

Electrical Engineering Department

Digital Communication Systems (802421) - G2

Dr. Mouaaz Nahas Term 2 (1434-1435) First Exam, Thursday 03/061435 H

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 Q1. Choose the correct answer:
 (19 Marks, 1 Each)

 In digital communication system, the filter placed
 a) Antialiasing filter.

1.	before the sampler is called:	b) Aliasing filter. c) Reconstruction filter. d) Low-pass filter.		
	If the sinusoidal signal $m(t)$ , band-limited to B Hz is sampled at P Hz, the reconstructed signal at the	a) Completely different AC signal		
2.	is sampled at <i>B</i> Hz, the reconstructed signal at the receiver will be:	b) Same as $m(t)$ waveform but with different amplitude		
2.		c) Same as $m(t)$ waveform but with different frequencies		
		a) DC signal	_	
	The bandwidth of the signal	a) 50 Hz		
3.	$m(t) = \operatorname{sinc} (50\pi t) + \operatorname{sinc}^2 (60\pi t) \operatorname{sinc} (60\pi t)$ is:	b) 90 Hz		Commented [W1]:
5.		c) 60 Hz		Bandwith of sinc $(50\pi t)$ is 25 Hz
		d) 120 Hz		Bandwith of sinc <sup>2</sup> ( $60\pi t$ ) is 60 Hz
				Bandwith of sinc $(60\pi t)$ is 30 Hz
	If the number of quantization levels used in an A/D	a) 20		Bandwith of sinc <sup>2</sup> ( $60\pi t$ ) sinc ( $60\pi t$ ) is $60+30=90$ Hz
4.	conversion is 262146, the minimum number of	b) 19		Bandwith of $m(t)$ is the higher between 25 and 90, it is 90 Hz.
4.	binary bits per sample is:	c) 18		
		d) None of the above		<b>Commented [m2]:</b> log <sub>2</sub> [ceiling(262146)] = 19
	Undersampling is the process of sampling a signal	a) Good because it helps to avoid aliasing	-	
	with a sampling frequency lower than Nyquist rate.	b) Good because it saves bandwidth.		
5.	Undersampling is therefore:	c) Bad because it reduces SNR and bandwidth.		
		d) Bad because it produces low quality or distorted signal at the receiver output.		
	An audio signal used for <i>intelligibility</i> application	a) $B = 3.5 \text{ kHz}, L = 256$	-	
	has the bandwidth <i>B</i> . If this signal is sampled,	b) $B = 3.5$ MHz, $L = 65,536$	-	
6.	digitized then binary-coded using <i>n</i> bits per	, , ,	_	
	sample. Practical values for <i>B</i> and <i>L</i> can be:	c) $B = 15$ kHz, $L = 65,536$		
		d) $B = 20$ kHz, $L = 256$	_	
	Given the signal $m(t) = \sin(60\pi t) + \sin(100\pi t) +$	a) 150		<b>Commented [m3]:</b> recall that the sinusoidal format is $\sin (2\pi f t)$ .
7.	sin (150 $\pi$ t). Nyquist sampling rate for this signal (in Hz) is:	b) 75		Here, we have three frequencies: $f_1 = 30$ Hz, $f_1 = 50$ Hz, $f_1 = 75$ Hz
/.		c) 300		$J_1 = 30$ Hz, $J_1 = 50$ Hz, $J_1 = 75$ Hz Nyquist rate equals twice the highest frequency, that is 75 x 2 = 150
		d) 310	_	Hz.
	Given the two signals $m_1(t)$ and $m_2(t)$ with	a) 300		
8.	bandwidths 100 Hz and 40 Hz, respectively. The	b) 380		<b>Commented [W4]:</b> 3x100 + 2x40 = 380
о.	bandwidth of $m_1^3(t)$ . $m_2^2(t)$ is:	c) 140		
		d) 340		

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	For the two signals given in Q.9, The bandwidth of	a) 280		
9.	$m_1(t) * m_2(t)$ is:	b) 140		
<i>y</i> .		c) 100	 	
		d) 40	 <b>Commented [W5]:</b> $m_1(t) * m_2(t) \leftrightarrow M_1(f) M_2(f)$	
	The signal shown in the figure is sampled at Nyquist rate, the bandwidth <i>B</i> of the signal (in Hz) is: $\int_{a}^{a} \int_{a}^{b} \int$	a) 1	Bandwidth is the lower (smaller).	
10.		b) 2		
		<mark>c)</mark> [3]	 <b>Commented [m6]:</b> In each second, we have 6 samples. So Nyquist rate is 6 samples/sec (or 6 Hz). Therefore, <i>B</i> is half this value which is 3 Hz.	
		d) 6	Note that B here is the highest sinusoidal frequency.	
	Consider the two unmodulated and modulated signals shown in the figure. The modulated signal is:	a) Flat-top PAM signal		
11.		b) Natural PAM signal		
		c) Natural PDM signal		
		d) Natural PPM signal		
	Which one of the following pulse-modulation	a) PWM		
12.	techniques is the most power-efficient?	b) PAM		
12.		c) PPM		
		d) PCM	 <b>Commented [m7]:</b> PPM is also power efficient as the modulated parameter is the pulse position while amplitude and width are fixed.	
	The block diagram below shows the building			
	blocks of the complete PCM system. The blocks A,	a) LPF, Quantizer, Sampler and Bit-Encoder	 However, with PCM, we can use a rectangular pulse for the bit 1 (as with PPM) and no pulse for the bit 0, allowing more power to be	
13.	B, C and D (in order) are:	b) LPF, Sampler, Quantizer and Bit-Encoder	saved.	
15.		c) Sampler, LPF, Quantizer and Bit-Encoder		
		d) Sampler, LPF, Bit-Encoder and Quantizer		
	In uniform quantization, SQNR increases as:	a) Quantization levels decrease		
		b) Signal peak amplitude increases		
14.		c) Signal average power increases		
		d) Quantizer limits expand		
		u) Quantizer minis expand		
	The sinusoidal signal $m(t)$ with frequency 50 Hz is to be sampled, the following condition should be	a) $f_s > 100 \text{ Hz}$	<b>Commented [W8]:</b> Since there is an impulse (high power) at <i>f</i> =	
		b) $f_s = 100 \text{ Hz}$	 50 Hz.	
15.	satisfied for proper reconstruction:	c) $f_s \ge 100 \text{ Hz}$	 $B = 50$ Hz, so $f_s$ should be larger than (but not equal to) $2B = 100$ Hz.	
		d) None of the above		
	A signal $m(t)$ with maximum amplitude 2.5 mV is	a) 64		
16.	sampled, quantized and digitally transmitted. If each quantization interval is 0.039 mV, the number	b) 128	 <b>Commented [W9]:</b> $L = 2m_p / \Delta v = 5m / 0.039m = 128.21 \approx 128$	
	of quantization levels used is:	c) 256		
	or quantization to tolo used 15.	d) None of the above		

	In the following figure, <u>Aliasing</u> is appointed by the letter:	a) A			
	A C C	b) B			
17.		c) C			
		d) D			
	Aliasing is described mathematically as:	a) Lost tail of $G(f)$ beyond $ f  < f_s/2$ .			
18.		b) Lost tail of $G(f)$ beyond $ f  > f_s/2$ .			
10.		c) Tail of $G(f)$ beyond $ f  < f_s/2$ inverted back on $G(f)$ .			
		d) Tail of $G(f)$ beyond $ f  > f_s/2$ inverted back on $G(f)$ .			
19.	The solution to aliasing is:	a) Passing the signal $G(f)$ into a low pass filter with cutoff frequency $2f_s$ Hz			
		<ul> <li>b) Passing the signal G(f) into a low pass filter with cutoff frequency fs Hz</li> </ul>			
		c) Passing the signal <i>G</i> ( <i>f</i> ) into a low pass filter with cutoff frequency <i>f</i> <sub>s</sub> /2 Hz			
		d) There is no practical solution			

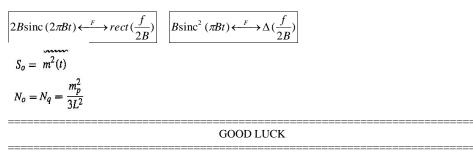
Q2. Solve this question on the back side of this page.

(3 Marks)

A television signal (video and audio) has a bandwidth of 4.5 MHz. This signal is sampled, quantized, and binary coded to obtain a PCM signal.

- (a) Determine the sampling rate if the signal is to be sampled at a rate 20% above the Nyquist rate.
- (b) If the samples are quantized into 1024 levels, determine the number of binary pulses required to encode each sample.
- (c) Determine the binary pulse rate (bits per second) of the binary-coded signal, and the minimum bandwidth required to transmit this signal.

**Useful relations:** 



## Q2. Solution:

RN=2B= 9MHZ (0) RA) = 1.2 X9 M = 10.8 MHZ. Actual 20% Sampling rate L = 1024 (6)  $n = \log 1024 = 10 \text{ bits sample.}$ (C) Binomy Pulse rate = nRA = 10×10,8M = 108 M bps. Min. bandwidth = nRA = 54 MHZ.