Rony Parvej's

Lecture- 1 & 2: Communication

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শুরুর কথাঃ

আসসালামু আলাইকুম। আলহামদুলিল্লাহ। শেষ পর্যন্ত লেকচার নাম্বার ১ এবং ২ এর কাজ শেষ করতে পেরেছি। ৩০০ কপি প্রিন্ট হচ্ছে প্রথম দফায়। প্রথমবার বলে খুব ভয়ে ছিলাম। মাত্র ১ সপ্তাহে সব প্রশ্ন গুছিয়ে লেকচার শিট কমপ্লিট করা আসলেই বেশ কঠিন। আমি আমার চেষ্টার প্রায় সর্ব্বোচ্চটুকু দিয়ে লেকচার শিটটা নির্ভুল এবং পূর্ণাঙ্গ করার চেষ্টা করেছি। গতকাল রাতে মাত্র ১ ঘন্টা ঘুমিয়েছি, আজ অফিস ছুটি নিয়ে নিয়েছি। এরপরও বারবার মনে হচ্ছে "আরেকটু ভাল করলে ভাল হতো" , "আহা! এটা তো বাদ পড়ে গেল! ", "এই নিয়মটা কি আদৌ ঠিক আছে?", "এটা কি আরেকটু সহজ করে উপস্থাপন করা যেত না?", "আরো কয়েকটা বইয়ে একবার চোখ বুলিয়ে নিলে ভাল হতো" । এরকম হাজারটা প্রশ্ন এই লেকচার শিটের প্রিন্টিংকে বারবার পিছিয়ে দিয়েছে। আজকে আবার হয়েছে কি শোনেন। সকালবেলা যখনই কম্পিউটার অন করতে যাব তখনই গেল ইলেকট্রিসিটি। পরে শুনলাম আজকে নাকি সন্ধ্যা ৭ টার আগে ইলেকট্রিসিটিই আসবেনা। আমার সব ফাইল ডেস্কটপে। ড্রপবক্সে আপডেটেড ফাইল আপলোড করা হয়নি। সারাটা দিন মাটি হবে ভেবে মন অনেক খারাপ হয়ে গেল। অবশ্য দুপুরেই ইলেকট্রিসিটি চলে আসল এরপর। শেষ পর্যন্ত কোন কিছু চিন্তা না করেই প্রিন্টিং এ নিয়ে যাচ্ছি এখন। সময় স্বল্পতার জন্য প্রুফ রিডিং বা রিভিশানও দিইনি। অন্তত একটা পান্ডুলিপি তো দাঁড়িয়ে যাক! এই লেকচার শিট কমপ্লিট করব বলে গত দু দিন ফেসবুকে বা ফোনে সময় দিতে পারিনি। অদেখা কয়েকশ ম্যাসেজ জমে গেছে ইনবক্সে। আমি একা হাতে আর কতদিক সামলাব? তার উপর প্রথমবার। সে হিসেবে একটু ক্ষমাসুন্দর দৃষ্টি আশা করছি আপনাদের কাছ থেকে।

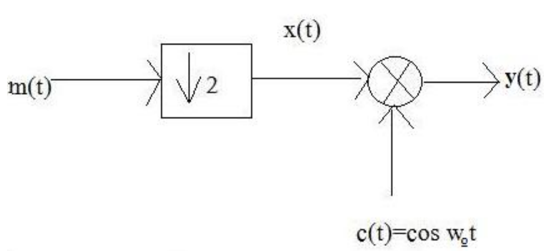
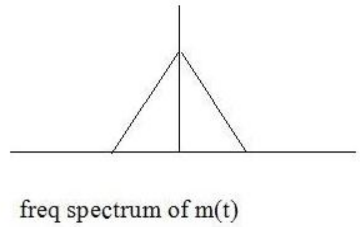
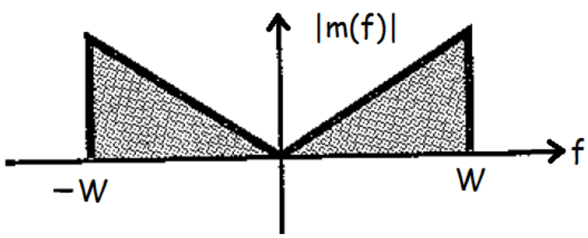
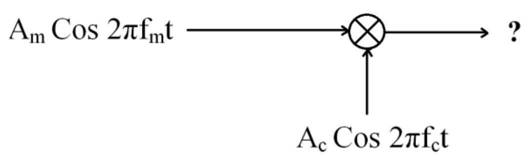
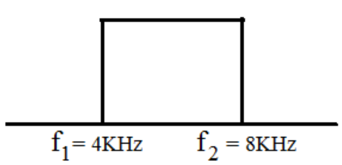
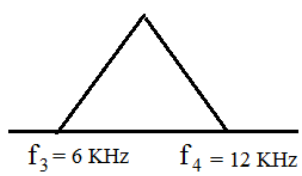
এই লেকচার শিটের কোন প্রশ্নের উত্তরই ১০০% সঠিক নয়। বেশ কিছু বই এবং ডকুমেন্টস পড়ার পর আমার কাছে যেগুলো মোটামুটি সঠিক মনে হয়েছে সেগুলোর সমষ্টি মাত্র। কাজেই এতে ভুল খুঁজে পাওয়া খুবই স্বাভাবিক। আমিও চাই আপনারা আমাকে আমার ভুলগুলো ধরিয়ে দিন কিংবা কোন সহজ পদ্ধতি থাকলে প্লিজ জানান। এই লেকচার শিটের ভুলগুলো পরবর্তী লেকচার শীটে অথবা "ভুল সংশোধন" নামের আরেকটা লেকচার শিটে সংশোধন করে দেওয়া হবে ইনশাল্লাহ। লেকচার শিট অথবা কোচিং এর আপডেট জানার জন্য নিচের ফেসবুক আইডিটাকে ফলো করার অনুরোধ করছি। এতে করে আমার আপডেট জানানোর কাজটা সহজ হয়ে যাবে। ভাল থাকবেন। দোয়া করবেন আমার জন্য।

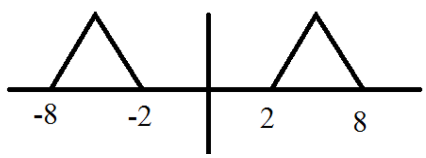
রনি পারভেজ

- মোবাইলঃ ০১৭১৮-৯৩৬৪১৫
- ফেসবুকঃ https://www.facebook.com/profile.php?id=100009221389358

Sl.	Question	Exam
	Question Review	
	COMMUNICATION	
Sl.	Question	Exam
Type: 1 – Bit Rate Calculation		
1	Write down the steps involved in PCM for baseband signal.	BUET M.Sc. 12
2	State Nyquist theorem. What is the condition to recover message signal from sampled signal.	EGCB-12, BUET M.Sc. 13
3	A signal $x(t)=5 \cos (1000 \pi t)$ is sampled at nyquist sampling rate and quantized using 8 bit PCM system. Determine the bit rate of the digital signal.	NWPGCL-14
4	A PCM system multiplexes 10 band limited voice channel (300-3400 Hz) and uses a 256 level quantizer, considering the standard sampling rate for telephone system the bandwidth of binary encoded signal is- (a) 640 Kb/s (b) 80 Kb/s (c) 248 Kb/s (d) 496 Kb/s	MCQ PGCB-14
5	A PCM system multiplexes 20 band limited voice channel (300-3400 Hz) and uses a 256 level quantizer, considering the standard sampling rate for telephone system the bandwidth of binary encoded signal is (a) 1280 Kb/s (b) 1088 Kb/s (c) 496 Kb/s (d) 992 Kb/s	EGCB-12
6	A PCM system multiplexes 20 band-limited voice channels (300-3400 Hz) and uses a 256-level quantizer. Considering the standard sampling rate for telephone system, the overall data rate of the binary encoded signal can be calculated as (a) 64 kbps (b) 1.28 Mpbs (c) 1.088 Mbps (d) 5.12 Mbps	MCQ BPDB – 14 (FF)
7	A PCM system multiplexes 20 band limited voice channel (300-3400 Hz). 15 of them are multiplexed and uses a 256 level quantizer, considering the standard sampling rate for telephone system what will be the bandwidth of binary encoded signal ?	PGCL-11
8	In PCM, the number of quantization level is increased from 4 to 64, then the bandwidth requirement will approximately be increased (a) 8 times (b)16 times (c) 3 times (d) 32 times	MCQ BPDB-15
9	A PCM-TDM system multiplexes 10 band limited voice channel (300-3400 KHz) and uses a 256 level quantizer. If the signal is sampled at a rate $17\frac{11}{17}$ % higher than Nyquist rate, then what will be the maximum energy bandwidth of the transmission channel?	DPDC-14
10	The signal $x(t) = 2 \sin (500 \pi t) + 3 \sin (1400 \pi t) + 2 \sin (3400 \pi t) + 2 \sin (6900 \pi t)$ has been band limited within (300-3400 Hz). If this signal is sampled at Nyquist rate, what will be output data rate if this signal is encoded with a 512 level uniform quantization. (a) 72 kb/s (b) 61.2kb/s (c) 55.8 kb/s (d) 68.1 kb/s	MCQ DPDC-14
11	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 minimum bandwidth required to transmit the signal.	PGCB-11
12	It is desired to set up a central station for simultaneous monitoring of the electrocardiograms (ECGs) of 10 hospital patients. The data from the rooms of the 10 patients are brought to a processing center over wires and are sampled, quantized,	BUET MSC-14

	binary coded, and time-division multiplexed. The multiplexed data are now transmitted to the monitoring station. The ECG signal bandwidth is 100 Hz. The maximum acceptable error in sample amplitudes is 0.25% of the peak signal amplitude. The sampling rate must be at least twice the Nyquist rate. Determine the minimum cable bandwidth needed to transmit these data.	
13	What is the resolution of a 8-bit ADC operating at 10 V range (a) 39.06 mV (b) 2.44 mV (c) 0.625 V (d) None of the above	MCQ BPDB-13
Type-2: SNR Related		
14	In which of the following noise level is reduced? (4 values of SNR were given)	MCQ BPDB -14 (FF)
15	Bandwidth = … KHz, SNR= … dB, what is bit rate?	MCQ BPDB -14 (FF)
16	For a voice channel, if the signal power level is -3dbm (-5 MW) and noise level is -20 dbm (0.01 MW), then SNR = (a) 17 dbm (b) 16.9 dbm (c) 50 dbm (d) None of these	MCQ EGCB-12
17	x (t) = 1.5 cos (800πt) is to be PCM with minimum SQNR of 25dB. How many bytes are required for encoding each having uniform quantization.	BUET MSC-14
18	Signal power = 2 MW, Noise power = 1.95 MW. Find maximum data rate and Shanon's capacity. [Find maximum delta modulation এবং shanon's theorem (300-3300) Hz এই টাইপ কি নাকি উল্লেখ ছিল। তথ্যদাতা পুরোপুরি মনে রাখতে পারেননি।]	BUET MSC-14
19	What is the SNR (in dB) of a voice channel if the signal power level is 0.52 mW and noise level is 0.01 mW? (a) 52 dB (b) 34.32 dB (c) 17.16 dB (d)	MCQ DPDC -14
20	The Bandwidth of a signal is 10 KHz and SNR is 12 dB. Find the bit rate (According to Jahid Sumon, Maximum bit rate) of the binary PCM .	BPDB-14 (FF)
21	If the signal at beginning of a cable with −0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km?	PGCB-11
22	A transmitter is transmitting data at a rate of 65 Kbps. At the receiver, the error detector detects 32 errors in the received bits in 15 seconds of the data transmission. Calculate Bit error rate (BER) of the communication system.	PGCB-11
23	Describe pre-emphasis and de-emphasis.	DWASA-11
Type-3: Spectrum Drawing		
24	For maximum frequency B draw the frequency spectrum of DSB-SC and SSB.	PGCB-14
25	A signal $A_m \sin f_m \pi t$ and carrier is $A_c \sin(2\pi f_c t + \partial)$. Find the DSB, Amplitude modulated signal and draw the upper and lower sideband frequency spectrum.	BPDB-11
26	x(t) = 2 sin (400 πt) + 4 cos (600 πt). Sampling of 2400 Hz. (a) Find the equation of x(n). (b) Find the period of x(n). (c) Draw the spectrum of x(n).	BUET MSC-14
27	If m(t)= 2 sin 2000πt + 4 sin 4000πt, then (i) Find minimum sampling frequency required to avoid aliasing (ii) If sampling frequency is 10KHz, draw the spectrum of the sampled signal.	DWASA-2014
28	A signal x(t)=2 sin(400πt)+ 6 sin(640πt) is ideally sampled at 500 Hz and then fed to an ideal lowpass filler with a cut-off frequency of 400Hz. Determine the frequencies that will be available at the output.	BPDB-13

29	Draw the frequency spectrum of x(t) and y(t):  	EGCB-14
30	The spectrum of a modulating signal is shown in the figure. Draw the spectrum of DSB-SC, SSB+C, and VSB modulated signals for this modulating signal assuming a carrier signal of C (t) = A_C Cos 2π f_C' t 	DPDC-14
31		BUET M.Sc. -14
32	If m(t) = B Sinc (2πBt) , B=1000 & $\acute{\omega}_c$ =10000π, then Draw the Spectrum of DSB-SC and LSB signal.	BUET M.Sc. - 12
33	Find a signal g(t) that is band-limited to B Hz and whose samples are $g(0) = 1$ and $8(\pm T_S) = g(\pm 2T_S) = 8(\pm 3\ T_S) = \ldots = 0$ where the sampling interval T_S is the Nyquist interval for g(t), that is, $T_S = 1/2B$.	DWASA-11
34	If m(t) = B cos ω_m t and index μ=1, then find Φ_{AM} (t) and sketch it.	DWASA-11
35	একটা square এবং একটা triangular wave দেওয়া ছিল। এর ফ্রিকুয়েন্সি মডুলেটেড সিগন্যালের কি যেন (স্পেকট্রাম?) আঁকতে দিয়েছিল।  	DWASA-14
Type-4: Modulation & Power of Modulated wave		
36	A 1KW Carrier is amplitude modulated to a depth of 60%. Calculate total power and Sideband Power of the modulated wave.	BUET M.Sc. 12
Type-5: Others		
37	Demonstrate OOK, FSK, PSK signal assuming a bit sequence 01001101.	EGCB-12
38	Write down the advantages and limitations of digital communication.	PGCL-11
39	The main reason for the superiority of digital communication over analog communication is (a) The use of simple electronic circuitry. (b) The use of amplifiers periodically (c) The use of regenerative repeaters (d) The use of A/D and D/A converters	MCQ DPDC -14

40	The main advantage of a digital communication system over that of an analog one is (a) reduced complexity of the receiver (b) robustness to noise (c) use of regenerative repeaters (d) all of the above	MCQ BPDB-13
41	Explain the slop overload effect of delta modulation.	BPDB-11
42	What is power line communication? Give some example.	BPDB-12
43	Write a few applications of Power Line Carrier Communication (PLCC).	BPDB-13
44	Abbreviate: VSAT, WiMAX, WLAN, ADSL, SONET, OFDMA	DWASA-14
45	What is meant by: OFDM, GMSK, WiMAX, DWDM, PSTN, BISDT.	BUET M.Sc. Unknown
46	What is erlang of telephone traffic? Related Math.	BUET M.Sc. Unknown
47	What are the Common Multiple Access Technologies? Differentiate between Multiplexing & Multiple Access Technologies.	BUET M.Sc. 12
48	why is parallel transmission more useful than serial transmission? (a) For long distance data transmission (b) For short distance data transmission (c) For synchronous transmission (d) For Asynchronous transmission	MCQ DPDC-14
49	Envelop detector is helpful for which of the following modulation? (এই টাইপ কিছু একটা ছিল) (a) ASK (b) ASK and FSK (c) FSK (d) PSK	MCQ DPDC-14
50	Find the probable bandwidth of the following signal 	MCQ DWASA-14
51	Inter-symbol interference occurs when (a) channel bandwidth (BW) is close to the signal BW (b) signal BW is much larger than channel BW (c) channel BW is much larger than signal BW (d) channel BW is large as signal BW	MCQ BPDB-13
52	Which one of the following is a valid uplink frequency band used in a GSM system (a) 1930-1990 MHz (b) 890-915 MHz (c) 440-460 MHz (d) 935-960 MHz	MCQ BPDB-13
53	For modulation, a GSM system generally employs (a) GMSK (b) 8-PSK (c) QPSK (d) both (a) and (b)	MCQ BPDB-13, MCQ BUET M.Sc.-13
54	Which statement is TRUE regarding analog modulation techniques? (a) FM signal offers better receptive quality compared with AM because it has narrower bandwidth than that of AM (b) FM signal is more noise resistant than PM signal (c) Synchronous detection can be used for AM, and PM signals (d) None of the above	MCQ BPDB-13
55	A discrete time signal is given by $x(n) = \text{Cos}[(n\pi)/9]$. The signal is (a) periodic with period N=9 samples. (b) periodic with period N=18 samples. (c) periodic with period N=32 samples. (d) aperiodic	MCQ BPDB-15
56	What is the carrier in Submarine Cable?	MCQ BUET M.Sc.-13

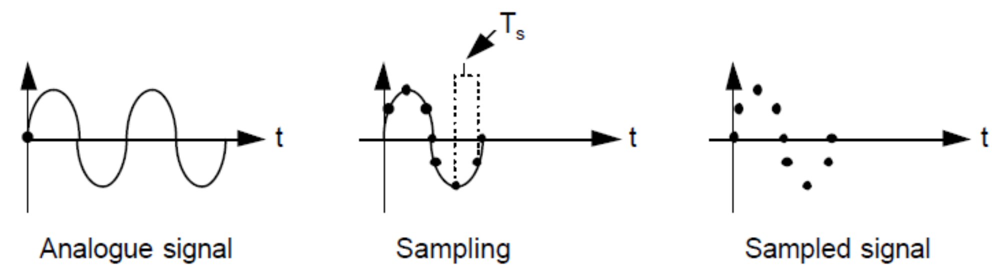
1	Write down the steps involved in PCM for baseband signal.	BUET M.Sc. 12

PCM involves three main steps:

- Sampling
- Quantization
- Coding

Sampling

Sampling involves measuring the analog signal at specific time intervals.



The accuracy of describing the analog signal in digital terms depends on how often the analog signal is sampled. This is expressed as the sampling frequency. The sampling theory states that:

To reproduce an analog signal without distortion, the signal must be sampled with at least twice the frequency of the highest frequency component in the analog signal.

Quantization

Quantization is to give each sample a value.

Coding

Coding involves converting the quantized values into binary.

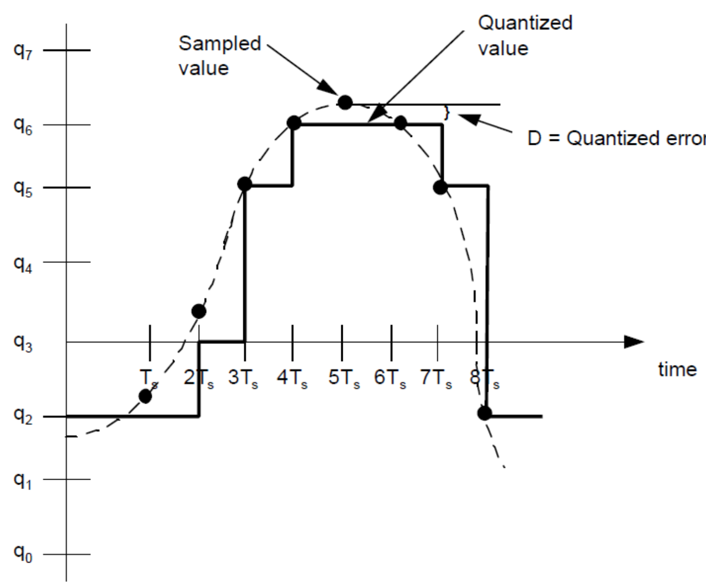


Figure: Quantization

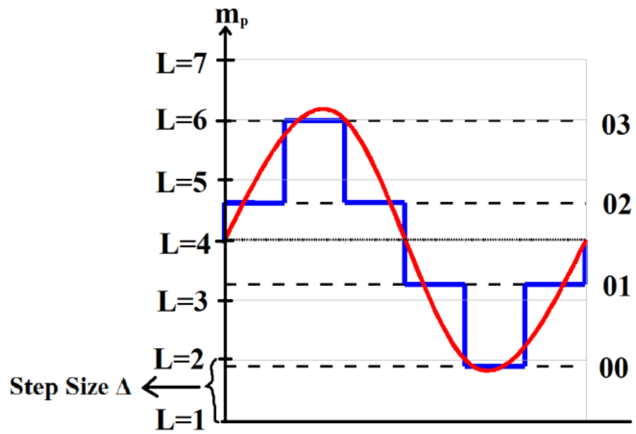
2	State Nyquist theorem. What is the condition to recover message signal from sampled signal.	EGCB-12, BUET M.Sc. 13

Nyquist theorem is a theorem that is followed in the digitization of analog signals. It is states that-

"A signal must be sampled at least twice as fast as the bandwidth of the signal to accurately reconstruct the waveform; otherwise, the high-frequency content will alias at a frequency inside the spectrum of interest (passband)."

To reproduce an analog signal without distortion, the signal must be sampled with at least twice the frequency of the highest frequency component in the analog signal.

Basics of Type: 1



সূত্রগুলো মনে রাখিঃ

* $\Delta = 2m_p / L$
* $P_M = m_p{}^2/2$
* $P_Q = \Delta^2/12 = m_p{}^2/ 3L^2$

$P_M / P_Q = 3L^2/2$

* SNR (in dB) $= 10 \log (P_M / P_Q)$
$= 10 \log (3L^2/2)$ *

$$\boxed{\text{SNR (in dB)} = 1.76 + 20 \log L}$$

* $L = 2^n$

$$\boxed{\text{SNR (in dB)} = 1.76 + 6.02\, n}$$

এই সূত্রকে **6dB Rule of SNR** বলা হয় ।

R_b = bit rate = No. of bit used to encode a sample * sampling Rate * $R_b = nf_S$ * $L = 2^n$

* $f_{NQ} = 2 f_m$
* $f_S \geq f_{NQ}$
* **Standard sampling rate for telephone system = 8KHz**
* **We can transmit up to 2 bit/s with 1Hz Bandwidth.** (Minimum Bandwidth Recquired চাইলে এইটা ব্যবহৃত হবে)
* **Maximum acceptable quantization error = Δ /2**

কোনটা যেন কি?

m_p = Signal peak value

$2m_p$ = Operating Range

Δ = step size = resulation

L = No. of Quantization level

n = No. of bits in the sample's code

P_M= Signal power (average/or R.M.S.)

P_Q= Quantization Noise power (average)

f_S = Sampling frequency

f_m = message signal's frequency

f_{NQ} = Nyquist frequency

3	A signal $x(t)=5\cos(1000\pi t)$ is sampled at nyquist sampling rate and quantized using 8 bit PCM system. Determine the bit rate of the digital signal.	NWPGCL-14

Frequency of the message signal, $f_m = 1000\pi / 2\pi = 500$ Hz

So, Sampling frequency, f_S = Nyquist frequency, $\mathbf{f_{NQ} = 2f_m}$ = 2*500 Hz = 1000 Hz

No. of bit, n =8.

So, Bit rate of the digital signal, $\mathbf{R_b = nf_S}$ = 8*1000 Hz = 8 KHz **Ans.**

4	A PCM system multiplexes 10 band limited voice channel (300-3400 Hz) and uses a 256 level quantizer, considering the **standard sampling rate for telephone system** the bandwidth of binary encoded signal is- (a) 640 Kb/s (b) 80 Kb/s (c) 248 Kb/s (d) 496 Kb/s	MCQ PGCB-14

$L = 256 = 2^8 = 2^n$.

So, No. of bits, n =8.

Standard sampling frequency, f_s = 8KHz *(মুখস্ত রাখতে পারেন)*

So, Bandwidth for every channel, $R_b = nf_s$ = 8*8 =64 KHz

So, Bandwidth for 10 channels = 64*10 = 640 KHz

5	A PCM system multiplexes 20 band limited voice channel (300-3400 Hz) and uses a 256 level quantizer, considering the standard sampling rate for telephone system the bandwidth of binary encoded signal is (a) 1280 Kb/s (b) 1088 Kb/s (c) 496 Kb/s (d) 992 Kb/s	EGCB-12

৪ নং এর অনুরূপ। এখানে শুধু চ্যানেলের সংখ্যা আগের (১০টার) তুলনায় দ্বিগুন (২০টা)। তাই উত্তরও আগেরটার দ্বিগুন হবে। অর্থাৎ,1280 Kb/s

6	A PCM system multiplexes 20 band-limited voice channels (300-3400 Hz) and uses a 256-level quantizer. Considering the standard sampling rate for telephone system, the overall data rate of the binary encoded signal can be calculated as (a) 64 kbps (b) 1.28 Mpbs (c) 1.088 Mbps (d) 5.12 Mbps	MCQ BPDB – 14 (FF)

৫ নং এর অনুরূপ। উত্তরঃ 1280 Kb/s = 1.28 Mpbs

7	A PCM system multiplexes 20 band limited voice channel (300-3400 Hz). **15 of them** are multiplexed and uses a 256 level quantizer, considering the standard sampling rate for telephone system what will be the bandwidth of binary encoded signal ?	PGCL-11

$L = 256 = 2^8 = 2^n$.

So, No. of bits, n =8.

Standard sampling frequency, f_s = 8KHz

So, Bandwidth for every channel, $R_b = nf_s$ = 8*8 =64 KHz

So, Bandwidth for 10 channels = 64*15 = 960 KHz

8	In PCM, the number of quantization level is increased from 4 to 64, then the bandwidth requirement will approximately be increased (a) 8 times (b)16 times (c) 3 times (d) 32 times	MCQ BPDB-15

$L_1 = 4 = 2^2 = 2^{n1}$. So, $n_1 = 2$
$L_2 = 64 = 2^6 = 2^{n2}$. So, $n_2 = 6$
$R_{b1} = n_1 * f_s$
$R_{b2} = n_2 * f_s$
So, $R_{b2} / R_{b1} = n_2*f_s / n_1*f_s = n_2/ n_1 = 6/2 = 3$ times. Ans.

9	A PCM-TDM system multiplexes 10 band limited voice channel (300-3400 KHz) and uses a 256 level quantizer. If the signal is sampled at a rate $17\frac{11}{17}$ % higher than Nyquist rate, then what will be the maximum energy bandwidth of the transmission channel?	DPDC-14

Here,

Maximum frequency of the message signal, $f_m = 3400$ Hz.

∴ Nyquist frequency of the signal, $f_{NQ} = 2 f_m = 2 * 3400$ Hz =6.8 KHz

∴ Sampling frequency of the signal, $f_S = 17\frac{11}{17}$ % (=17.65%) higher than Nyquist rate
= 1.1765*6.8 KHz = 8KHz

Quantization Level, $L = 256 = 2^8 = 2^n$
∴ No. of bits in the code, n=8

∴ Maximum energy bandwidth of one channel = $n f_S$ = 8*8 KHz = 64 KHz.

∴ Maximum energy bandwidth of 10 channel = 10* 64 KHz =640 KHz **Ans**.

10	The signal $x(t) = 2 \sin(500 \pi t) + 3 \sin(1400 \pi t) + 2 \sin(3400 \pi t) + 2 \sin(6900 \pi t)$ has been **band limited within (300-3400 Hz).** If this signal is sampled at Nyquist rate, what will be output data rate if this signal is encoded with a 512 level uniform quantization. (a) 72 kb/s (b) 61.2kb/s (c) 55.8 kb/s (d) 68.1 kb/s	MCQ DPDC-14

অনেকগুলা সাইন কিংবা কস ওয়েভ একসাথে যোগ আকারে থাকলে যে অংশের ফ্রিকুয়েন্সি সবচেয়ে বেশি সেটাই সম্মিলিত সিগন্যালের ফ্রিকুয়েন্সি। অর্থাৎ এক্ষেত্রে ফ্রিকুয়েন্সি হবার কথা ছিল $6900\pi/2\pi = 3450$ KHz. কিন্তু সিগন্যালটা (300-3400 Hz) এ ব্যান্ড লিমিটেড।অর্থাৎ সিগন্যালের ফ্রিকুয়েন্সি 3400 Hz এর বেশি হতে পারবে না! অর্থাৎ,

Maximum frequency of the message signal, $\mathbf{f_m = 3400}$ **Hz.**
So, Sampling frequency, f_S = Nyquist frequency, $f_{NQ} = 2f_m = 2*3400$ Hz = 6800 Hz = 6.8 KHz.
Quantization Level, $L = 512 = 2^9 = 2^n$
∴ No. of bits in the code, n=9
∴ Output Data rate, $R_b = n f_S = 9*6.8$ kb/s = 61.2 kb/s **Ans**.

11	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 **minimum bandwidth required** to transmit the signal.	PGCB-11

Frequency of the message signal, $f_m = 4.5$ MHz

Nyquist frequency, $f_{NQ} = 2f_m = 2* 4.5$ MHz = 9 MHz

(20% above মানে 1.2 গুন)

(a) So, Sampling frequency, $f_S = 1.2*9$ MHz = 10.8 MHz **Ans.**

(b) $L=1024=2^{10}=2^n$.

n = 10

$R_b = n\, f_S = 10*10.8 = 108$ bit/s.

But We can transmit up to 2 bit/s with 1Hz Bandwidth. (Reference Example 6.2, B.P. Lathi)

So, **Minimum bandwidth required** to transmit the signal = 108/2 = 54Hz. **Ans.**

12	It is desired to set up a central station for simultaneous monitoring of the electrocardiograms (ECGs) of 10 hospital patients. The data from the rooms of the **10 patients** are brought to a processing center over wires and are sampled, quantized, binary coded, and time-division multiplexed. The multiplexed data are now transmitted to the monitoring station. The ECG **signal bandwidth is 100 Hz**. The maximum acceptable error in sample amplitudes is **0.25% of the peak signal amplitude.** The sampling rate must be at least **twice the Nyquist** rate. Determine the minimum cable bandwidth needed to transmit these data.	BUET MSC-14

Signal bandwidth, $f_m = 100$ Hz.

Nyquist rate of each signal, $f_{NQ} = 2f_m = 2*100$ Hz=200Hz

So, Sampling frequencyof each signal, $f_S = 2* f_{NQ} = 2*200$ Hz=400Hz

Given, maximum acceptable error, $\Delta /2 \leq 0.0025\, m_p$

$$\text{or, } (2m_p / L)/2 \leq 0.0025\, m_p$$

$$\text{or, } 1/L \leq 0.0025$$

$$\text{or, } L \geq 400$$

But L should be a power of 2. So, $L=512=2^9=2^n$

(কারণ $2^8 = 256$ যা 400 এর চেয়ে ছোট। তাই 400 এর চেয়ে বড় 2- এর পরবর্তী পাওয়ার 29 = 512 ধরা হয়েছে। তা না হলে no. of bits, n এর মানে দশমিক আসত। কিন্তু বিট সংখ্যা ভগ্নাংশ বা দশমিক হতে পারেনা, পূর্ণ সংখ্যা হতে হয়।)

So, No. of bit required, n = 9.

So, minimum bit rate required for one patient, $R_b = n\, f_S = 9*400$ bit/s = 3600 bit/s

We can transmit up to 2 bit/s with 1Hz Bandwidth.

So, minimum cable bandwidth needed to transmit these data for each patient= 3600/2 = 1800 Hz.

So, minimum cable bandwidth needed to transmit these data for 10 patients= 1800*10 Hz = 18000Hz

= 18KHz **Ans.**

13	What is the resolution of a 8-bit ADC operating at 10 V range (a) 39.06 mV (b) 2.44 mV (c) 0.625 V (d) None of the above	MCQ BPDB-13

Here, Operating Range, $\mathbf{2m_p = 10\ V}$

No. of bit, n = 8

So, Quantization Level, L= $2^n = 2^8 = 256$

So, resulation, $\Delta = 2m_p / L = 10/256 = 0.0390625$ V =39.0625 mV **Ans.**

Basics of Type: 2

- ❖ $$\mathrm{SNR} = \frac{P_{\text{signal}}}{P_{\text{noise}}} = \left(\frac{A_{\text{signal}}}{A_{\text{noise}}}\right)^2 = \frac{\sigma^2_{\text{signal}}}{\sigma^2_{\text{noise}}}$$ where P is average power & σ is variance
- ❖ $$\mathrm{SNR_{dB}} = 10\log_{10}\left(\frac{P_{\text{signal}}}{P_{\text{noise}}}\right) = P_{\text{signal,dB}} - P_{\text{noise,dB}},$$
- ❖ $$\mathrm{SNR_{dBm}} = 10\log_{10}\left(\frac{P_{\text{signal}}\text{ in W}}{P_{\text{noise}}\text{ in mW}}\right) = P_{\text{signal,dBm}} - P_{\text{noise,dBm}}$$
- ❖ $$\mathrm{SNR_{dB}} = 10\log_{10}\left[\left(\frac{A_{\text{signal}}}{A_{\text{noise}}}\right)^2\right] = 20\log_{10}\left(\frac{A_{\text{signal}}}{A_{\text{noise}}}\right).$$
- ❖ $$\mathrm{SNR_{dB}} \approx 6.02 \cdot n + 1.761$$ একে 6dB rule of SNR ও বলা হয়। n=bit rate
- ❖ $$C = B\log_2\left(1 + \frac{S}{N}\right)$$ একে Shanon's Capacity Theorem ও বলা হয়।

C = Channnel capacity in bit/s

S=Signal power

N= Noise power.

B = frequency/Bandwidth of the message signal

এই সূত্রে S/N বা SNR এর মান ওয়াটে বসাতে হবে । সেজন্য S/N বা SNR এর মান ডেসিবেলে দেওয়া থাকলে তাকে ওয়াটে কনভার্ট করে নিতে হবে।

- ❖ $\log_b(x) = c$ হলে, $b^c = x$

অর্থাৎ, b এর পাওয়ার যত হলে তার মান x হবে সেটাই হচ্ছে $\log_b(x)$ এর মান।

14	In which of the following noise level is reduced? (4 values of SNR were given)	MCQ BPDB-14 (FF)

$$\mathrm{SNR} = \frac{P_{\text{signal}}}{P_{\text{noise}}}$$

কাজেই যার SNR বেশি তার নয়েজ পাওয়ার বা নয়েজ লেভেলে কম। কাজেই SNR এর মান যথাক্রমে ২, ৩, ৪ ও ৫ ডেসিবেল হলে এক্ষেত্রে ৫ ডেসিবেল সিগন্যালেরই নয়েজ লেভেল সবচেয়ে কম।

15	Bandwidth = … KHz, SNR= … dB, what is bit rate?	MCQ BPDB -14 (FF)

এখানে, ধরি, Bandwidth = 2.4 KHz= 2400Hz এবং S/N = 20 dB =10^(20/ 10) =100 W

$$C = B\log_2\left(1+\frac{S}{N}\right)$$

= 2400 $\log_2$ 101 = 2400*6.658211=15,965 bt/s

16	For a voice channel, if the signal power level is -3dbm (-5 MW) and noise level is -20 dbm (0.01 MW), then SNR = (a) 17 dbm (b) 16.9 dbm (c) 50 dbm (d) None of these	MCQ EGCB-12

$$\mathrm{SNR_{dBm}} = P_{\mathrm{signal,dBm}} - P_{\mathrm{noise,dBm}}$$

= -3 – (-20) dBm = 17dBm **Ans.**

17	x (t) = 1.5 cos (800πt) is to be PCM with minimum SQNR of 25dB. How many bytes are required for encoding each having uniform quantization.	BUET MSC-14

Frequency of the signal, B = 800π / 2π = 400 Hz.

SQNR = 25 dB = $10^{2.5}$ W = 316.23 W

(শর্টকাটে ওয়াটে রূপান্তরের জন্য ডেসিবেলকে ১০ দিয়ে ভাগ করে ১০ এর পাওয়ার হিসেবে বসালেই হয়।)

C= B $\log_2$ (1+SQNR) = 400*(1+316.23) bit/s = 126892 bit/s.

So, bytes required per second = 126892/8 = 15861.5 byte **Ans.**

(1 byte=8bit)

18	Signal power = 2 MW, Noise power = 1.95 MW. Find maximum data rate and Shanon's capacity. [Find maximum delta modulation এবং shanon's theorem (300-3300) Hz এই টাইপ কি নাকি উল্লেখ ছিল। তথ্যদাতা পুরোপুরি মনে রাখতে পারেননি।]	BUET MSC-14

Let, Maximum frequency of the signal, B = 3300 Hz.

$$C = B\log_2\left(1+\frac{S}{N}\right)$$

= 3300 $\log_2$ (1+1.02564)

= 3300 $\log_2$ (2.02564)

= 3300 * 1.018378

= 3360.6474 bit/s **Ans.**

19	What is the SNR (in dB) of a voice channel if the signal power level is 0.52 mW and noise level is 0.01 mW? (a) 52 dB (b) 34.32 dB (c) 17.16 dB (d)	MCQ DPDC -14

$$\mathrm{SNR_{dB}} = 10\log_{10}\left(\frac{P_{\mathrm{signal}}}{P_{\mathrm{noise}}}\right)$$

= 10 $\log_{10}$ (.52/.01) = 10*1.716 =17.16 dB

20	The Bandwidth of a signal is 10 KHz and SNR is 12 dB. Find the bit rate (According to Jahid Sumon, Maximum bit rate) of the binary PCM .	BPDB-14 (FF)

B=10 KHz

SNR = 12 dB = $10^{1.2}$ W =15.8489 W

C = B $\log_2$ (1+SNR) = 10* $\log_2$ (1+15.8489) =10* 4.07458 =40.7458 Kbit/s **Ans.**

21	If the signal at beginning of a cable with −0.3 dB/km has a power of 2 mW, what is the power of the signal at 5 km?	PGCB-11

The loss in the cable in decibels is 5 × (−0.3) = −1.5 dB.

We can calculate the power as

$$\text{dB} = 10 \log_{10} \frac{P_2}{P_1} = -1.5$$

$$\frac{P_2}{P_1} = 10^{-0.15} = 0.71$$

$$P_2 = 0.71 P_1 = 0.7 \times 2 = 1.4 \text{ mW}$$ **Ans**.

[Reference: Example 3.30, Data Communications and Networking By Behrouz A. Forouzan, Sophia Chung Fegan]

22	A transmitter is transmitting data at a rate of 65 Kbps. At the receiver, the error detector detects 32 errors in the received bits in 15 seconds of the data transmission. Calculate Bit error rate (BER) of the communication system.	PGCB-11

Data transmitted in 15 seconds = 65*15 = 975 bits

Errors in 15 seconds = 32 bits

So, bit error rate (BER) = (32 / 975)*100% =3.28% **Ans.**

23	Describe pre-emphasis and de-emphasis.	DWASA-11

Emphasis is the intentional alteration of the amplitude-vs.-frequency characteristics of the signal to reduce adverse effects of noise in a communication system.

In processing electronic audio signals, pre-emphasis refers to a system process designed to **increase** (within a frequency band) **the magnitude of some** (usually higher) **frequencies** with respect to the magnitude of other (usually lower) frequencies .

A system process designed to decrease, (within a band of frequencies), the magnitude of some (usually higher) frequencies with respect to the magnitude of other (usually lower) frequencies is called De-emphasis.

It improves the overall signal-to-noise ratio by minimizing the adverse effects of such phenomena as attenuation distortion or saturation of recording media in subsequent parts of the system.

Type-2: Spectrum Drawing

Fourier Transform of sine & Cosine:

$$\Im\{\cos(2\pi At)\} = \frac{1}{2}[\delta(f-A)+\delta(f+A)]$$

$$\Im\{\sin(2\pi At)\} = \frac{1}{2i}[\delta(f-A)-\delta(f+A)]$$

Important Trigonometric Formulas:

2 sinA cosB = sin(A+B) + sin(A-B)

2 cosA cosB = cos(A+B) + cos(A-B)

2 cosA sinB = sin(A + B) - sin(A-B)

2 sinA sinB = cos (A-B) - cos(A+B)

Sinc Function:

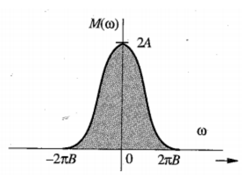
The cardinal sine function or sinc function, denoted by sinc(x).
In mathematics, the historical **unnormalized sinc function** is defined by $\operatorname{sinc}(x) = \frac{\sin(x)}{x}$.

In digital signal processing and information theory, the normalized sinc function is commonly defined by

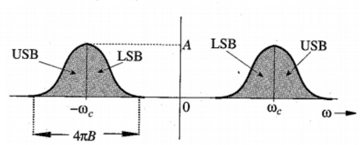
$$\boxed{\operatorname{sinc}(x) = \frac{\sin(\pi x)}{\pi x}\ .}$$

In either case, the value at x = 0 is defined to be the limiting value sinc (0) = 1.

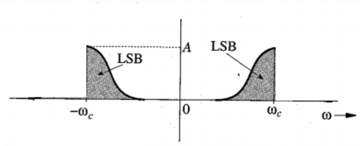
24	For maximum frequency B draw the frequency spectrum of DSB-SC and SSB.	PGCB-14



message signal of bandwidth B Hz



DSB-SC



SSB (LSB)

Reference: Figure 4.1, B.P. Lathi

25	*A signal $A_m Sin\, f_m \pi\, t$ and carrier is $A_c sin(2\pi f_c t + \partial)$. Find the DSB, Amplitude modulated signal and draw the upper and lower sideband frequency spectrum.*	BPDB-11

(এই প্রশ্নের উত্তরটা সঠিক হয়েছে কিনা তা নিয়ে আমার সন্দেহ আছে। যাহোক, তারপরও নাই মামার চেয়ে কানা মামা ভাল তাই দিলাম।)

Message signal = $A_m \text{ Sin } f_m \pi\, t$

Carrier Signal = $A_c \sin(2\pi f_c t + \partial)$.

DSB, Amplitude modulated signal = $[A_m \text{ Sin } f_m \pi\, t] * [A_c \text{ Sin}(2\pi f_c t + \partial)]$

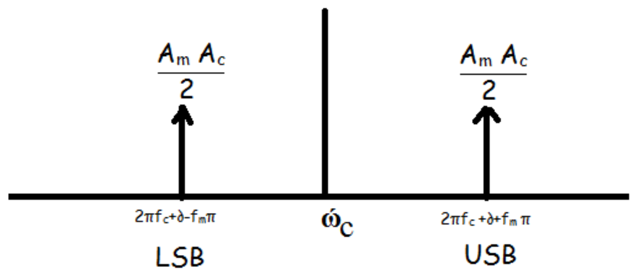
$$= A_m A_c * \text{Sin } f_m \pi t * \text{Sin}(2\pi f_c t + \partial)$$

$$= (A_m A_c/2)*[2 \text{ Sin } f_m \pi t * \text{ Sin}(2\pi f_c t + \partial)]$$

$$= (A_m A_c/2)*[2 \text{ Sin}(2\pi f_c t + \partial)* \text{ Sin } f_m \pi t]$$

$$= (A_m A_c/2)*[\text{ Cos }(2\pi f_c t + \partial - f_m \pi t) - \text{Cos }(2\pi f_c t + \partial + f_m \pi t)$$

$$= (A_m A_c/2)*[\text{ Cos }\{(2\pi f_c t + \partial - f_m \pi t) - \text{Cos }(2\pi f_c t + \partial + f_m \pi t)$$



26	x(t) = 2 sin (400 πt) + 4 cos (600 πt). Sampling of 2400 Hz. (a) Find the equation of x(n). (b) Find the period of x(n). (c) Draw the spectrum of x(n).	BUET MSC-14

(a) x(n) = 2 sin (400 πn/2400) + 4 cos (600πn/ 2400)

= 2 sin (πn/6) + 4 cos (πn/ 4) **Ans.**

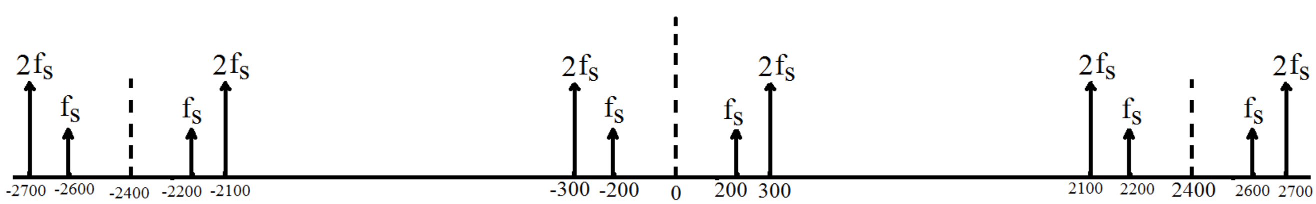
(b) x(n) = 2 sin (2πn/12) + 4 cos (2πn/ 8)

So, period of first (sin) part of the signal is 12s

period of second (cos) part of the signal is 8s

So, period of the signal is LCM (12, 8) = 24 s **Ans.** *LCM = ল.সা.গু.

(c)



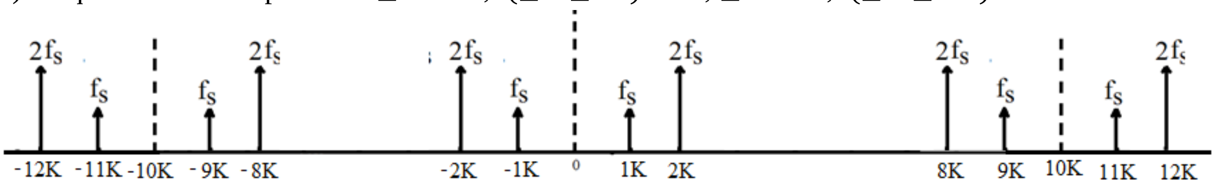
এই অংশটা বুঝতে আমার অনেক সময় লেগেছে। ফুরিয়ার ট্রান্সফর্মসহ একগাদা ফাইল পড়ে (এখন রাত 3.13 Am এ ☹) কিছুটা বুঝতে পেরেছি মনে হয়। যাহোক, এই টাইপ ম্যাথ করার একটা শর্টকাট নিয়ম মনে হয় দাঁড় করাতে পেরেছি শেষ পর্যন্ত। নিয়মটা হচ্ছে আগে ফ্রিকুয়েন্সি বের করে নেওয়া। যেমন, এই প্রশ্নে সাইন অংশের ফ্রিকুয়েন্সি 400 π /2π = 200 Hz, এবং কস অংশের ফ্রিকুয়েন্সি 600 π /2π = 300 Hz. স্যাম্পলিং ফ্রিকুয়েন্সি 2400 Hz. তাহলে স্পেকট্রাম ড্রয়িং এ ফ্রিকুয়েন্সিগুলো হবেঃ সাইন অংশের জন্য ± 200 Hz, (± 200 ± 2400) Hz, এবং কস অংশের জন্য ± 300 Hz, (± 300 ± 2400) Hz. উভয়ক্ষেত্রেই এমপ্লিচিউড অর্ধেক হয়ে যাবে।

27	If m(t)= 2 sin 2000πt + 4 sin 4000πt, then (i) Find minimum sampling frequency required to avoid aliasing (ii) If sampling frequency is 10KHz, draw the spectrum of the sampled signal.	DWASA-2014

(i) The maximum frequency of the signal, $f_m = 4000\pi/2\pi = 2000\text{Hz} = 2\text{KHZ}$

So, minimum sampling frequency required to avoid aliasing, $f_S = 2f_m = 4000$ Hz = 10 KHz **Ans**.

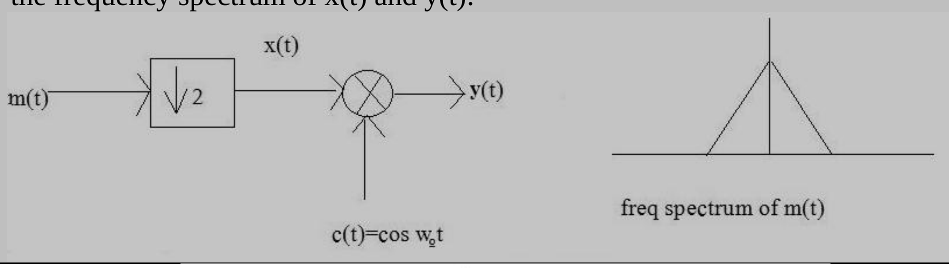
(ii) Frequencies of the spectrum: $\pm$ 1 KHz, ($\pm$ 1 $\pm$ 10) KHz, $\pm$ 2 KHz, ($\pm$ 2 $\pm$ 10) KHz.

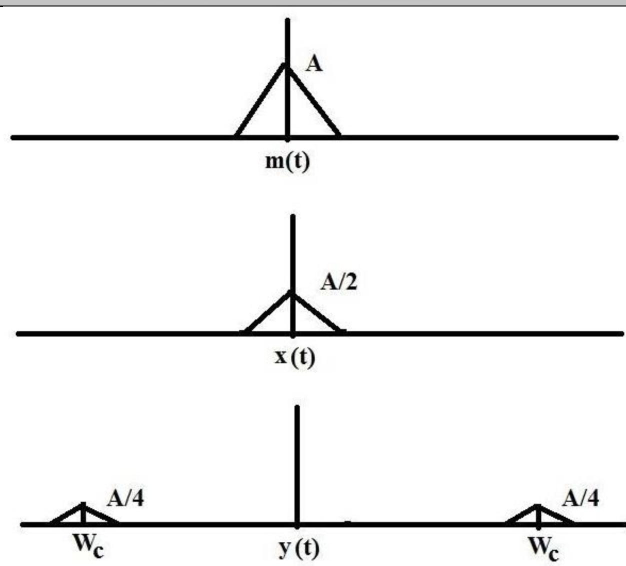


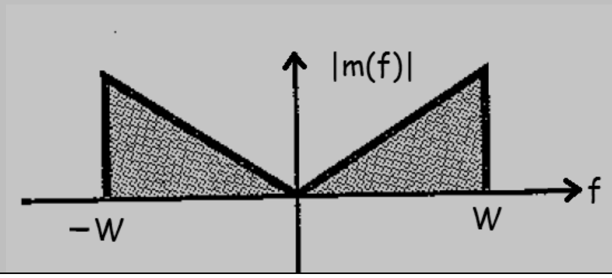
28	A signal x(t)=2 sin(400πt)+ 6 sin(640πt) is ideally sampled at 500 Hz and then fed to an ideal low pass filler with a cut-off frequency of 400Hz. Determine the frequencies that will be available at the output.	BPDB-13

In the spectrum of sampled signal, the frequencies would be $\pm$ 200 Hz, ($\pm$ 200 $\pm$ 500) Hz, $\pm$ 320 Hz, ($\pm$ 320 $\pm$ 500) Hz . But as ideal low pass filler with a cut-off frequency of 400Hz is used, so output frequencies will be less than 400Hz.

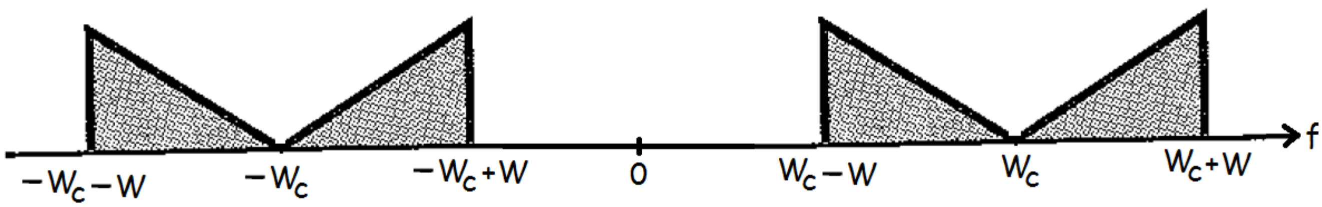
So, Output frequencies will be 200Hz, 300Hz, 320Hz, 180Hz. **Ans**.

29	Draw the frequency spectrum of x(t) and y(t): 	EGCB-14

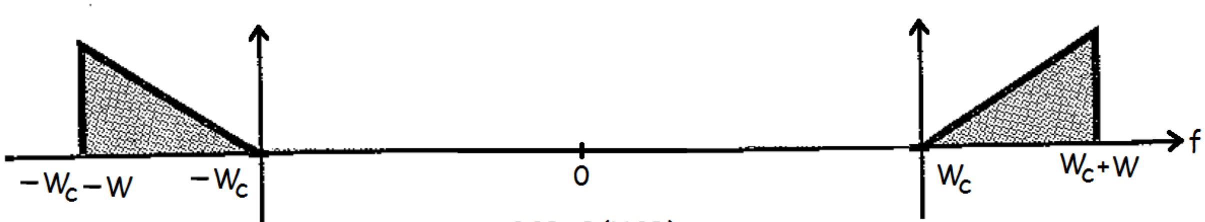


30	The spectrum of a modulating signal is shown in the figure. Draw the spectrum of DSB-SC, SSB+C, and VSB modulated signals for this modulating signal assuming a carrier signal of C (t) = $A_C \text{Cos } 2\pi f_C' t$ 	DPDC-14

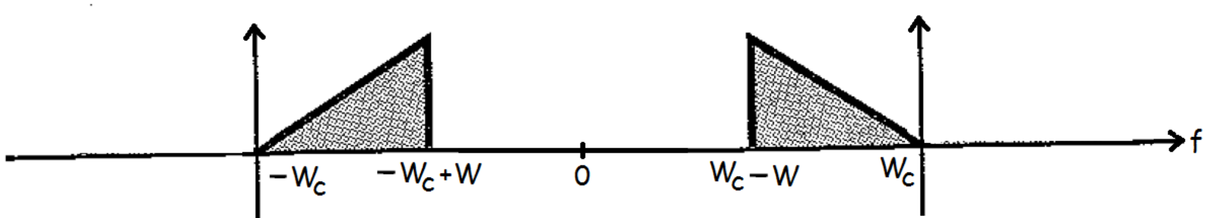
Ans:



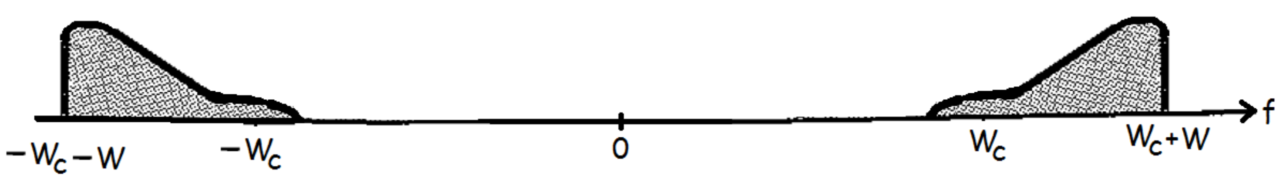
DSB-SC



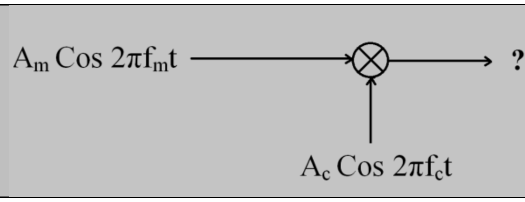
SSB+C (USB)



SSB+C (LSB)



VSB

31		BUET M.Sc. -14

Output = $A_m \cos 2\pi f_m t * A_C \cos 2\pi f_C t$

$= (A_m A_c/2)*[2 \cos 2\pi f_m t * \cos 2\pi f_c t]$

$= (A_m A_c/2)*[\cos(2\pi f_C t + 2\pi f_m t) + \cos(2\pi f_C t - 2\pi f_m t)]$

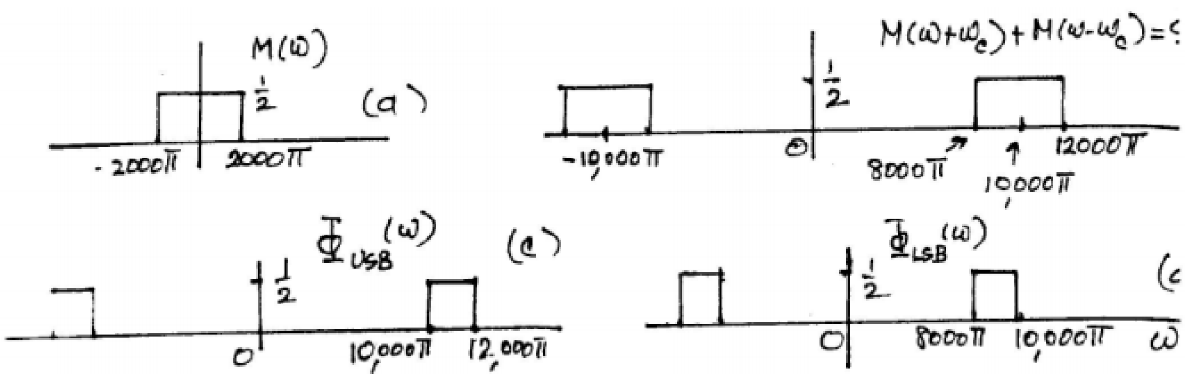
$= (A_m A_c/2)*[\cos 2\pi (f_C+f_m)t + \cos 2\pi (f_C-f_m)t]$ **Ans.**

32	If m(t) = B Sinc (2πBt) , B=1000 Hz & $\acute{\omega}_c$ =10000π, then Draw the Spectrum of DSB-SC and LSB signal.	BUET M.Sc. - 12

Related Problem:

1. The modulating signal m(t)=Bsinc(2πBt) with B=1000Hz and carrier wc=10,000 π. Please sketch the spectra of m(t) and the corresponding DSC-SC signal 2m(t)coswct, as well as USB and LSB spectra. Find the inverse Fourier transforms of LSB and USB spectra.

sol: hint : $rect(\frac{t}{\tau}) \Leftrightarrow \tau \text{sinc}(\frac{w\tau}{2})$ $\quad B\sin c(2\pi Bt) \Leftrightarrow \frac{1}{2} rect(-\frac{w}{4\pi B})$



$$\varphi_{LSB}(t) = 1000\sin c(1000\pi t)\cos 9000\pi t$$

$$\varphi_{USB}(t) = 1000\sin c(1000\pi t)\cos 11000\pi t$$

Reference: 2014 通信技术与系统作业 2 参考答案 授课教师: 梁菁

(Reference Answers of Mr. Liang Jing's Communication Technology and Systems Assignment-2, 2014)

Comment: মিস্টার জিং কাকু পিডিবি'র প্রশ্ন দেখে তাঁর স্টুডেন্টদের এসাইনমেন্ট দেন! :D

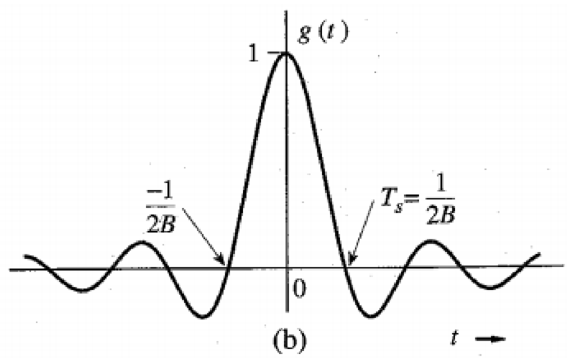
33	Find a signal g(t) that is band-limited to B Hz and whose samples are $g(0) = 1$ and $8(\pm T_S) = g(\pm 2T_S) = 8(\pm 3\ T_S) = \ldots. = 0$ where the sampling interval T_S is the Nyquist interval for g(t), that is, $T_S = 1/2B$.	DWASA-11

We use the interpolation formula

$$g(t) = \sum_k g(kT_s)\,\mathrm{sinc}\,(2\pi Bt - k\pi)$$

to construct $g(t)$ from its samples. Since all but one of the Nyquist samples are zero, only one term (corresponding to $k = 0$) in the summation on the right-hand side of Eq. survives. Thus,

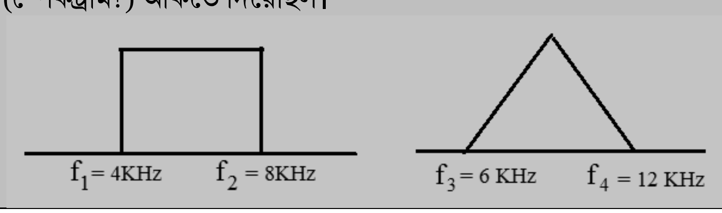
$$g(t) = \mathrm{sinc}\,(2\pi Bt)$$



(b)

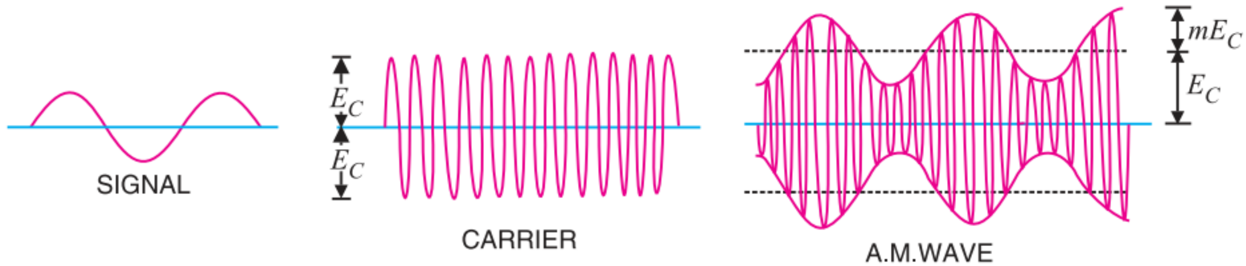
This signal is shown in Fig. Observe that this is the only signal that has a bandwidth B Hz and with the sample values $g(0) = 1$ and $g(nT_s) = 0\ (n \neq 0)$. No other signal satisfies these conditions.

Refernce: B.P. Lathi

35	একটা square এবং একটা triangular wave দেওয়া ছিল। এর ফ্রিকুয়েন্সি মডুলেটেড সিগন্যালের কি যেন (স্পেকট্রাম?) আঁকতে দিয়েছিল।  	DWASA-14

প্রশ্নটাই বুঝতে পারিনি ঠিকমত

Basics of Type-4: Amplitude Modulation & Power of Modulated wave



A carrier wave may be represented by :

$e_c = E_C \cos \omega_c t$ where e_c = instantaneous voltage of carrier

E_C = amplitude of carrier

$\omega_c = 2\pi f_c$

= angular velocity at carrier frequency f_c

A message signal can be represented by :

$e_s = E_S \cos \omega_s t$ where e_s = instantaneous voltage of signal

$E_S = m E_C$ = amplitude of signal

m = modulation index

$\omega_s = 2\pi f_s$ = angular velocity at signal frequency f_s

Amplitude of AM wave $= E_C + m E_C \cos \omega_s t = E_C(1 + m \cos \omega_s t)$

$$\boxed{\begin{aligned}&\text{The instantaneous voltage of AM wave} = \text{Amplitude} \times \cos \omega_c t\\ &= E_C \cos \omega_c t + \frac{mE_C}{2} \cos(\omega_c + \omega_s)t + \frac{mE_C}{2}\cos(\omega_c - \omega_s)t\end{aligned}}$$

The following points may be noted from the above equation of amplitude modulated wave:

(*i*) The AM wave is equivalent to the summation of three sinusoidal waves; one having amplitude E_C and frequency **f_c, the second having amplitude $mE_C/2$ and frequency $(f_c + f_s)$ and the third having amplitude $mE_C/2$ and frequency $f_c - f_s$.

(*ii*) The AM wave contains three frequencies *viz* f_c, $f_c + f_s$ and $f_c - f_s$. The first frequency is the carrier frequency. Thus, the process of modulation does not change the original carrier frequency but produces two new frequencies $(f_c + f_s)$ and $(f_c - f_s)$ which are called sideband frequencies.

(*iii*) The sum of carrier frequency and signal frequency *i.e.* $(f_c + f_s)$ is called *upper sideband frequency*. The *lower sideband frequency* is $f_c - f_s$ *i.e.* the difference between carrier and signal frequencies.

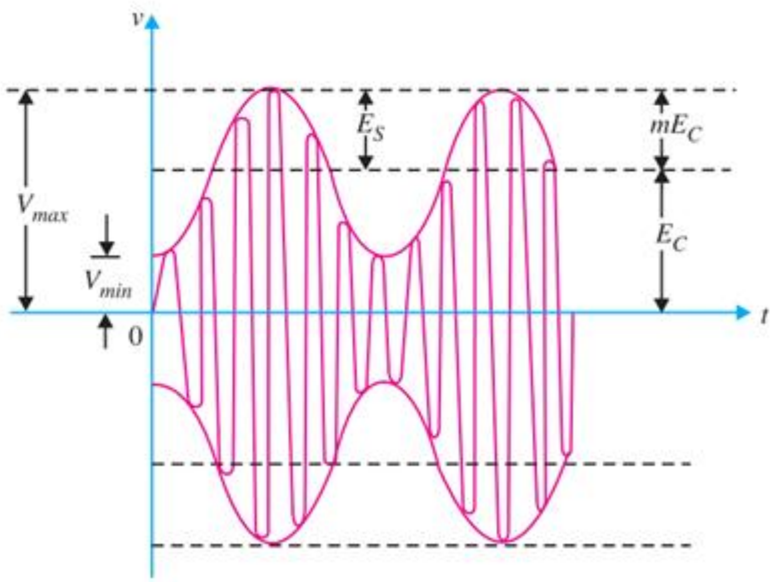
* $P_T = P_C + P_S$
* $\frac{P_S}{P_T} = \frac{m^2}{2+m^2}$
* $\frac{P_S}{P_C} = \frac{1}{2}m^2$

P_C = Carrier power,
P_S = Total power of sidebands,
P_T = Total power of AM wave,
* r.m.s. values are considered.

$$E_C = \frac{V_{max} + V_{min}}{2}$$

$$E_S = \frac{V_{max} - V_{min}}{2}$$

$$m = \frac{E_S}{E_C} = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$$



36	A 1KW Carrier is amplitude modulated to a depth of 60%. Calculate total power and Sideband Power of the modulated wave.	BUET M.Sc. 12

Carrier Power, P_C = 1KW

Modulation index, m = depth of modulation = 60% =0.6

Sideband Power, $P_S = (m^2/2)* P_C = (0.6^2/2)*1 = 0.18$ KW **Ans.**

Total Power, $P_T = P_C + P_S = 1+0.18$ KW =1.18 KW **Ans.**

37	If m(t) = B cos ω_m t and index μ=1, then find Φ_{AM} (t) and sketch it.	DWASA-11

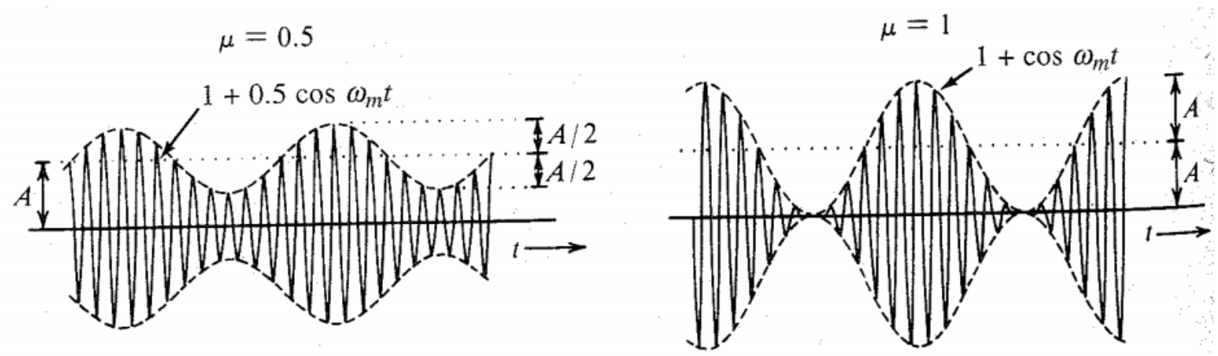
Some books use μ to represent modulation index instead of m
And B to represent Amplitude of Signal instead of E_S
And A to represent Amplitude of Carrier instead of E_C

$$\text{Amplitude of AM wave} = E_C + m E_C \cos \omega_s t = E_C (1 + m \cos \omega_s t)$$

$$= A(1+\mu \cos \omega_m t)$$

$$\text{The instantaneous voltage of AM wave} = \text{Amplitude} \times \cos \omega_c t$$

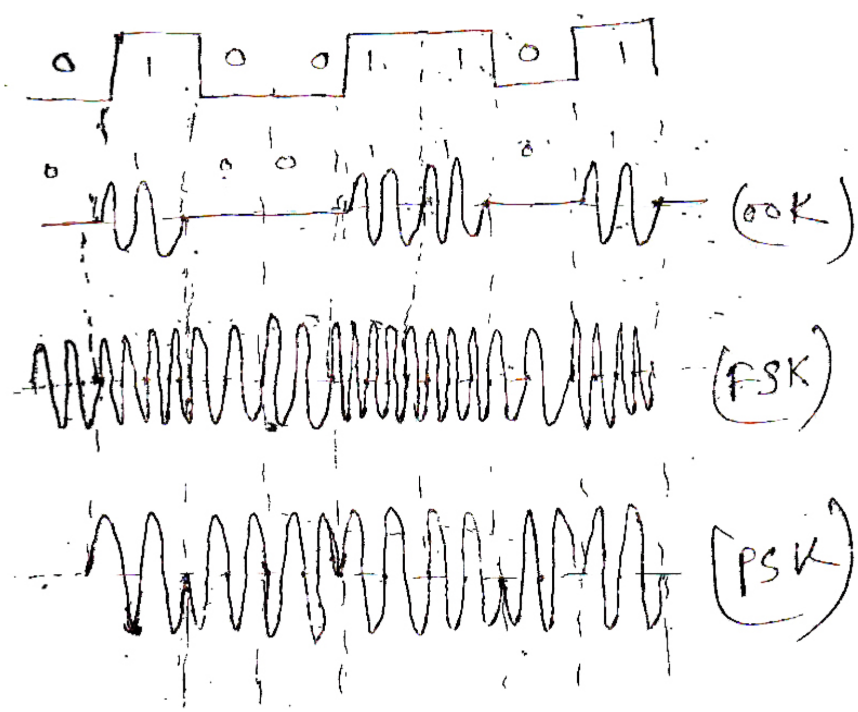
So, $\Phi_{AM}(t) = A(1+\mu \cos \omega_m t) \cos \omega_C t$ **Ans.**



(a) $\mu = 0.5$ (b) $\mu = 1$

Type-5: Others

37	Demonstrate OOK, FSK, PSK signal assuming a bit sequence 01001101.	EGCB-12



38	Write down the advantages and limitations of digital communication.	PGCL-11

Advantages:

- ❖ Reliable communication; less sensitivity to changes in environmental conditions (temperature, etc.)
- ❖ Easy multiplexing

- Easy signaling
    - Hook status, address digits, call progress information
- Voice and data integration
- Easy processing like encryption and compression
- Easy system performance monitoring
    - QOS monitoring
- Integration of transmission and switching
- Signal regeneration, operation at low SNR, superior performance
- Integration of services leading to ISDN

Disadvantages:

- Increased bandwidth

    64 KB for a 4 KHz channel, without compression (However, less with compression)
- Need for precision timing

„ Bit, character, frame synchronization needed
- Analogue to Digital and Digital to Analogue conversions

    „ Very often non-linear ADC and DAC used, some performance degradation
- Higher complexity

[Reference: Bangalore NPTEL Course material]

39	The main reason for the superiority of digital communication over analog communication is (a) The use of simple electronic circuitry. (b) The use of amplifiers periodically (c) The use of regenerative repeaters (d) The use of A/D and D/A converters	MCQ DPDC -14

One main reason for the superior quality of digital systems over analog ones is **the viability of regenerative repeaters** and network nodes in the former.

(Reference: Chapter:1, B.P. Lathi, Topic: 1.2.2 Viability of distortionless regenerative repeaters)

40	The main advantage of a digital communication system over that of an analog one is (a) reduced complexity of the receiver (b) robustness to noise (c) use of regenerative repeaters (d) all of the above	MCQ BPDB-13

One main reason for the superior quality of digital systems over analog ones is **the viability of regenerative repeaters** and network nodes in the former.

(Reference: Chapter:1, B.P. Lathi, Topic: 1.2.2 Viability of distortionless regenerative repeaters)

<u>একটি মজার বিষয়ঃ</u>

৩৯ এবং ৪০ নম্বর প্রশ্নের সঠিক উত্তর বের করার জন্য আমাকে অনেক খোঁজাখুঁজি করতে হয়েছে। বিশেষ করে ৩৯ নাম্বারের প্রায় সবগুলোকেই সঠিক মনে হচ্ছিল। কিন্তু সঠিক উত্তর তো হবে ১ টা! শেষমেষ ৩৯ এবং ৪০ নাম্বারের অপশানগুলো মিলিয়ে দেখি শুধু রিপিটারের অপশানটাই ২টাতে কমন। তাই ভেবেছিলাম এটাই উত্তর হবে। শেষে অবশ্য লাথি'র বইতে সরাসরিই পেয়ে যাই লাইনটা। অনেক সময় একটা প্রশ্নের ভেতরই আরেকটা প্রশ্নের উত্তর লুকিয়ে থাকে, ৩৯ এবং ৪০ নং এমসিকিউ তার একটা উদাহরণ।

41	Explain the slop overload effect of delta modulation.	BPDB-11

Delta modulation (DM or Δ-modulation) is an analog-to-digital and digital-to-analog signal conversion technique used for transmission of voice information. DM attempts to **quantize** an input signal using a **simple comparison algorithm**. Instead of measuring the signal level, it measures the **difference between the level of the input signal** from the **beginning to the end** of a sampling period.

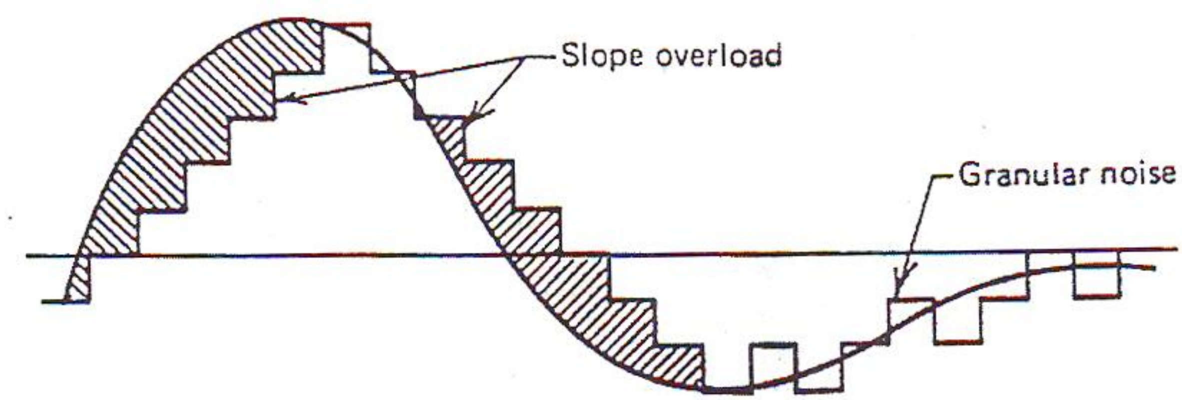
Delta Modulation attempts to represent an analog signal with a resolution of 1 bit. This is accomplished by successive steps, either up or down, by a preset step size. In delta modulation, we have the step size (Δ) that is defined for each sampler, and we have the following rules for output:

1. If the input signal is higher than the current reference signal, increase the reference by Δ, and output a 1.
2. If the input signal is lower than the current reference signal, decrease the reference by Δ, and output a 0.

Some benefits of delta modulation are as follows:

- 1 bit of resolution, and therefore requires very little bandwidth and very little hardware.
- No preset upper or lower bounds, so Delta modulation can (theoretically) be used to modulate unbounded signals.

These benefits are countered by the problems of **Slope Overload**, and **Granular Noise**, which play an important role when designing a Delta Modulated system.



Slope Overload

If the input signal is rising or falling with a slope larger than Δ/T, where T is the sampling time and the Δ is the size of the individual steps, we say that the sampler is suffering from **Slope Overload**. When the slope of the sound waveform **exceeds the ability of DM's step size to keep up**, this creates **infidelity** known as slope overload distortion.

Granular Noise

A problem with delta modulation is that the output signal must always either increase by a step, or decrease by a step, and cannot stay at a single value. This means that if the input signal is level, the output signal could potentially be oscillatory. That is, the output signal would appear to be a wave, because it would go up and down regularly. This phenomena is called **Granular Noise**.

When used in ADCs (Analog to Digital Converters), this problem can be solved by internally adding additional bit(s) of resolution that correspond to the value of Δ. This way, the LSBs (Least significant bits) that were added can be ignored in the final conversion result.

42	What is power line communication? Give some example.	BPDB-12

Power-line communication (PLC) carries data on a conductor that is also used simultaneously for AC electric power transmission or electric power distribution to consumers.

It is also known as power-line carrier, power-line digital subscriber line (PDSL), mains communication, power-line telecommunications, or power-line networking (PLN).

Examples:
A wide range of power-line communication technologies are needed for different applications, ranging from **home automation to Internet access** which is often called **broadband over power lines (BPL).** Most PLC technologies limit themselves to one type of wires (such as premises wiring within a single building), but some can cross between two levels (for example, both the distribution network and premises wiring).
[Reference: Wikipedia]

43	Write a few applications of Power Line Carrier Communication (PLCC).	BPDB-13

Applications of PLCC

PLCC technology can be deployed into different types of applications in order to provide economic networking solutions. Hence merging with other technologies it proves useful in different areas. These are few key areas where PLC communications are utilized:

a. **Transmission & Distribution Network:** PLCC was first adopted in the electrical transmission and distribution system to transmit information at a fast rate.

b. **Home control and Automation:** PLCC technology is used in home control and automation. This technology can reduce the resources as well as efforts for activities like power management, energy conservation, etc.

c. **Entertainment:** PLCC is used to distribute the multimedia content throughout the home.

d. **Telecommunication:** Data transmission for different types of communications like telephonic communication, audio, video communication can be made with the use of PLCC technology.

e. **Security Systems:** In monitoring houses or businesses through surveillance cameras, PLCC technology is far useful.

f. **Automatic Meter Reading** – Automatic Meter reading applications use the PLCC technology to send the data from home meters to Host Central Station.

44	Abbreviate: VSAT, WiMAX, WLAN, ADSL, SONET, OFDMA	DWASA-14

VSAT : *Very Small Aperture Terminal*
WiMAX : Worldwide Interoperability for Microwave Access
WLAN : Wireless Local Area Network
ADSL : Asymmetric Digital Subscriber Line
SONET : Synchronous Optical Network
OFDMA : Orthogonal Frequency-Division Multiple Access

45	What is meant by: OFDM, GMSK, WiMAX, DWDM, PSTN, BISDT.	BUET M.Sc. Unknown

OFDM : Orthogonal Frequency Division Multiplexing
GMSK : Gaussian Minimum Shift Keying
WiMAX : Worldwide Interoperability for Microwave Access
DWDM : *Dense Wavelength Division Multiplexing*
PSTN : Public Switched Telephone Network
BISDT :
BISDT নামের কিছু খুঁজে পাইনি আমি।কাছাকাছি ২ টা শব্দ পাওয়া গেছে। সেগুলা নীচে দিলামঃ
BISDN : Broadband Integrated Services Digital AT&T Network
BIST : Built-In Self Test

46	What is erlang of telephone traffic? Related Math.	BUET M.Sc. Unknown

Erlang is a unit of telecommunications traffic measurement. It is used to describe the total traffic volume of one hour.

For example, if a group of user made 30 calls in one hour, and each call had an average call duration of 5 minutes, then the number of Erlangs this represents is worked out as follows:

Minutes of traffic in the hour = number of calls x duration
= 30 x 5=150
Hours of traffic in the hour = 150 / 60
= 2.5
Traffic figure = **2.5 Erlangs**

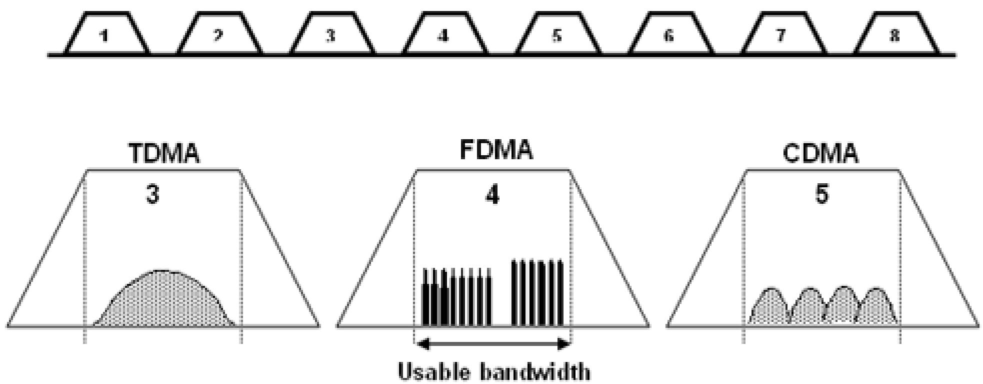
Erlang traffic measurements are made in order to help telecommunications network designers understand traffic patterns within their voice networks. This is essential if they are to successfully design their network topology and establish the necessary trunk group sizes.

Erlang traffic measurements or estimates can be used to work out how many lines are required between a telephone system and a central office (PSTN exchange lines), or between multiple network locations.

47	What are the Common Multiple Access Technologies? Differentiate between Multiplexing & Multiple Access Technologies.	BUET M.Sc. 12

There are three types of common Multiple Access Methods :

- Frequency Division Multiple Access (FDMA) - flexible and simple
- Time Division Multiple Access (TDMA) - popular
- Code Division Multiple Access (CDMA Spread Spectrum) - highly secure



This illustration shows how the most common Multiple Access Methods allow the capacity of a standard transponder to be shared.

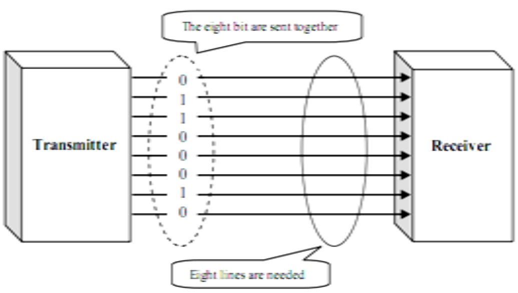
Difference between Multiplexing and Multiple Access:

Multiplexing	Multiple Access
Multiplexing is a process where **multiple analog message signals or digital data streams are combined into one signal** over a shared medium.	Multiple access method allows **several terminals connected to the same multi-point physical medium** to transmit over it and to share its capacity."
It works on the **physical layer** (L1) of OSI model.	It works on the **Data Link** layer (L2) of OSI model.
A device that performs the multiplexing is called a multiplexer (MUX), and a device that performs the reverse process is called a demultiplexer (DEMUX).	A channel-access scheme is also based on a Multiple access protocol and control mechanism, also known as media access control (MAC). This protocol deals with issues such as addressing, assigning multiplex channels to different users, and avoiding collisions.

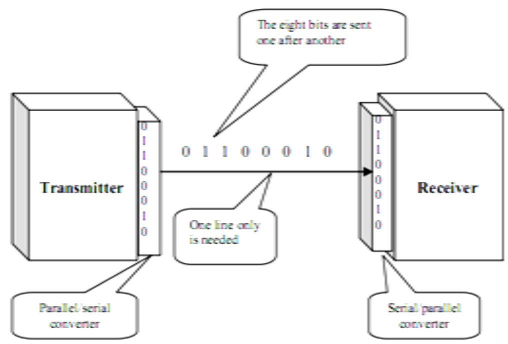
48	why is parallel transmission more useful than serial transmission? (a) For long distance data transmission (b) For short distance data transmission (c) For synchronous transmission (d) For Asynchronous transmission	MCQ DPDC-14

Parallel data is preferred for short distance data transmission because the amount of data that can be sent is greater than serial data transmission. However in long range transmission, the data along a wire could get distorted by the voltages from the other wires parallel to the wire. This problem is known as skew. Another reason it is not used over long distances is that the cost of cabling is very high as many wires are required.

Serial Data Transmission - Single bits are sent one after another along a single data channel.
Parallel Data Transmission - Bits are sent down several data channels simultaneously

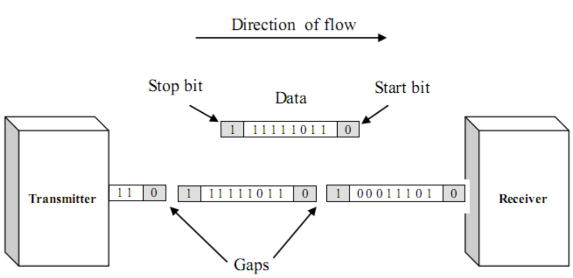


Parallel Transmission

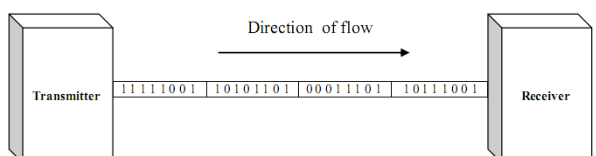


Serial Transmission

Serial transmission occurs in one of two ways; asynchronous or synchronous.



Asynchronous transmission



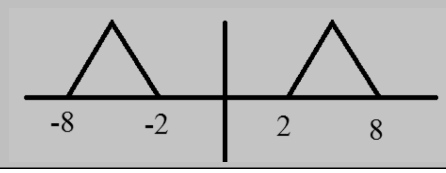
Synchronous transmission

49	Envelop detector is helpful for which of the following modulation? (এই টাইপ কিছু একটা ছিল) (a) ASK (b) ASK and FSK (c) FSK (d) PSK	MCQ DPDC-14

Digital radio links require a digitally modulated carrier. Options are ampli-tude shift **keying (ASK), normally used with envelope detection**, frequency shift keying **(FSK), also using envelope detection**, or phase shift keying (PSK). **PSK requires coherent detection**, either by regenerating a local car-rier in the receiver, or by using differential detection in which the previous bit is used as the phase reference for the current bit.

কাজেই সঠিক উত্তর সম্ভবতঃ (b) ASK and FSK

Reference: Topic: 4.2.8 Digital Radio Links, Handbook of Electrical Engineering Calculations by Arun G. Phadke

50	Find the probable bandwidth of the following signal 	MCQ DWASA-14

Bandwidth = USB – LSB = 8 – (-2) = 10 Hz

Or, 2-(-8) = 10 Hz. **Ans.**

51	Inter-symbol interference occurs when (a) channel bandwidth (BW) is close to the signal BW (b) signal BW is much larger than channel BW (c) channel BW is much larger than signal BW (d) channel BW is large as signal BW	MCQ BPDB-13

When the **signal's bandwidth becomes larger than the channel bandwidth**, the channel starts to introduce distortion to the signal. This distortion usually manifests itself as **intersymbol interference.** *[Reference: http://en.wikipedia.org/wiki/Pulse_shaping]*

52	Which one of the following is a valid uplink frequency band used in a GSM system (a) 1930-1990 MHz (b) 890-915 MHz (c) 440-460 MHz (d) 935-960 MHz	MCQ BPDB-13

System	GSM 900	GSM 1800	UMTS 2100
Uplink	**890-915 MHz**	1710-1785 MHz	1920 - 1980 MHz
Downlink	935-960 MHz	1805-1880 MHz	2110 – 2170 MHz

53	For modulation, a GSM system generally employs (a) GMSK (b) 8-PSK (c) QPSK (d) both (a) and (b)	MCQ BPDB-13, MCQ BUET M.Sc.-13

Communication system	Used modulation scheme
GSM 2G GPRS 2.5G	**GMSK**
EDGE 2.75G	8 PSK
CDMA 2000	QPSK in forward Channel (From BTS to Mobile) OQPSK in reverse channel
UMTS 3G	QPSK
HSDPA 3.5G	Adaptive Modulation: QPSK 16 QAM
Wi-Fi	BPSK, QPSK, 16 QAM, 64 QAM
WiMax	Adaptive Modulation: QPSK, 16 QAM, 64 QAM

54	Which statement is TRUE regarding analog modulation techniques? (a) FM signal offers better receptive quality compared with AM because it has narrower bandwidth than that of AM (b) FM signal is more noise resistant than PM signal (c) Synchronous detection can be used for AM, and PM signals (d) None of the above	MCQ BPDB-13

- এফ এম এর রিসেপটিভ কোইয়ালিটি বেটার প্রশ্নে দেওয়া আছে ,এর ব্যান্ডউইডথ তুলনামূলক বেশি বলে। কিন্তু , তাই !কম বলেa সম্ভবত হবেনা।
- B হতে পারে। কারণ এফ এম বেশি নয়েজ রেজিস্ট্যান্ট।
- AM এ সিনক্রোনাস ডিটেকশান ব্যবহৃত হয়। কিন্তু PM এ হয় এরকমটা কোথাও খুঁজে পেলাম না।

	AM	FM
Modulating differences	In AM, a radio wave known as the "carrier" or "carrier wave" is modulated in amplitude by the signal that is to be transmitted. The frequency and phase remain the same.	In FM, a radio wave known as the "carrier" or "carrier wave" is modulated in frequency by the signal that is to be transmitted. The amplitude and phase remain the same.
Pros and cons	AM has poorer sound quality compared with FM, but is cheaper and can be transmitted over long distances. It has a lower bandwidth so it can have more stations available in any frequency range.	FM is less prone to interference than AM. However, FM signals are impacted by physical barriers. FM has **better sound quality due to higher bandwidth.**
Frequency Range	AM radio ranges from 535 to 1705 KHz (OR) Up to 1200 bits per second.	FM radio ranges in a higher spectrum from 88 to 108 MHz. (OR) 1200 to 2400 bits per second.
Bandwidth Requirements	Twice the highest modulating frequency. In AM radio broadcasting, the modulating signal has bandwidth of 15kHz, and hence the bandwidth of an amplitude-modulated signal is 30kHz.	Twice the sum of the modulating signal frequency and the frequency deviation. If the frequency deviation is 75kHz and the modulating signal frequency is 15kHz, the bandwidth required is 180kHz.
Zero crossing in modulated signal	Equidistant	Not equidistant
Complexity	Transmitter and receiver are simple but synchronization is needed in case of	Transmitter and receiver are more complex as variation of modulating signal has to

	AM	FM
	SSBSC AM carrier.	beconverted and detected from corresponding variation in frequencies.(i.e. voltage to frequency and frequency to voltage conversion has to be done).
Noise	AM is more susceptible to noise because noise affects amplitude, which is where information is "stored" in an AM signal.	FM is less susceptible to noise because information in an FM signal is transmitted through varying the frequency, and not the amplitude.

55	A discrete time signal is given by $x(n) = \text{Cos}\,[(n\pi)/9]$. The signal is (a) periodic with period N=9 samples. (b) periodic with period N=18 samples. (c) periodic with period N=32 samples. (d) aperiodic	MCQ BPDB-15

$f= (\pi/9)*(1/2\pi) = 1/18$. So, The signal is periodic with sampling period of 18.

56	What is the carrier in Submarine Cable?	MCQ BUET M.Sc.-13

প্রশ্ন বুঝিনি ঠিকমতো। ☹

Modern submarine cables use optical fiber technology to carry digital data, which includes telephone, Internet and private data traffic.