BITS Pilani, Dubai Campus Dubai International Academic City, Dubai, UAE

B.E. Third Year EEE, II Semester, 2012-13 COMPREHENSIVE EXAMINATION

Course No. / Course Title: EEE C383 / COMMUNICATION SYSTEMS

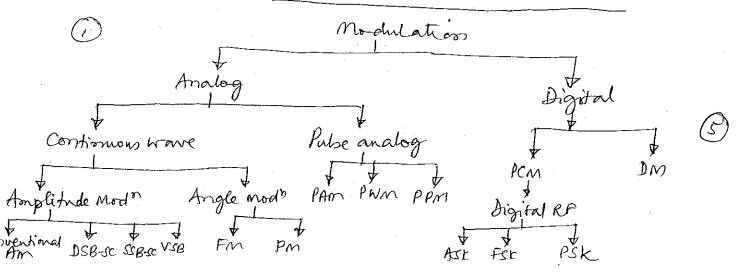
Duration: 2 Hours Max. Marks: 60 Weightage: 30%

Note:- Answer all questions

- 1. Draw a tree diagram indicating the names of all the modulation techniques, both analog and digital, that you have studied in this course. Ensure proper classification. (5)
- 2. Assuming a quarter-wave antenna, what would be the required antenna height for effective radio communication at 30 MHz? (3)
- 3. Define the term *multiplexing*. Name and distinguish between the various types of multiplexing that you have come across. (3)
- 4. You have come across the Hilbert transform during your study of SSB-SC modulation. Obtain the Hilbert transform of the unit impulse function $\delta(t)$. (6)
- 5. Consider a channel whose output-input relationship is given by $v_o(t) = a_1 v_i(t) + a_2 v_i^2(t)$. Show that if an AM signal passes through this channel, it would suffer distortion at the output. On the other hand, show that an FM signal goes through without distortion. Support your answer with appropriate mathematical analyses. (10)
- 6. Assuming an AWGN channel, sketch the power spectral density of noise at the output of (a) an FM receiver, and (b) a PM receiver. Label the axes correctly. (6)
- 7. What is aperture effect in pulse sampling? How can it be minimized? (5)
- 8. Name the errors that can be encountered in Delta Modulation. If a sine wave of amplitude A and frequency f_m is passed through a Delta Modulator whose step size is Δ , sketch the approximated signal for the condition $\Delta > 2A$. (6)
- 9. Define *entropy* of a discrete memory less source. What is the entropy of the English alphabet assuming that all the letters occur with the same probability? (6)
- 10. With the help of block diagrams and mathematical analysis, show how an FSK signal can be generated and demodulated. (10)

*** Paper ends ***

SOLUTIONS/MARKING SCHEME



$$f = 30 \text{ MHz}$$

$$\lambda = \frac{c}{f} = \frac{3 \times 10^8}{30 \times 10^6} = 10 \text{ m}$$

$$\frac{\lambda}{4} = 2.5 \text{ m}$$

For Am, vi(t) = to[I+kam(t)] (D) wet

for FM, vi(t) = Ac (Os [wet + kg Smtt) dt]

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Substituting for vo(t), it can be shown that

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for the Am case we have distortion components

in the neceiver input signal.

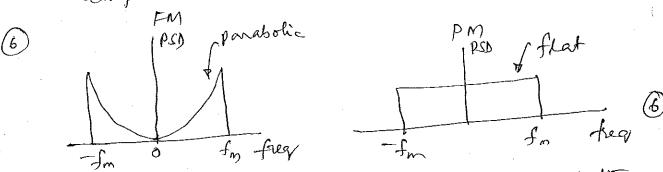
in the neceiver input signal.

for FM, it is seen that vi(t) component is

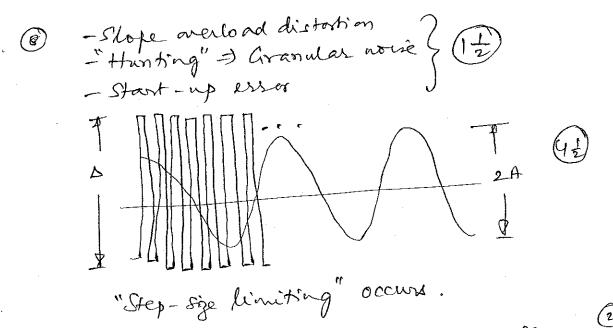
present at neceiver input without any distortion

present at neceiver input without any distortion

components.



Finite pulse with results in higher flees of the recovered message signal suffering greater attenuation than lower frees due to suix effect.



- Average info/message of the source. (2) $H(s) = -\log(\frac{1}{26}) \text{ bit/message} \sim 4.7 \text{ bits/onessage}$
- (10) As discussed in class.

 Should include both coherent detection (5)

 and non-coherent detection (6)

 with mathematical analysis.

BITS Pilani, Dubai Campus B.E. (Hons.) EEE Third Year, Second Semester, 2012-13 Test 1 (Closed Book)

EEE C383 Communication Systems

Duration: 50 min

Max. Marks: 30

Weightage: 15%

Note: Answer all questions. All questions carry equal marks.

- 1. Obtain and sketch the Fourier transform of a periodic train of unit impulses. Assume that the period is T_0 .
- 2. An unmodulated carrier wave has a normalized power P_C . When the carrier is amplitude modulated to a depth of μ by a sinusoid, the power increases to P_T . Obtain the relationship between μ , P_C , and P_T .
- 3. Obtain the Hilbert transform of the function $\cos \omega_0 t \cos \omega_1 t$, $\omega_1 > \omega_0$.
- 4. Recall the square-law modulator for AM that was discussed in the class. If the input is now a conventional AM signal and the bandpass filter is replaced by a lowpass filter, we obtain a square-law demodulator. Show, with appropriate mathematical analysis, that this demodulator allows distortionless recovery of the baseband signal only if the AM wave has a low percentage modulation.
- 5. Obtain the expression for modulation index of a single-tone phase-modulated signal.

SOLUTIONS / ANSWERING SCHEME

1. Using Fourier series approach it can be shown that

$$\sum_{k=-\infty}^{\infty} \delta(t-kT_0) = \omega_0 \sum_{n=-\infty}^{\infty} \delta(w-n\omega_0) \text{ where } \omega_0 = \frac{2\pi}{T_0} G$$

2. S(t) = Ac[1+ Mas wort] asupt

$$P_{c} = \frac{Ac^{2}}{2}; \quad S_{sB} = \frac{A_{c} \mu^{2} + A_{c} \mu^{2}}{2} = \frac{Ac^{2} \mu^{2} \cdot \frac{1}{2} \cdot \frac{1}{2}}{4} = \frac{Ac^{2} \mu^{2}}{4} + \frac{Ac^{2} \mu^{2}}{2} = \frac{Ac^{2} \mu^{2} \cdot \frac{1}{2}}{2} = \frac{Ac^{2} \mu^{2} \cdot \frac{1}{2}}{4} + \frac{Ac^{2} \mu^{2}}{2} = \frac{Ac^{2} \mu^{2} \cdot \frac{1}{2}}{2} = \frac{Ac^{2} \mu^{2$$

(3) Using the definition of Hilbert transform, it can be shown that Coswot Coswot H.T. > Coswot Sino, t for WIND. (6)

(a) $v_i(t) = A_c [1 + k_a m(t)] coswet$ $v_o(t) = a_i v_i(t) + a_2 v_i^2(t)$ Output of LPF can be shown to be $y(t) = \frac{a_2 A_c^2}{2} + a_2 A_c^2 k_a m(t) + \frac{a_2 A_c^2 k_a^2}{2} . LPF[m^2(t)]$ Desired component: $a_2 A_c^2 k_a m(t)$ Distartion component: $a_2 A_c^2 k_a^2 LPF[m^2(t)]$

Ratio of desired to distortion component:

Ram(t)

Ideally, this should be infinite.

Possible if |kam(t) | (4) => & mall percentage modulation.

(3) Single tone PM:

$$O(t) = w_c t + k_p on(t)$$

$$= w_c t + k_p Am Gnwmt$$

$$w_i(t) = \frac{dO(t)}{dt} = w_c + k_p Am w_m(-Sui) w_mt)$$

$$|\Delta w| = |c_p| A_{on} \omega_m$$

$$|c_p| = \frac{\Delta w}{w_{on}} = |c_p| A_{on}$$

*** Paper ends ***

Probability of selecting one word from 10,000 word is 10,000 :. Info/word = - log (10000) bis No. of words used to describe pricture is look. :. An. of bits/picture = -1000 x log (10000) = 13.29 × 10³ = 1.33 × 10 7 bits Lato/pixel = - log_2(to) = 4 bits No. g pixels/pichue = 0.1 × 106 Info forchure = 0.4×106 = 4×105 bits Raho = $\frac{4 \times 10^{5}}{1.33 \times 10^{4}} = \frac{40}{1.33} = 30:1$ Minimum sampling rate is 2 fm samples/pec Assume or bits/pample =) 2rfm bik Here n = 8 2x8fm < 48×106 .. for < 48 × 106 => frax = 3 mbs Francisco January -> Encoder > ... R -.. -> D/A In = 3 kg fs = 20 Kly Am & A. Fs Am = IV $\Delta > \frac{A_{m} \cdot 21 \text{ fm}}{f_{S}} > \frac{1 \times 21 \times 3 \times 10^{3}}{20 \times 10^{3}}$ → 12,3TV.

BITS PILANI, DUBAI CAMPUS Dubai International Academic City, Dubai, UAE

BE (Hons.) EEE Third Year, II Semester, 2012-2013 Course No. / Course Title: EEE C383 COMMUNICATION SYSTEMS

Quiz 2

Duration: 20 min	Max. Marks: 10	Weightage: 5%		
Name:				
ID No	o.:			
Answer all questions in the blanks provided o	against each question. Numerica	l answers should be supported with		
appropriate rough work in the space provide				
1. White noise is applied as the input to	a filter having the transfer fo	unction $H(\omega)$. The power spectral		
density (PSD) of the output is				
2. For a channel to be distortionless, its		eover the entire		
frequency range and the phase response				
3. AWGN stands for		(0.5)		
1. A 10 kHz bandwidth signal is transmi	tted using DSB-SC modulation	n. Noise with two-sided PSD of 10 ⁻⁹ W/Hz		
encountered in the channel. For an output SNR of 40 dB, the required signal power at the input of the				
demodulator is (2)				
6. Coherent detection is preferred over	envelope detection for conve	entional AM under low input SNR condit		
for the following reason:	•			
		out <i>SNR</i> increases by a factor of (0.		
. The PSD of noise at the output of an				
. In order to reduce the effect of noise	in the higher frequency region	on of the message signal, FM systems		
employ a fil	ter at the transmitter and a $_$	filter at the receiver		
. DPSK stands for		At the output of a DPSK receiver		
errors tend to occur in	(1)			
O. In an FSK receiver using coherent det	ection, the filter is a	filter. (1)		

(Space for rough work. Use other side too, if necessary.)

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Quiz 2

	Duration: 20 min	Max. Marks: 10	Weightage: 5%
	So L	NTIONS	
	· ·		
	swer all questions in the blanks provided aga propriate rough work in the space provided; o		
1.	White noise is applied as the input to a density (PSD) of the output is $\frac{\mathcal{M}}{2}$		unction $H(\omega)$. The power spectral
2.	For a channel to be distortionless, its m frequency range and the phase respons	agnitude response must b	e <u>Coustout</u> over the entire with frequency. (1)
3.	AWGN stands for Additive w	hite Caussian a	√rî se . (0.5)
4.		d using DSB-SC modulatio	n. Noise with two-sided PSD of 10^{-9} W/Hz is
5.	Coherent detection is preferred over en	velope detection for conv	entional AM under low input SNR condition
	for the following reason: if does not exhibit	- threshold ef	fect ' out SNR increases by a factor of <u>4</u> . (0.5)
6.	In FM, for every doubling of the transm	ission bandwidth, the outr	out SNR increases by a factor of <u>4</u> . (0.5)
7.	The PSD of noise at the output of an FM	l receiver has a <u>身のしぬ</u>	spectrum. (1)
8.	In order to reduce the effect of noise in employ a <u>pre-lumphanis</u> filter	at the transmitter and a	on of the message signal, FM systems de_lh_phasso filter at the receiver. (1)
9.	errors tend to occur in bauxa. In an FSK receiver using coherent detect	(1)	on of the message signal, Fivi systems de-lhaphase filter at the receiver. (1) 1 mg. At the output of a DPSK receiver
10.	In an FSK receiver using coherent detect	tion, the filter is a $\frac{1}{2}$	∼ filter. (1)

(Space for rough work. Use other side too, if necessary.)

BITS Pilani, Dubai Campus B.E. (Hons.) EEE Third Year, Second Semester, 2012.2013 Quiz 1

EEE C383 Communication Systems Max. Marks: 10

Weightage: 5%

Duration: 20 min

Nan	ne: ID No.:
Note	:- For numerical answers, indicate the calculations in the space for rough work in order to get full credit.
1.	Consider a source X that produces five symbols with probabilities $\frac{1}{2}$, $\frac{1}{4}$, $\frac{1}{4}$, and $\frac{1}{16}$. The source entropy $H(X)$ is bits/symbol. (1.5)
2.	An analog signal having 4 kHz bandwidth is sampled at 1.25 times the Nyquist rate. The sampling rate is samples/sec. Each sample is quantized into one of 256 equally likely levels. Assume that the successive samples are statistically independent. The entropy of the source is bits/sample and the information rate of this source is kilobits/sec. (0.5+0.5+1)
3.	Sampling theorem states that: (1.5+0.5)
	Undersampling a signal would result in
4.	In Delta Modulation, the consequences of choosing a small step size are to experience
5.	PCM stands for and it involves the
	operations of, and (0.5x4)
6.	Name the three analog pulse modulation schemes:
	, (0.5x3)

(Space for rough work. Show all numerical calculations for full credit. Use other side too, if necessary)

BITS Pilani, Dubai Campus B.E. (Hons.) EEE Third Year, Second Semester, 2012.2013 Quiz 1

EEE C383 Communication Systems

	Duration: 20 min Max. Marks: 10 Weightage: 5%
Nan	ne:ID No.:
Note	:- For numerical answers, indicate the calculations in the space for rough work in order to get full credit.
1.	Consider a source X that produces five symbols with probabilities $\frac{1}{2}$, $\frac{1}{2}$, and $\frac{1}{16}$. The source entropy $H(X)$ is bits/symbol. (1.5)
2.	An analog signal having 4 kHz bandwidth is sampled at 1.25 times the Nyquist rate. The sampling rate is 10 k samples/sec. Each sample is quantized into one of 256 equally likely levels. Assume that the
	successive samples are statistically independent. The entropy of the source is 80° bits/sample and the information rate of this source is 80° kilobits/sec. (0.5+0.5+1)
3.	Sampling theorem states that: (1.5+0.5) Given a signal boundlimited to fully, it can be seconstructed from its samples provided that the sampling rate
	from its samples provided that the sampling rate
	Undersampling a signal would result in Alasing.
4.	In Delta Modulation, the consequences of choosing a small step size are to experience Sloke overload and that of large step size is "Hun tog" (0.5x2)
5.	PCM stands for Pulse-Code modulation and it involves the
6.	operations of <u>Sampling</u> <u>Quantization</u> and <u>Suarding</u> . (0.5x4) Name the three analog pulse modulation schemes: <u>PAM</u>
	PWM . (0.5x3)

(Space for rough work. Show all numerical calculations for full credit. Use other side too, if necessary)