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DEPARTMENT OF COMPUTER ENGINEERING
UNIVERSITY OF BENIN, BENIN CITY
2nd SEMESTER EXAMINATIONS 2018/2019 SESSION

CPE 512: DIGITAL SIGNAL PROCESSING — **28/11/2019**, **Time: 3hours**

INSTRUCTION: ATTEMPT ANY FIVE QUESTIONS ONLY WITH AT MOST TWO QUESTIONS FROM EACH SECTION

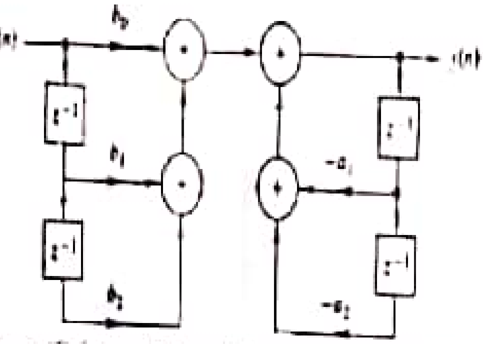
SECTION A

QUESTION ONE (1)

a) Mathematically define the z–transform and explain how it can be used to analyse signals.

b) State the properties of z-transform

c) What is the system function of the system represented by the block diagram below?



$X(n) = \sum_{n=0}^{N-1} x(n) e^{-jwn}$

$w = \frac{2\pi k}{N}$

d) Sketch the Region of Convergence (ROC) of the following types of signals:
i. Causal finite-duration ii. Two-sided finite-duration iii. Anti-causal infinite-duration

QUESTION TWO (2)

a) i. Determine the Fourier transform of $u(n)x(n-1)$

ii) What is the relationship between z-transform and Fourier transform?

b) Mathematically represent the inverse z-transform of a discreet-time signal and state the methods that can be used for its evaluation.

c) Given a system described by $y(n) = \frac{1}{2}y(n-1) + 2x(n)$

i) Determine the system function

ii) Determine the unit sample response

$\frac{1}{2\pi j}\oint \frac{1}{2\pi j}\oint x(z) z^{n-1}$

$\frac{1}{1-az^{-1}}$

$2 \int(z) - \frac{1}{2}\frac{z^{-n}}{2}$

SECTION B

QUESTION THREE (3)

a. Consider a DSP system as shown in fig.1.0. suppose the audio signal from the mic., bandlimited to 10khz is corrupted by high frequency noise and the spectrum of the noise is from 20khz to 30khz. The noisy analog signal is sampled at 50khz. A smoothing algorithm is to be developed for the DSP system so as to remove the noise from the signal.

i. Write a DSP codes for the algorithm and explain the concept of the algorithm been used

ii. Supposed the audio signal was sampled one sample per time determine the time interval between samples and also.

iii. Determine the time interval between frames if the audio signal was sampled frame per time

iv. Suppose the algorithm developed in (a)i. require 25 processing operations to be performed between samples. Calculate the minimum require DSP processor speed

b. Explain the function of the CODEC in the figure

$\frac{2}{z^{-n}}$

$\frac{1}{2\pi j}\int \frac{X(z)}{z} \cdot u(n)x(n)$ $a u(n) z^{-n}$

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$y(n)$
$Y(z) = \frac{1}{2}Y(z)z^{-1} + 2x(z)$ $\frac{n}{m} - k$
$Y(z)(1 - \frac{1}{2}z^{-1}) = x(z)$ $n = mk$
$X(\frac{n}{m}) z^{-n}$ $z^{m} + z$ z^{-mk} $\frac{a}{z^{-n}}$

QUESTION FOUR (4)

a. Explain the functional units and the following components of a DSP processor architecture

i. Cross-path

ii. PLL

iii. JTAG

b. Explain how DSP can be used to implement a speaker recognition system or any scheme in image processing

c. Explain the term 'linear and circular' addressing mode as it relates to digital signal processing

d. Write an ASM function program for the DSP function $y(n) = \sum_{n}^{k} ak.x(n-K)$

e. Add comments to the program below

```
LDH   *+A4[0], A5
LDH   *+A4[1], A2
LDH   *+A4[2], A3
NOP3
MPY B4, A2, A8
NOP1
SHR A8, 14, A8
SUB A8, *+A4[2]
STH A2, *+A4[0]
SUB A2, *+A4[1]
B B3
NOP5
end
```

$\frac{1}{2\pi\beta} \int \frac{1}{\;}$

QUESTION FIVE (5)

a. With the aid of block diagram, explain how analog signals and digital signals may be processed by digital signal processing technique implemented on a DSP chip. Briefly describe the function of each block in your diagram and indicate how its specification is affected by the bandwidth of the analog and digital input and output signals and the choice of sampling rate. (10.5 marks)

b. Outline the steps you would take to achieve the following in CCS (or visualDSP++) environment

i. Create a FIR project

ii. Load and Run FIR program

iii. Plot and View FIR data

c. Write a DSP program in 'C 'that calls an ASM function to generate a 32- bit noise sequence

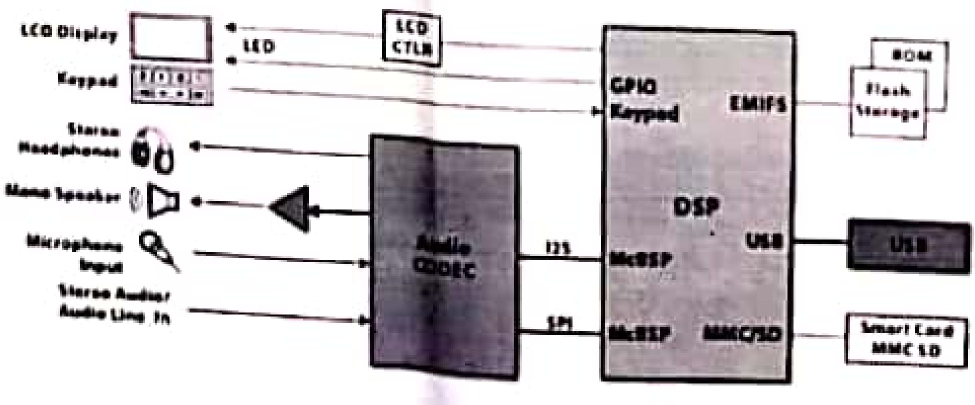

Fig. 1.0: DSP in a MP3 player/recorder system

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