(t) = \int_{-\infty}^{\infty} x(\tau) \cdot h(t-\tau) d\tau$$

[continuous time  
convolution sum]

2020 (9 - c)

c)  $x(n) = \{2, 5, 7, 8\}$

$h(n) = \{1, 2, 3, 4\}$

$y(n) = h(n) * x(n)$

length of  $y(n) = 4+4-1 = 7$

$$\begin{array}{r}
 2 \quad 5 \quad 7 \quad 8 \\
 \quad \quad 4 \quad 3 \quad 2 \quad 1 \\
 \hline
 32 = y(6)
 \end{array}$$

$$\begin{array}{r}
 2 \quad 5 \quad 7 \quad 8 \\
 \quad \quad 4 \quad 3 \quad 2 \quad 1 \\
 \hline
 28 + 24 = 52 = y(5)
 \end{array}$$

~~25~~

<del>y(n)</del>	2	5	7	8
1	2	5	7	8
2	4	10	14	16
3	6	15	21	24
4	8	20	28	32

$y[n] = \{2, 9, 23, 45, 57, 52, 32\}$

2022-9.(a)

Importance of ROC in Z Transform;

- The Region of Convergence determines for which values of  $Z$  the  $Z$ -transform converges.
- It is essential to identify system stability and causality.
- Two different time-domain signals can have the same  $Z$ -transform expression but different ROC, which makes the ROC critical for identifying the signal equality.

Given  $x(n) = -u(-n-1)$

The unit step function is defined

$$u(n) = 1 \quad \text{for } n \geq 0$$

$$u(n) = 0 \quad \text{for } n < 0$$

Therefore

$$u(-n-1) = 1 \quad \text{for } -n-1 \geq 0$$
$$\therefore n \leq -1$$

$$u(n-1) = 0 \quad \text{for } -n-1 \leq 0$$

~~$n < 107$~~

~~$n > 0$~~

$x(n) = -u(-n-1)$  means it is

$$x(n) = -1 \quad \text{for } n \leq -1$$

$$x(n) = 0 \quad \text{for } n > -1$$

This is a left sided sequence.

The Z-transform is defined as:

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

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

~~$\text{Let } m = -n$~~

~~$\text{when } n = -\infty \quad m = \infty$~~

$$n = -1 \quad m = 1$$

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

$$= - (z^1 + z^2 + z^3 + z^4 + \dots)$$

$$(z - 5 + 5z + 5z^2 + 5z^3 + \dots)$$

$$= \frac{z}{z-1}$$

$$= \frac{z}{z-1}$$

~~ROC: For the geometric series~~

~~to converge~~

~~we need  $|z| < 1$~~

~~ROC is  $|z| < 1 \Rightarrow (1) z$~~

~~$|z| < 1 \Rightarrow (1) z$~~

~~2022-4(b)~~

$$x(n) = \{2, 1, 0, 1\}$$

$$n(n) = \{1, 2, 3, 1\} \quad y(n) = x(n) * h(n)$$

n	2	1	0	1	
1	2	1	0	1	$= (1) x$
2	4	2	0	2	$n=1 \rightarrow 4$
3	6	3	0	3	$n=2 \rightarrow 6$
1	2	1	0	1	$n=3 \rightarrow 2$

$$y(n) = \{2, 5, 8, 6, 3, 3, 1\}$$

Multiplication of Z-domain:

$$X(z) = (2z^0 + 1z^{-1} + 0z^{-2} + 1z^{-3})$$

$$= (2 + z^{-1} + z^{-3})$$

$$H(z) = 1z^0 + 2z^{-1} + 3z^{-2} + 1z^{-3}$$

$$H(z) = 1 + 2z^{-1} + 3z^{-2} + z^{-3} - 10z^{-3}$$

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

$$= (2 + z^{-1} + z^{-3}) \cdot (1 + 2z^{-1} + 3z^{-2} + z^{-3})$$

$$= 2 + 4z^{-1} + 6z^{-2} + 2z^{-3} + z^{-1} + 2z^{-2} + 3z^{-3} + z^{-4} + 3z^{-5} + z^{-6}$$

$$Y(z) = 2 + 5z^{-1} + 8z^{-2} + 6z^{-3} + 3z^{-4} + 3z^{-5} + z^{-6}$$

The coefficients of  $Y(z)$  are  $y(n)$ .

$\therefore$  The convolution in the time domain

is equal to multiplication in  $z$  domain.

$$(n)y[n]h[n] = (n)x[n]$$

$$\therefore y[n]h[n] \sum_{n=0}^{\infty} = (n)x[n]$$

$$\therefore h[-n]x[n] \sum_{n=0}^{\infty} =$$

2022- A(c)

### Causal signal:

A discrete time signal  $x(n) = 0$  for  $n < 0$ .  
This means the signal only exists  
for non-negative time indices.

$$x(n) = (0.5)^n u(n)$$

$u(n)$  makes sure that  $x(n)$  is zero  
for  $n < 0$ .

### ROC for causal signal:

For a causal signal, the Z-transform  $X(z)$  will have an ROC that is the exterior of a circle. That is the ROC will be of the form  $|z| > R$  where  $R$  is the magnitude of the outermost pole.

$$x(n) = (0.5)^n u(n)$$

$$X(z) = \sum_{n=0}^{\infty} (0.5)^n z^{-n}$$

$$= \sum_{n=0}^{\infty} (0.5 z^{-1})^n$$

$$\text{where } = 1 + 0.5z^{-1} + (0.5z^{-1})^2 + \dots$$

So H(z) is stable if  $|H(z)| < 1$

$$R.C: \frac{1}{|1 + 0.5z^{-1}|} < 1 \Rightarrow |1 + 0.5z^{-1}| > 1$$

$$\Rightarrow |0.5z^{-1}| < 1 \Rightarrow |z| < 2$$

$$\therefore |z| > 0.5$$

$$(a) \circ (b) \circ (c) = (d) \circ$$

Anti causal signal:  $\sum_{n=-\infty}^{\infty} x(n) = (S)x$

A discrete time signal  $x(n)$  is anti-causal if  $x(n) = 0$  for  $n > 0$ . This means the signal only exists for non-positive time indices.

$$x(n) = (0.5)^n u(-n)$$

$u(n)$  makes sure that  $x(n)$  is zero

for  $n > 0$

$$u(-n) = 1 \text{ for } n \leq 0$$

$$= 0 \text{ for } n > 0$$

$$L(S)$$

$$S \circ (S)$$

$$S \circ S \circ L(S)$$

ROC for anti causal signals

For an anti causal signal, the  $\mathcal{Z}$  transform  $X(z)$  will have an ROC that is the interior of a circle. That is, the ROC will be of the form  $|z| < R$ .

$$x(n) = (0.5)^n u(-n)$$

$$X(z) = \sum_{n=-\infty}^0 (0.5)^n z^{-n}$$

Let  $m = -n$  (using shift A)

when  $n = -\infty \Rightarrow m = \infty$ :  $(0.5)^m \rightarrow 0$  as  $m \rightarrow \infty$

$$\begin{aligned} X(z) &= \sum_{m=0}^{\infty} (0.5)^m z^m \\ &= (0.5) \sum_{m=0}^{\infty} (0.5 z)^m = (0.5) \frac{1}{1 - 0.5z} \end{aligned}$$

$$\therefore (0.5)^m \text{ for } m \geq 0 \Rightarrow \sum_{m=0}^{\infty} (0.5 z)^m$$

$$= \frac{1 + 0.5z + (0.5z)^2 + \dots}{1 - 0.5z}$$

$$0.5z \Rightarrow z = (0.5)^{-1}$$

$$0.5z = \frac{1}{1 - 2z}$$

$$|2z| < 1$$

$$|2z| < \frac{1}{2}$$

$$\therefore |z| < 0.5$$