Rajeev Gandhi Memorial College of Engineering and Technology, Nandyal
(Autonomous)
Department of Electronics and Communication Engineering

COURSE FILE

ACADEMIC YEAR : 2022-23
SUBJECT : Analog communication
YEAR /SEM : III B.Tech I-Sem
DEPARTMENT : Electronics & Communication Engineering
FACULTY NAME : Mr. D.Rajesh Setty

S. No	Content	Page No.
1	Time Table	1
2	Syllabus & Lesson Plan	2 – 5
3	Attendance Register	6 - 15
4	Course Material, PPTs	16 – 312
5	Assignment 1, Mid 1Questions and answer scripts	313 – 333
6	Assignment 2, Mid 2 Questions and answer scripts	334 – 357
7	External Question Paper with solutions	358 – 379
8	External Exam Answer Script	380 – 414
9	CO-PO Calculation	415 – 423
10	Contents covered beyond Syllabus	424 - 429
11	Expert Suggestion	430
12	Self Assessment	431

Dr. Kethepalli Mallikarjuna
HOD of ECE

Dr. KETHEPALLI MALLIKARJUNA
Professor & HOD
Department of ECE
RGM College of Engg. & Tech. (Autonomous)
NANDYAL - 518 501, Kurnool (Dist), A.P.

Dr.T.Jayachandra prasad
Principal

Dr. T. JAYACHANDRA PRASAD M.E., Ph.D.
PRINCIPAL
R.G.M College of Engg. & Tech.,
(Autonomous)
NANDYAL-518 501, Nandyal (Dist),A.P.

R.G.M.COLLEGE OF ENGINEERING & TECHNOLOGY, NANDYAL – 518 501
DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

III B.Tech., I-Semester **Academic Year: 2022-23**
w.e.f: 01-08-2022

A-Section : RB3130 B-Section : RB3010
C-Section : RB3020 D-Section : RB3030

Period/ Day	Section	1 **9.00 AM To 9.50 AM**	2 **9.50 AM To 10.40 AM**	3 **11.00 AM To 11.50 AM**	4 **11.50 AM To 12.40 PM**	5 **1.50 PM To 2.40 PM**	6 **2.40 PM To 3.30 PM**	7 **3.30 PM To 4.20 PM**
MON	A	LDICA	AWP	AUD	AC	LDICA	CSE	AWP
	D	AUD	AC	CSE	IHC	AWP	AC	LDICA
	B	LDICA Lab/AC Lab			AWP	CSE	LDICA	AUD
	C	AWP	CSE	COU	LDICA	AUD	AC	IHC
TUE	A	AUD	AWP	AC	CSE	LDICA Lab/AC Lab		
	D	LDICA	AC	IEI/ISTE	AWP	CBP		
	B	AWP	CSE	AUD	AC	IHC	COU	LDICA
	C	AC	LDICA	AWP	CSE	LDICA	AC	CSE
WED	A	LDICA	IHC	CSE	IEI/ISTE	AUD	AC	LIB
	D	AWP	LDICA Lab/AC Lab			AC	CSE	LDICA
	B	LDICA	CSE	AUD	AC	CBP		
	C	AUD	AC	IHC	AWP	LDICA Lab/AC Lab		
THU	A	AC	AWP	CSE	LDICA	CSE	COU	IHC
	D	CSE	LDICA	AWP	AC	AWP	CSE	AUD
	B	AC	CSE	AWP	LDICA	LDICA Lab/AC Lab		
	C	LDICA	AUD	AC	CSE	AWP	IEI/ISTE	LIB
FRI	A	AWP	CSE	LDICA	AC	CBP		
	D	AC	LDICA	CSE	AUD	LDICA Lab/AC Lab		
	B	AUD	IHC	CSE	AWP	CSE	AC	LDICA
	C	CSE	LDICA Lab/AC Lab			LDICA	AWP	AUD
SAT	A	LDICA Lab/AC Lab			AWP	LDICA	AC	AUD
	D	AWP	IHC	LIB	LDICA	AUD	CSE	COU
	B	AC	LDICA	AWP	LIB	AC	AWP	IEI/ISTE
	C	CSE	AWP	AC	LDICA	CBP		

Subject	Section	Name of the Faculty
LDICA	A	Dr.N.Ramanjaneyulu
AC	A	Smt.L.Jyothi Reddy
AWP	A	Smt.K.Mounika
CSE	A	Mr.C.Dastagiraiah
AUD	A	Mr.K.Nagendra Kumar
IHC	A	Mr.C.Parameswara Reddy
LDICA Lab	A	**Dr.NR/KM/CD**
AC Lab	A	**Mr.SVRK**/Miss.NFS/LJR
Counselling	A	Mr.M.V.Raja Sekar
IEI/ISTE	A	Dr.N.Ramanjaneyulu
CBP	A	Dr.K.Mallikarjuna

Subject	Section	Name of the Faculty
LDICA	B	Smt.N.Sireesha
AC	B	Miss.L.Ujjawala
AWP	B	Mr.S.Kashif Hussain
CSE	B	Mrs.G.I.Lakshmi Prasanna
AUD	B	Dr.P.V.Gopi Krishna Rao
IHC	B	Mr.M.Venkateswara Reddy
LDICA Lab	B	**Smt..BN**/SLVK/CD
AC Lab	B	**Smt.MM**/Mr.DRS/GILP
Counselling	B	Mr.M.V.Raja Sekar
IEI/ISTE	B	Dr.N.Ramanjaneyulu
CBP	B	Dr.K.Mallikarjuna

Subject	Section	Name of the Faculty
LDICA	C	Smt.B.Nazma
AC	C	Mr.D.Rajesh Setty
AWP	C	Mr.S.L.Venkata Krishna
CSE	C	Smt.R.Sireesha
AUD	C	Dr.P.V.Gopi Krishna Rao
IHC	C	Mr.M.Venkateswara Reddy
LDICA Lab	C	**Dr.PVGKR**/Smt.BN/SLVK
AC Lab	C	**Mr.DRS**/Smt.MM/GILP
Counselling	C	Mr.M.V.Raja Sekar
IEI/ISTE	C	Dr.N.Ramanjaneyulu
CBP	C	Dr.K.Mallikarjuna

Subject	Section	Name of the Faculty
LDICA	D	Mrs.M.Prasanthi
AC	D	Mr.S.V.Ratan Kumar
AWP	D	Dr.V.N.V.Satya Prakash
CSE	D	Mr.P.Ananda Rao
AUD	D	Mr.K.Nagendra Kumar
IHC	D	Mr.C.Parameswara Reddy
LDICA Lab	D	**Dr.VNVSP/KM/MH**
AC Lab	D	**Miss.NFS**/Mr.SVRK/LJR
Counselling	D	Mr.M.V.Raja Sekar
IEI/ISTE	D	Dr.N.Ramanjaneyulu
CBP	D	Dr.K.Mallikarjuna

Dr.Kethepalli Mallikarjuna
HOD OF ECE

Dr.KETHEPALLI MALLIKARJUNA
M.Tech, Ph.D, MISTE, FIETE, MIE
Professor & HOD
Department of ECE
RGM College of Engg. & Tech. (Autonomous)
NANDYAL - 518 501, Kurnool (Dist), A.P.

Dr.T.Jaya Chandra Prasad
Principal

Dr. T. JAYACHANDRA PRASAD
M.E,Ph.D.,
PRINCIPAL
R G M College of Engg. & Tech.,
(Autonomous)
NANDYAL-518 501, Nandyal (Dt), A.P.

III B.Tech, I-Sem (ECE) RGM-R-2020

L	T	C
2	1	3

(A0413205) ANALOG COMMUNICATION

COURSE OBJECTIVES:

- To study the fundamental concept of the communication systems.
- To study various analog modulation techniques.
- To study various transmitters and receivers.
- To study the influence of noise in communication systems.

COURSE OUTCOMES:

At the end of this course the students are able to;

- Acquire the basic knowledge of various Continuous Wave and pulse Modulation techniques and distinguish between these systems.
- Comprehend the basics of Communication System and Analog Modulation Methods.
- Apply the basic knowledge of signals and systems and basic mathematics while analyzing the modulation systems mathematically.
- Choose and design appropriate type of transmitter and receiver for the given area of communication system.
- Recognize and purge the effects of noise on different modulation systems.

MAPPING WITH COS & POS:

	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12	PSO1	PSO2	PSO3
CO1	3													2	
CO2	2	3	1										1		
CO3	3	3	2											2	1
CO4		3	3	2									2		
CO5	2	2	3	2											3

UNIT I

INTRODUCTION TO COMMUNICATION SYSTEMS: Communication process, Elements of Communication Systems; Types of communication systems, its frequency ranges, Modulation: Need for Modulation, Types of modulation: AM, FM, PM, Advantages, Disadvantages and Applications.

UNIT II

LINEAR CW MODULATION: Introduction, Mathematical Representation of AM, single tone modulation index and multi tone modulation index, Power Relationships, AM signal generation (Square law, switching modulation), demodulation (Envelop detector), Virtues and Limitations of AM.

DSB-SC: Mathematical Representation of DSB-SC, DSB-SC generation (Ring modulation), Demodulation: Coherent detection, filtering of AM Signals and Spectra, DSB signals and spectra.

SSB-SC: Filtering of sidebands, SSB signal generation and demodulation using Hilbert transform, VSB Generation and demodulation, illustrative problems.

UNIT III

ANGLE CW MODULATION: Introduction, Types of Frequency Modulation (FM), Mathematical representation of FM, Modulation index, Deviation sensitivity, Deviation ratio, Transmission bandwidth of FM (Carson's rule), Narrow band FM, Wide band FM, generation of FM: Direct Method, Indirect Method, demodulation of FM. Voltage and Power for FM, Pre-emphasis and De-emphasis, Illustrative Problems.

Phase modulation and indirect FM; FM demodulators: Slope detector, Balanced slope discriminators, Phase difference discriminators, Ratio detector, PLL Detector.

UNIT IV

TRANSMITTERS AND RECEIVERS: AM TRANSMITTERS: Low level and high level transmitters, FM TRANSMITTERS: Armstrong FM transmitters, RECEIVERS: TRF receivers, super heterodyne receiver.

UNIT V

PULSE MODULATION TECHNIQUES: Definition, Types: PAM, PWM, PPM, Sampling, Nyquist rate, Different sampling techniques, Generation and Detection of PAM, PWM, PPM.

UNIT VI

NOISE IN COMMUNICATION SYSTEMS: Introduction, Noise in Base band Systems, Noise figure, different types of noise, System Model and Parameter, SNR at the output of a Base band System. Noise in AM systems: System model and parameter, Noise in DSB and SSB Systems. Noise in Angle Modulation Systems: Output SNR in Angle Modulation, Threshold effects in Angle Modulation Systems. Improvement of SNR using Pre-emphasis and De-emphasis.

TEXT BOOKS:

1. Simon Haykin, "Communication Systems", Wiley-India edition, 5th edition, 2010.
2. B.P. Lathi, ZhiDing, & Hari Mohan Gupta, "Modern Digital & Analog Communication Systems", Oxford University Press, 4th edition, 2017.

REFERENCES:

1. A.Bruce Carlson, & Paul B. Crilly, "Communication Systems – An Introduction to Signals & Noise in Electrical Communication", McGraw-Hill International Edition, 5th Edition, 2012.
2. Electronic Communication Systems by George Kennedy, Bernard Davis, S. R. M Prasanna, McGraw Hill Education, 6th edition 2017.
3. Linear Integrated Circuits – D. Roy Chowdhury, New Age International (p) Ltd, 5th Edition, 2018.
4. Digital and Analog Communication Systems, Sham Shanmugam, Wiley, 1st edition, 2019.
5. "Electronic Communications systems" Modulation and Transmission-Robert Schoenbeck, Prentice Hall India Learning Private Limited, 2nd edition, 1998.

RGM COLLEGE OF ENGINEERING & TECHNOLOGY (AUTONOMOUS)
NANDYAL-518501

DEPARTMENT OF ELECTRONICS & COMMUNICATION ENGINEERING

LESSON PLAN

Name of the Faculty: D.Rajesh Setty
Year & Semester: III B. Tech, I Semester
Subject: **ANALOG COMMUNICATION (A0413205)**

Academic Year : 2022-23
Branch & Section: ECE- C
Total Hours: 70

Unit	Topic	No. of Periods	Total Periods per unit
I	**UNIT-I: INTRODUCTION TO COMMUNICATION SYSTEMS**		7
	Introduction to communication	1	
	Communication process	1	
	Elements and types of Communication Systems	1	
	frequency ranges of communication	1	
	Modulation: Need for Modulation	1	
	Types of modulation: AM,FM,PM	1	
	Advantages, Disadvantages, Applications	1	
II	**UNIT-II: LINEAR CW MODULATION**		15
	Introduction, Mathematical Representation of AM	1	
	single tone modulation index and multi tone modulation index	1	
	Power Relationships and problem solving	2	
	AM signal generation -Square law modulator	1	
	Switching modulation	1	
	Demodulation (Envelop detector)	1	
	Virtues and Limitations of AM.	1	
	DSB-SC: Mathematical Representation of DSB-SC	1	
	DSB-SC generation (Ring modulation), Demodulation: Coherent detection	1	
	Filtering of AM Signals and Spectra, DSB signals and spectra.	1	
	SSB signal generation	1	
	demodulation using Hilbert transform	1	
	VSB Generation and demodulation	1	
	illustrative problems	1	
III	**UNIT-III: ANGLE CW MODULATION**		15
	Introduction, Types of Frequency Modulation (FM)	1	
	Mathematical representation of FM, Modulation index	1	
	Deviation sensitivity, Deviation ratio, Transmission bandwidth of FM (Carson's rule)	1	
	Narrow band FM, Wide band FM	1	
	generation of FM: Direct Method, Indirect Method	1	
	demodulation of FM, Voltage and Power for FM	1	

	Pre-emphasis and De-emphasis, Illustrative Problems.	2	
	Phase modulation and indirect FM	1	
	FM demodulators: Slope detector	1	
	Balanced slope discriminators	1	
	Phase difference discriminators, Ratio detector	1	
	PLL Detector	2	
	Assignment	1	
IV	**UNIT IV: TRANSMITTERS AND RECEIVERS**		9
	Introduction, AM Transmitters	1	
	Low level and high level transmitters	1	
	FM Transmitters	1	
	Armstrong FM transmitters	1	
	Receivers: TRF receivers	1	
	Super heterodyne receiver	2	
	Problem Solving	2	
V	**UNIT V: PULSE MODULATION TECHNIQUES**		9
	Definition, Types: PAM, PWM , PPM	1	
	Sampling,	1	
	Nyquist rate	1	
	Problems	1	
	Different sampling techniques	2	
	Generation and Detection of PAM	1	
	Generation and Detection of PWM	1	
	Generation and Detection of PPM	1	
VI	**UNIT VI: NOISE IN COMMUNICATION SYSTEMS**		15
	Introduction, Noise in Base band Systems	1	
	Noise figure	1	
	different types of noise	1	
	System Model and Parameter	1	
	SNR at the output of a Base band System	1	
	Noise in AM systems: System model and parameter	2	
	Noise in DSB and SSB Systems	1	
	Noise in Angle Modulation Systems: Output SNR in Angle Modulation	2	
	Threshold effects in Angle Modulation Systems	1	
	Improvement of SNR using Pre-emphasis and De-emphasis	1	
	problem solving	2	
	Assignment	1	

Signature of the H.O.D

D. Rajesh Sekar
Signature of the Faculty

C

RGM COLLEGE OF ENGINEERING & TECHNOLOGY

(AUTONOMOUS)

NANDYAL - 518 501. Kurnool (Dist.) A.P.

EDUCATION FOR PEACE

(ESTD-1995)

ATTENDANCE REGISTER

ACADEMIC YEAR	: 2022-23	SEM	: I / II
COURSE	:	CLASS :	III B.Tech
BRANCH / SECTION	: ECE / C		
SUBJECT & CREDITS	: AC & 3		
FACULTY	: D. Rajesh Setty		

R.G.M. COLLEGE OF ENGINEERING

ATTENDANCE

Class : III B.Tech

Branch : ECE - C

I Semister / Year II

Roll No.	Name	Date	1/8/22	2/8/22	2/8/22	3/8/22	4/8/22	6/8/22	8/8/22	10/8/22	11/8/22	13/8/22	16/8/22	16/8/22
			1	2	3	4	5	6	7	8	9	10	11	12
2009A0405	K Anok Kumar		A	A	A	A	A	A	A	1	2	3	4	5
407	N Aravind		A	A	A	1	2	3	4	A	5	6	7	8
415	M Bharath Kumar		1	2	3	4	5	6	7	8	9	A	A	A
416	K Bhargavi		A	A	A	A	1	2	3	4	5	6	7	8
422	G charan		A	A	A	A	A	A	1	2	3	4	5	6
423	Y chanitha		A	A	A	A	A	A	1	2	3	4	5	6
424	Y chetan		A	A	A	1	2	3	4	A	5	6	7	8
428	G Deva Kumar		A	A	A	A	1	2	3	4	5	6	7	8
433	M Dinesh Kumar		A	1	A	A	A	A	2	3	4	5	6	7
441	M Hareeshwar		A	1	A	A	2	A	3	4	5	6	A	7
442	S Hari Priya		A	A	A	A	A	1	2	3	4	5	6	7
448	B Hema Sai Pranathi		1	2	3	4	5	6	7	8	9	A	10	11
451	J Jagadeesh		A	A	A	1	A	A	2	3	4	5	6	7
453	S Jahnavi		A	A	A	A	1	2	3	4	5	6	7	8
457	A Joel Abhishekam		1	2	3	4	5	6	7	8	9	10	11	12
458	K Joel Peter singh		A	A	A	A	A	1	2	3	A	4	5	6
460	C Kalyan Babu		A	1	A	2	3	4	5	6	7	8	9	10
462	B Kalyan		A	A	A	A	A	A	A	A	A	1	2	3
463	C Kalyani		A	A	A	A	A	1	2	3	4	A	5	6
492	S Manju		A	A	A	A	A	1	2	3	4	5	6	7
493	K Meghana		A	A	A	A	A	A	A	A	A	A	A	A
4A0	K Naga Dhanalakshmi		A	A	A	1	2	3	A	4	A	A	5	6
4A4	K Narasimha		A	A	A	A	A	A	A	A	A	1	2	3
4A5	S Naveed Ahammed		A	A	A	A	A	A	A	A	A	A	A	A
4A7	M Niranjan		A	A	A	A	A	A	1	2	3	4	5	6
4A9	M Nithin Kumar		A	A	A	1	2	3	4	5	6	7	A	8
4B2	M Noor Mohammed		A	A	A	A	A	A	A	A	A	1	2	3
4B3	K om Prakash		A	A	A	1	2	3	4	A	5	6	7	8
4B4	P Pallavi		A	1	A	2	3	4	5	6	7	8	9	10
4B8	K Pavan Kumar		A	A	A	A	A	1	2	3	4	5	6	7
Signature of Teacher														
Signature of H.O.D.														
Signature of Principal														

& TECHNOLOGY (Autonomous), NANDYAL.

REGISTER Academic Year from...1-8-2022...to...6-12-22...Teacher...D. Rajeshsetty

Subject & Credits...AC - (C).....Dept...ECE - C.....

INTERNAL / RECORD MARKS

Roll No.	Assignments 1	2	Average (A)	Internal Test 1	2	Internal Marks 75% of MAX	25%of MIN	Total (I)	Final Internal (I+A)	Remarks
20N1A0405	10	10	10	10	05				19	
07	10	10	10	10	11				21	
15	10	10	10	11	06				20	
16	10	10	10	12	05				21	
22	10	10	10	13	09				22	
25	10	10	10	10	05				19	
26	10	10	10	14	07				23	
28	10	10	10	-9-	07				19	
33	10	10	10	11	04				20	
41	10	10	10	-9-	13				22	
42	10	10	10	12	08				21	
48	10	10	10	-5-	08				18	
51	10	10	10	10	08				20	
53	10	10	10	10	08				20	
57	10	10	10	10	09				20	
58	10	10	10	11	15				24	
60	10	10	10	-6-	11				20	
62	10	10	10	-6-	10				19	
63	10	10	10	-9-	11				21	
92	10	10	10	-9-	13				22	
93	-0-	-0-	-0-	A	A				00	
A0	10	10	10	-8-	05				18	
A4	10	10	10	-6-	13				22	
A5	10	10	10	12	16				25	
A7	10	10	10	10	11				21	
A9	10	10	10	-5-	03				15	
B2	10	10	10	-7-	08				18	
B3	10	10	10	-5-	03				15	
B4	10	10	10	-7-	14				23	
B8	10	10	10	-7-	10				20	

R.G.M. COLLEGE OF ENGINEERING

ATTENDANCE

Class : III B.Tech

Branch : ECE-C

I Semister / Year IV

Roll No.	Name	Date: 1/8/22	2/8/22	2/8/22	3/8/22	4/8/22	6/8/22	8/8/22	10/8/22	11/8/22	13/8/22	16/8/22	16/8/22
		1	2	3	4	5	6	7	8	9	10	11	12
2009A0													
4C2	P Prabhakar	A	A	A	A	A	1	2	3	A	4	5	6
4C5	M Praharshini	A	A	A	A	A	A	A	1	2	3	4	5
4D2	M Puja	A	A	A	1	2	3	A	4	5	A	6	7
4D6	Y Rajesh Kumar	A	A	A	A	A	1	2	3	4	5	6	7
4E3	G Ranjith	A	A	A	A	A	A	1	2	3	4	5	6
4G4	S Sameera	A	1	2	3	4	5	A	6	7	8	9	10
4G7	G Sandhya Rani	A	A	A	A	1	2	3	4	5	6	7	8
4H1	G Sathvik	A	A	A	A	A	1	2	3	4	5	6	7
4H4	G Shireesha	A	A	A	A	A	A	1	2	3	4	5	6
4H6	A Siva Ganesh	A	A	A	A	A	A	1	2	3	A	4	5
4J4	D Sreekanth Venkatesh	A	A	A	A	A	A	A	1	2	3	4	5
4J5	M Sreekanth	A	A	A	A	A	A	1	2	3	4	A	A
4J6	C Sreenath	A	A	A	A	1	2	3	A	A	4	5	6
4J7	V Sreenivasulu	A	A	A	A	1	2	3	4	5	6	7	8
4N5	M Varun	A	A	A	A	1	2	3	A	A	A	4	5
4P2	I Victor Babu	A	A	A	1	A	2	A	3	4	A	5	6
4P4	U Vijaya Lakshmi	A	A	A	A	A	A	1	2	3	4	A	A
4Q0	P Yaswanth Reddy	A	A	A	A	A	A	1	2	A	A	3	4
2109SA0 401	K Abhignan	1	2	3	4	5	6	7	8	A	9	10	11
402	M Balaji	A	A	1	A	A	2	3	4	5	A	6	7
403	P Chandra Mouli	A	A	A	A	A	A	A	A	1	2	3	4
405	G Harish Kumar	A	A	A	A	A	1	2	3	4	5	A	A
406	P Harshitha	A	A	A	A	A	1	2	3	A	A	4	5
407	S Hyder Ali	1	2	3	4	5	6	7	8	9	10	11	12
409	G Jayanth	A	A	A	A	1	2	3	4	5	A	6	7
418	B Nikhitha	A	A	A	A	A	A	A	A	1	2	3	4
421	M Shashi Priya	A	A	A	1	2	3	A	4	5	6	7	8
422	B Shireesha	A	A	A	1	A	2	3	4	5	6	7	8
428	Y Sunil	A	A	A	A	A	1	2	3	4	5	6	7
433	K Pranesh Vittal	A	A	A	A	1	2	A	3	4	5	6	7
Signature of Teacher													
Signature of H.O.D.													
Signature of Principal													

& TECHNOLOGY (Autonomous), NANDYAL.

REGISTER Academic Year from 1-8-2022 to 6-12-22 Teacher D. Rajesh Sethy

Subject & Credits AC (C) Dept ECE-C

INTERNAL / RECORD MARKS

Roll No.	Assignments 1	2	Average (A)	Internal Test 1	2	Internal Marks 75% of MAX	25%of MIN	Total (I)	Final Internal (I+A)	Remarks
C2	10	10	10	13	A				20	
C5	10	10	10	-6-	08				18	
D2	10	10	10	13	13				23	
D6	10	10	10	12	10				22	
E3	10	10	10	-3-	09				18	
G4	10	10	10	13	03				21	
G7	10	10	10	11	09				21	
H1	10	10	10	-2-	11				19	
H4	10	10	10	-7-	08				18	
H6	10	10	10	12	15				25	
J4	10	10	10	-8-	09				19	
J5	10	10	10	-6-	11				20	
J6	10	10	10	-9-	14				23	
J7	10	10	10	•12	12				22	
N5	10	10	10	10	11				21	
P2	10	10	10	-6-	10				19	
P4	10	10	10	12	12				22	
Q0	10	10	10	18	13				27	
21095A0401	10	10	10	17	11				26	
02	10	10	10	11	12				22	
03	10	10	10	13	12				23	
05	10	10	10	12	10				22	
06	10	10	10	10	15				24	
07	10	10	10	13	11				23	
09	10	10	10	11	14				24	
18	10	10	10	10	10				20	
21	10	10	10	11	12				22	
22	10	10	10	15	16				26	
28	10	10	10	10	03				19	
33	10	10	10	14	13				24	

R.G.M. COLLEGE OF ENGINEERING

ATTENDANCE

Class : III B.Tech

Branch : ECE-C

I Semister / Year III

Roll No. 19091A	Name	Date	1/8/22	2/8/22	2/8/22	3/8/22	4/8/22	6/8/22	8/8/22	10/8/22	11/8/22	13/8/22	16/8/22	16/8/22
			1	2	3	4	5	6	7	8	9	10	11	12
412	V Anvikumar		A	1	2	3	4	A	A	A	A	5	6	A
Signature of Teacher			DRS	DRS	DRS	DRS	DRS	DRS	DRS	DRS	DRS	DRS	DRS	DRS
Signature of II.O.D.														
Signature of Principal														

& TECHNOLOGY (Autonomous), NANDYAL.

REGISTER Academic Year from...1-8-2022...to...6-12-22...Teacher...D.Rajesh...

Subject & Credits...AC (3).....Dept...ECE-C.....

INTERNAL / RECORD MARKS

Roll No.	Assignments 1	Assignments 2	Assignments Average (A)	Internal Test 1	Internal Test 2	Internal Marks 75% of MAX	Internal Marks 25%of MIN	Internal Marks Total (I)	Final Internal (I+A)	Remarks
19071A0412	10	10	10	01	05				14	

LECTURE RECORD

Subject : AC
Total Exams 2
Credits : 3
Each for 2 hrs.
Internal Mid Exam Marks : 20
Total quizes —
Internal Quiz Marks : —
No. of Assignment 2

S.No.	Date	Topic Covered / Exercise Completed	Remarks
1	1-8-22	Introduction to Analog communication	
2	2-8-22	Unit I : Communication process	
3	2-8-22	elements of Communication system.	
4	3-8-22	Types of Communication systems	
5	4-8-22	frequency ranges, modulation	
6	6-8-22	Need for modulation	
7	8-8-22	Types of modulation: AM, FM, PM	
8	10-8-22	Advantages, disadvantages & applications	
9	11-8-22	Unit II. Introduction	
10	13-8-22	Mathematical representation of AM Single tone modulation index	
11	16-8-22	multitone modulation index	
12	16-8-22	power relationships	
13	17-8-22	Square law modulation	
14	18-8-22	Switching modulation	
15	20-8-22	demodulation (envelop detector) Virtues and limitations of AM	
16	20-8-22	Problems on AM	
17	22-8-22	Training	
18	23-8-22	"	
19	23-8-22	"	
20	24-8-22	"	
21	25-8-22	"	
22	27-8-22	"	
23	30-8-22	Mathematical representation of DSBSC DSBSC generation (ring modulation)	
24	1-9-22	demodulation (Coherent detection)	30/8/22
25	3-9-22	filtering of AM Signals & Spectra DSB signals and spectra	

LECTURE RECORD

S.No.	Date	Topic Covered / Exercise Completed	Remarks
26	5-9-22	SSBSC : filtering of sidebands	
27	5-9-22	SSB signal generation SSB demodulation	
28	6-9-22	VSB generation and demodulation	
29	6-9-22	Problems	
30	7-9-22	Unit III : Introduction, Types of freq-	
31	8-9-22	uency modulation, Mathematical representation of FM	
32	9-9-22	Modulation index, deviation sens- itivity, deviation ratio, Transmission BW of FM	
33	12-9-22	Narrow band FM, Wideband FM	
34	13-9-22	Generation of FM : direct method	
35	13-9-22	Indirect method	
36	15-9-22	Demodulation of FM, Voltage and power for FM	
37	19-9-22	Preemphasis and deemphasis	
38	20-9-22	Problems	
39	22-9-22	Phase modulation and Indirect FM	
40	24-9-22	FM demodulators : Slope detector	24/9/2022
41	8-10-22	Balanced slope discriminators	
42	8-10-22	"	
43	10-10-22	Mid answer scripts distributed.	
44	11-10-22	Phase difference discriminators	
45	11-10-22	"	
46	12-10-22	Ratio detector	
47	13-10-22	PLL detector.	
48	15-10-22	PLL detector	
49	18-10-22	PLL detector	
50	19-10-22	Unit 4 : Transmitters and Receivers AM Transmitters : Low level Transmitter	

LECTURE RECORD

S.No.	Date	Topic Covered / Exercise Completed	Remarks
51	20-10-22	FM Transmitters: Armstrong FM Transmitter	
52	21-10-22	Receivers: TRF receivers	
53	22-10-22	Superheterodyne receiver.	
54	22-10-22	Problems	
55	26-10-22	Problems	
56	25-10-22	Unit 5: Pulse modulation techniques Definitions	
57	26-10-22	Types: PAM, PWM, PPM	
58	27-10-22	Sampling	
59	29-10-22	Nyquist rate	
60	31-10-22	Instantaneous sampling	Hall e 31/10/22
61	1-11-22	Natural sampling	
62	2-11-22	Flat top sampling	
63	3-11-22	Problems	
64	5-11-22	PAM generation	
65	7-11-22	PAM detection	
66	8-11-22	PWM generation	
67	8-11-22	PWM detection	
68	10-11-22	PPM generation	
69	10-11-22	PPM detection	
70	12-11-22	Unit 6: Noise in Communication systems: Introduction	
71	15-11-22	Noise in Base band systems	
72	15-11-22	Noise figure, different types of noise, system model and parameter	
73	16-11-22	SNR at the o/p of base band system	
74	17-11-22	Noise in AM systems: system model & parameter	
75	19-11-22	Noise in DSB&SSB systems	
76	21-11-22	Noise in angle modulation systems: o/p SNR in angle modulation.	
77	22-11-22	Threshold effects in angle modulation systems,	
78	23-11-22	Improvement of SNR using pre-emphasis and de emphasis.	
79 to 86	24-11-22 to 3-12-22	Training	Hall e 6/12/22
87	5-12-22	Revision	
88	6-12-22	Revision	
89	6-12-22	Revision	

UNIT I

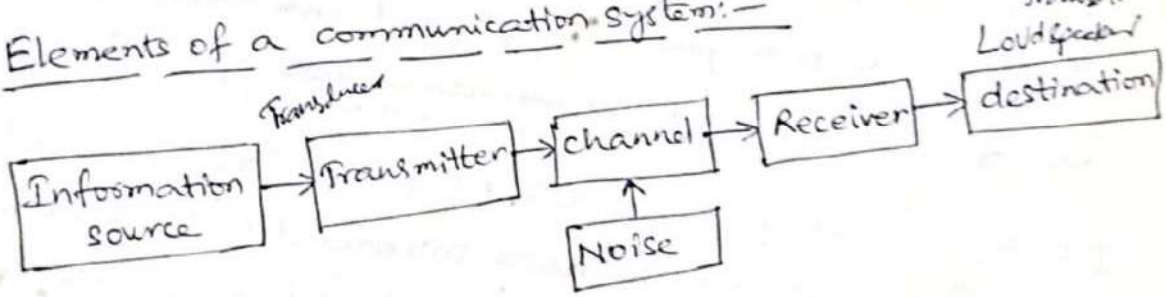
Introduction to communication system:-

- Communication is the process of exchanging information.
- The electronic equipments which are used for communication purpose are called communication equipments.
- Different communication equipments when assembled together form a communication system.

→ Examples of communication systems:

line telegraphy
line telephony
radio broadcasting
television broadcasting
radar communication
computer communication
mobile communication. etc;

Elements of a communication system:-



Information source: The communication system exists to communicate a message. This message or information originates from information source.

- In general there can be various messages in the form of words, groups of words, codes, symbols, sound signals etc; The desired message is selected and conveyed.
- The fn of inform. source is to produce required message which is to be transmitted.

(*) When a message is not electrical in nature, we have to convert it into an electrical signal, ∵ all signal processing devices are electrical in nature. Hence a transducer is required after the source.

Transmitter:- The function of the transmitter is to process the electrical signal from different aspects.

- In radio broadcasting, electrical signal obtained from sound signal is processed to restrict its range of audio freqs (upto 5KHz in AM) and is often amplified.
- In wire telephony, no real processing is needed.
- In long distance radio comm. amplification is necessary before modulation.
- Modulation is the main function of tx^r.
- In modn message signal is superimposed upon the HF carrier signal.
- Hence we can say inside the tx^r, signal processing such as restriction of AF range, amplification, modn are achieved.

Channel:- The term channel means the medium through which the message travels from tx^r to rx^r.

- The fn of ch. is to provide a physical connection b/n tx^r & rx^r.
- exs of chs are wired lines, microwave links, optical fibers. (EM wave in free space)

Noise:- During the process of tx^n and rx^n the signal gets disturbed due to noise introduced in the system.

- Noise is an unwanted signal which tends to interfere with the required signal. Noise signal is always random in char. Noise may interfere with signal at any point in a comm. sys. It will have its greatest effect when the signal is weakest.

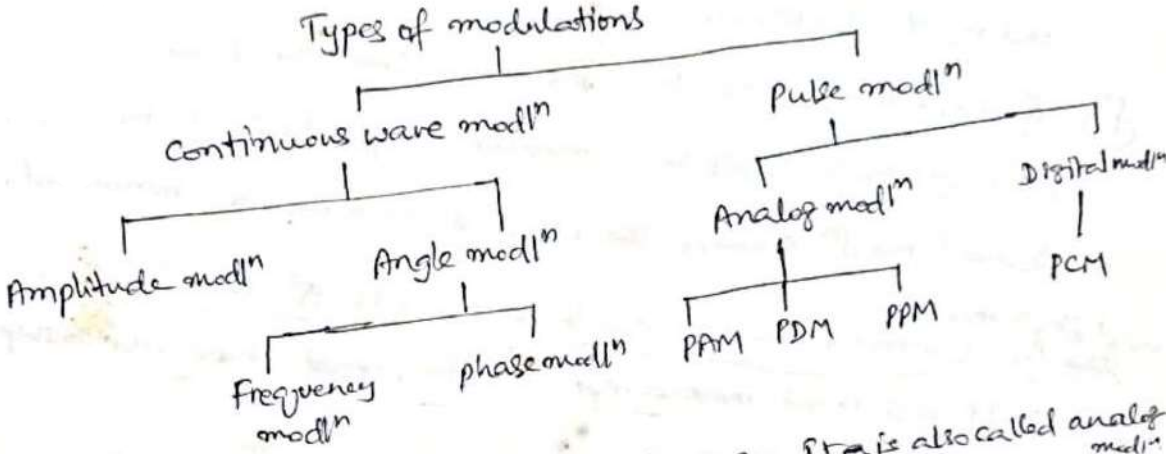
Receiver:- The fn of rx^r is to reproduce the message signal in electrical form from the distorted rxd signal. Reproduction of original signal is done by demodulation. It is the reverse process of modulation.

Destination:- It is the final stage which is used to convert an electrical message in to its original form. In radio broadcasting the destination is a loud speaker. (electrical → sound signal)

Modulation:-

(*) Modulation is defined as the process in which some characteristic of the carrier is varied in accordance with the instantaneous value of modulating signal.

- Signals containing information or intelligence are referred to as modulating signal. (message or base band signal).
- carrier frequency is greater than the modulating frequency.
- The signal resulting from the process of modulation is called modulated signal.



CW modⁿ:- carrier wave is continuous. Eta is also called analog modⁿ

Pulse modⁿ:- carrier is pulse wf

Need for modulation:-

(Benefits or adv. of modⁿ.)

(*) 1st statement:- modg signal (message) can't be txd itself.

(i) Practicality of antenna:- when free space is used as a txd medium (channel) messages are transmitted and received with the help of antennas.

- For efficient radiation and reception the txd and rxd ants must have heights comparable to a quarter wavelength of the freq used
- In the broadcast band for $f = 1MHz$; $\frac{\lambda}{4} = \frac{c}{4f} = \frac{3\times10^8}{4\times10^6} = 75m$

but at ~~75~~ 15 KHz; $\frac{\lambda}{4} = \frac{3\times10^8}{4\times15\times10^3} = 5Km.$

- A vertical antenna of this height is impracticable

(ii) To remove interference:

Another reason for not radiating modulating signal itself is that all signal is concentrated within the range from 20Hz to 20kHz, so that all signals from different sources would be mixed up and distortion will exist)

- In order to seperate the various signals, it is necessary to translate (shift) them all to different portions of the electromagnetic spectrum. each must be given its own bw (ch.BW)
- This also overcomes the difficulties of poor radiation at LFs.
- Once the signals have been translated, a tuned ckt is employed in the front end of the receiver to select desired section of spectrum and reject all others.

(iii) Reduction of noise: Noise is the major limitation of any communn. Although noise can't be eliminated completely; with the help of several modn schemes, the effect of noise can be minimized.

2nd Statement

An unmodd carrier can't be used to convey information (txt intelligence).

(4) ∵ It has const max amp, const freq, const phase relationship w.r.t some reference

(3) • A message consists of varying quantities (ex: speech)

Speech is made up of rapid and unpredictable variations in amplitude (volume) and frequency (pitch).

(3) • ∵ it is impossible to rep. these two variables by a set of three constant parameters, an unmodd carrier can't be used to convey information.

Voice signal can't travel longer distances in air, the signal gets attenuated rapidly.

Unit 2 Linear CW Modulation

Amplitude Modulation:- It is defined as a system of modulation in which the amplitude of the carrier is made proportional to the instantaneous amplitude of the modulating voltage.

Single tone AM:-

Let the carrier voltage is $c(t) = v_c = V_c \cos \omega_c t$

" " modulating voltage is $m(t) = v_m = V_m \cos \omega_m t$

When the carrier is amplitude modulated the instantaneous modulating variations are superimposed on to carrier amplitude.

When modulation is present the amplitude of the carrier is varied by by its instantaneous value. (See fig)

modulation index $m = \frac{V_m}{V_c}$; $m = 0$ to 1

It can be expressed as a % and is called % modln.

$V_m > V_c \Rightarrow$ distortion occurs.

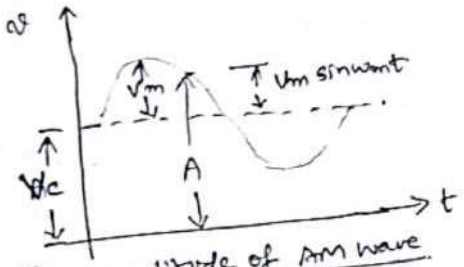


fig. amplitude of AM wave

Amplitude of the amplitude modulated voltage is

$$A = V_c + v_m = V_c + V_m \cos \omega_m t = V_c \left(1 + \frac{V_m}{V_c} \cos \omega_m t\right)$$

$$= V_c (1 + m \cos \omega_m t)$$

Instantaneous voltage of resulting AM wave is

$$s(t) = v = A \cos \omega_c t.$$

$$v = V_c (1 + m \cos \omega_m t) \cos \omega_c t$$

$$= V_c \cos \omega_c t + m V_c \cos \omega_m t \cos \omega_c t$$

we know that $2\cos A \cos B = \cos(A+B) + \cos(A-B)$

$\Rightarrow s(t) = v = V_c \cos\omega_c t + \frac{mV_c}{2}\cos(\omega_c+\omega_m)t + \frac{mV_c}{2}\cos(\omega_c-\omega_m)t$

Time domain representation

① Unmodulated carrier ② Upper Sideband ③ Lower Sideband.

$\Rightarrow s(t) = V_c \cos 2\pi f_c t + \frac{mV_c}{2}\cos 2\pi(f_c+f_m)t + \frac{mV_c}{2}\cos 2\pi(f_c-f_m)t.$

freq. of USB $= f_c + f_m$

LSB $= f_c - f_m$

Bandwidth required for AM $= f_{USB} - f_{LSB} = (f_c+f_m) - (f_c-f_m)$

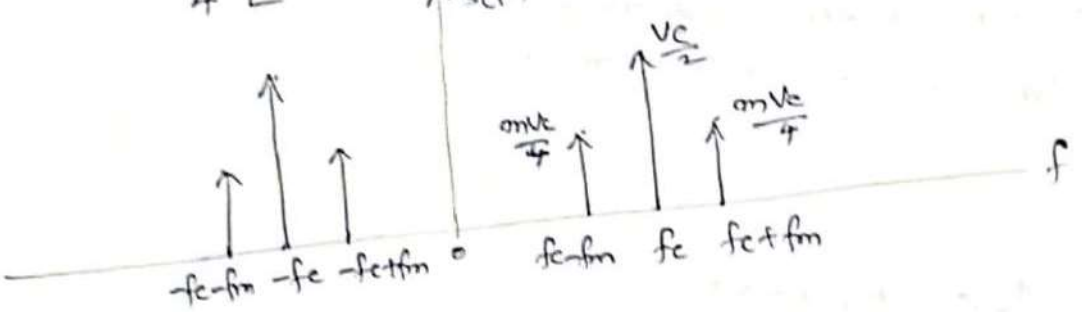
$= 2f_m.$

Freq. domain representation:-

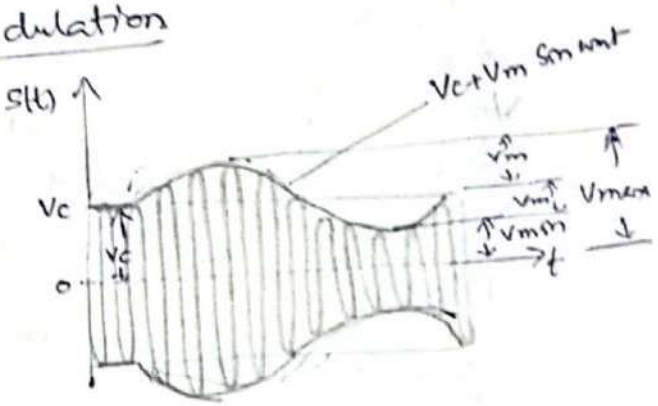
Take F.T. of $s(t)$.

$$S(f) = \frac{V_c}{2}\left[\delta(f-f_c)+\delta(f+f_c)\right] + \frac{mV_c}{4}\left[\delta\{f-(f_c+f_m)\}+\delta\{f+(f_c+f_m)\}\right]$$

$$+ \frac{mV_c}{4}\left[\delta\{f-(f_c-f_m)\} + \delta\{f+(f_c-f_m)\}\right.$$



Modulation index (or) depth of modulation



$V_{max} - V_{min} = 2V_m$

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

$V_c = V_{max} - V_m = V_{max} - \frac{(V_{max}-V_{min})}{2}$

$= \left(V_{max} - \frac{V_{max}}{2}\right) + \frac{V_{min}}{2}$

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

$m = \frac{V_m}{V_c} = \frac{V_{max}-V_{min}}{2} \div \frac{V_{max}+V_{min}}{2} = \frac{V_{max}-V_{min}}{V_{max}+V_{min}}.$

Power relations in AM wave.

AMd wave contains 2 sideband components in addition to carrier.

∴ The modd wave contains more power than unmodd carrier power.

Total Power in the modd wave $P_t = P_c + P_{USB} + P_{LSB}$.

$$P_t = \frac{V_{carr}^2}{R} + \frac{V_{USB}^2}{R} + \frac{V_{LSB}^2}{R}.$$

where all three voltages are rms values.

R = antenna resistance.

$$P_t = \frac{(V_c/\sqrt{2})^2}{R} + \left(\frac{mV_c/2}{\sqrt{2}}\right)^2 \div R + \left(\frac{mV_c/2}{\sqrt{2}}\right)^2 \div R$$

$$= \frac{V_c^2}{2R} + \frac{m^2V_c^2}{8R} + \frac{m^2V_c^2}{8R}$$

$$= \frac{V_c^2}{2R}\left(1 + \frac{m^2}{4} + \frac{m^2}{4}\right) = \frac{V_c^2}{2R}\left(1 + \frac{m^2}{2}\right) = \frac{(V_c/\sqrt{2})^2}{R}\left(1 + \frac{m^2}{2}\right)$$

$$= \frac{V_{carr}^2}{R}\left(1 + \frac{m^2}{2}\right)$$

$$P_t = P_c\left(1 + \frac{m^2}{2}\right)$$

$$P_t = P_c + \frac{m^2}{2}P_c = P_c + \frac{m^2}{4}P_c + \frac{m^2}{4}P_c$$

$$P_t = P_c + P_{SB} = P_c + P_{USB} + P_{LSB}$$

$$\frac{P_t}{P_c} = \left(1 + \frac{m^2}{2}\right)$$

Current calculations :-

I_c = unmodulated current (rms)

I_t = total or modulated current of an AM Tx. (rms).

R = resistance in which these currents flow, then.

$$\frac{P_t}{P_c} = \frac{I_t^2 R}{I_c^2 R} = \left(\frac{I_t}{I_c}\right)^2 = 1 + \frac{m^2}{2}$$

$$\frac{I_t}{I_c} = \sqrt{1 + \frac{m^2}{2}} \quad \text{or} \quad I_t = I_c \sqrt{1 + \frac{m^2}{2}}$$

(Pb) The antenna current of an AM transmitter is 8 Amps. when only the carrier is sent, but increases to 8.93 A when the carrier is modulated by a single sine wave. find the percentage modulation. Determine the antenna current when the depth of modulation changes to 0.8.

$I_c = 8A$, $I_t = 8.93A$ $\%\ m = ?$

$$m = \sqrt{2\left[\left(\frac{I_t}{I_c}\right)^2 - 1\right]} = 0.701$$

$\%\ m = 70.1$

$m = 0.8$ $I_t = ?$ $I_c = 8$

$$I_t = I_c \sqrt{1 + \frac{m^2}{2}} = 9.19A$$

Modulation by several sine waves :-

1. Let V_1, V_2, V_3 etc; be the simultaneous modulating voltages. Then the total modulating voltage $V_t = \sqrt{V_1^2 + V_2^2 + V_3^2 + \cdots}$

$$\frac{V_t}{V_c} = \sqrt{\frac{V_1^2}{V_c^2} + \frac{V_2^2}{V_c^2} + \frac{V_3^2}{V_c^2} + \cdots}$$

$$\Rightarrow m_t = \sqrt{m_1^2 + m_2^2 + m_3^2 + \cdots}$$

2. Total power in the AM wave consists of carrier power and sideband power. $P_t = P_c\left(1 + \frac{m^2}{2}\right) = P_c + \frac{P_c m^2}{2} = P_c + P_{SB}$.

P_{SB} is Total sideband power $= \frac{P_c m^2}{2}$.

If several sinewaves simultaneously modulate the carrier, The carrier power will be unaffected, but the total sideband power will now be the sum of the individual sideband powers.

$$\therefore P_{SB_T} = P_{SB_1} + P_{SB_2} + P_{SB_3} + \cdots$$

$$\Rightarrow \frac{P_c m_t^2}{2} = \frac{P_c m_1^2}{2} + \frac{P_c m_2^2}{2} + \frac{P_c m_3^2}{2} + \cdots$$

$$m_t^2 = m_1^2 + m_2^2 + m_3^2 + \cdots$$

$$\Rightarrow m_t = \sqrt{m_1^2 + m_2^2 + m_3^2 + \cdots}$$

(Pb) A certain transmitter radiates 9 kW with the carrier unmodd, and 10.125 kW when the carrier is sinusoidally modulated. Calculate the modulation index. If another sinewave corresponding to 40% modulation, is tx'd simultaneously, determine the total radiated power.

(i) $m = \sqrt{2\left(\frac{P_t}{P_c} - 1\right)} = \sqrt{2\left(\frac{10.125}{9} - 1\right)} = 0.5$

(ii) $m_1 = 0.5$; $m_2 = 0.4$ $\quad P_t = P_c\left(1 + \frac{m_t^2}{2}\right)$; $m_t = \sqrt{m_1^2 + m_2^2}$

$$m_t = \sqrt{0.5^2 + 0.4^2} = 0.64$$

$$P_t = 9k\left(1 + \frac{0.64^2}{2}\right) = 10.84 \text{ kW}.$$

(Pb) The antenna current of an AM broadcast transmitter, modulated to a depth of 40% by an audio ~~to~~ sine wave, is 11A. It increases to 12A as a result of simultaneous ~~radiation~~ modulation by another audio sinewave. what is the modln index due to this 2nd wave?

$I_t = 11A \quad m = 0.4 \quad I_t = I_c\sqrt{1 + \frac{m^2}{2}}$

$$I_c = \frac{I_t}{\sqrt{1 + \frac{m^2}{2}}} = \frac{11}{\sqrt{1 + \frac{0.4^2}{2}}} = 10.58A$$

$$m_t = \sqrt{2\left[\left(\frac{I_t}{I_c}\right)^2 - 1\right]} = \sqrt{2\left[\left(\frac{12}{10.58}\right)^2 - 1\right]} = 0.757$$

$$m_2 = \sqrt{m_t^2 - m_1^2} = \sqrt{0.757^2 - 0.4^2} = 0.643.$$

Generation of AM waves:

(i) Square law modulator (ii) switching modulator

Non linear elements are used in these modulators.

These two devices are well suited for low power modulation purposes.

Square-Law Modulator:-

A square law modulator requires three features

(i) a means of summing the carrier and modulating waves

(ii) a non linear element and

(iii) a Bandpass filter for extracting the desired modulation product

Semiconductor diodes and transistors are the most common non linear devices used for implementing square law modulators. The filtering requirement is usually satisfied by using a single or double tuned filter.

When a non linear element such as a diode is suitably biased and operated in a restricted portion of its char. curve, i.e; the signal applied to the diode is relatively weak, we find that the transfer char. of the diode-load resistor combination can be represented closely by a square law:

$$v_2(t) = a_1 v_1(t) + a_2 v_1^2(t) \quad -(1)$$

where a_1 and a_2 are constants.

The i/p voltage $v_1(t)$ consists of the carrier wave plus the modulating wave, that is

$$v_1(t) = A_c \cos 2\pi f_c t + m(t) \quad -(2)$$

$\therefore$ Substituting (2) in (1),

$$v_2(t) = a_1 A_c \cos 2\pi f_c t + a_1 m(t) + a_2 A_c^2 \cos^2 2\pi f_c t + a_2 m^2(t) + 2 a_2 m(t) A_c \cos 2\pi f_c t$$

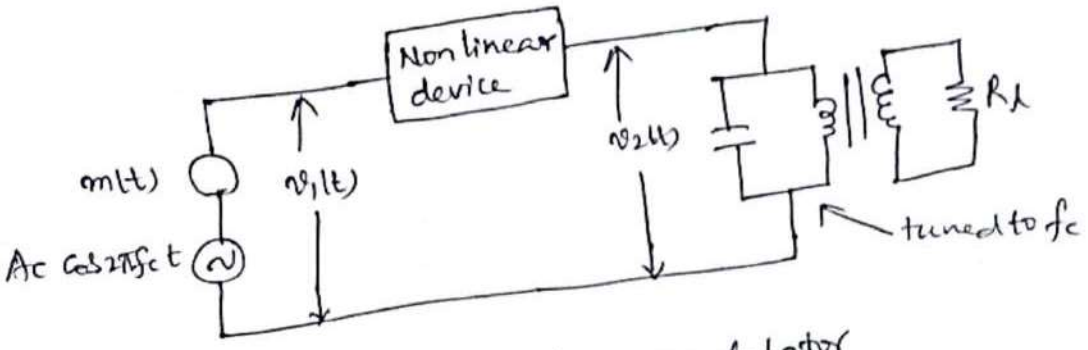


fig. Square law modulator

$$a_2(t) = a_1 A_c \cos 2\pi f_c t \left(1 + 2\frac{a_2}{a_1} m(t)\right)$$

$$+ a_1 m(t) + a_2 m^2(t) + a_2 A_c^2 \cos^2(2\pi f_c t)$$

The resulting voltage developed across the primary winding of the output transformer is given by

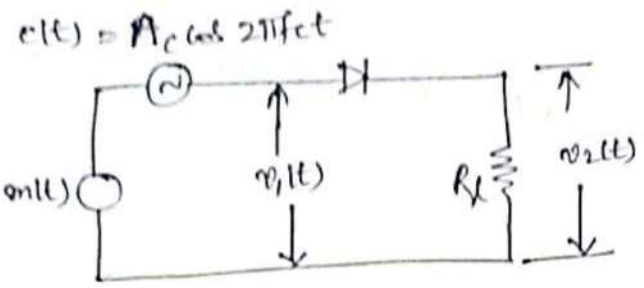
$$a_2(t) = \underbrace{a_1 A_c \left[1 + \frac{2a_2}{a_1} m(t)\right] \cos 2\pi f_c t}_{\text{AM wave}} + \underbrace{a_1 m(t) + a_2 m^2(t) + a_2 A_c^2 \cos^2(2\pi f_c t)}_{\text{unwanted terms.}}$$

The first term is the desired AM wave with amplitude sensitivity $k_a = 2a_2/a_1$.

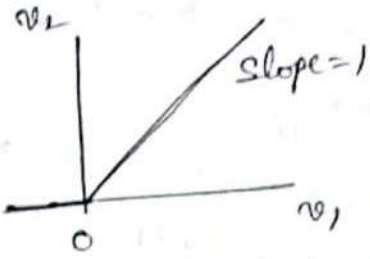
The remaining three terms are unwanted terms; they are removed by appropriate filtering.

Switching Modulator:-

- It is assumed that the carrier wave c(t) applied to the diode is large in amplitude, so that it swings right across the char curve of the diode.
- We assume that the diode acts as an ideal switch, that is, it presents zero impedance when it is forward biased [corresponding to c(t)>0] and infinite impedance when it is reverse biased [corresponding to c(t)<0].



(a) ckt diagram



(b) idealized i/p-o/p relation

fig: switching modulator

- we may thus approximate the transfer char. of the diode load resistor combination by a piecewise linear characteristic, as shown in fig (b).
- Accordingly for an i/p voltage $v_1(t)$ given by $v_1(t) = A_c \cos 2\pi f_c t + m(t)$, where $|m(t)| << A_c$, the resulting load voltage $v_2(t)$ is

$$v_2(t) \cong v_1(t) \quad ; c(t) > 0$$
$$= 0 \quad ; c(t) < 0$$

- i.e; the load voltage $v_2(t)$ varies periodically between the values $v_1(t)$ and zero at a rate equal to f_c.
- In this way, by assuming a modg wave that is weak compared with the carrier wave, we have effectively replaced the non linear behaviour of the diode by an approximately eqt linear time varying operation.
- ∴ The above eqn may be expressed as

$$v_2(t) \cong [A_c \cos 2\pi f_c t + m(t)]\, g_p(t)$$

- where $g_p(t)$ is a periodic pulse train of duty cycle equal to one half and period $T_0 = 1/f_c$ as shown in fig below.

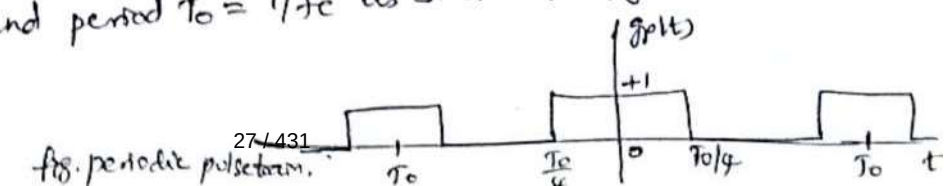


fig. periodic pulse train.

representing this $g_p(t)$ by its fourier series, we have

$$g_p(t) = \frac{1}{2} + \frac{2}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos\left[2\pi f_c t (2n-1)\right]$$

$$= \frac{1}{2} + \frac{2}{\pi} \cos 2\pi f_c t + \text{odd harmonic components.}$$

$\therefore$ Substituting $g_p(t)$ in $v_2(t)$

$$v_2(t) = \left[A_c \cos 2\pi f_c t + m(t)\right]\left[\frac{1}{2} + \frac{2}{\pi} \cos 2\pi f_c t + \text{odd harmonic components}\right]$$

$$= \frac{A_c}{2}\left[1 + \frac{4}{\pi A_c} m(t)\right] \cos 2\pi f_c t + \text{unwanted terms.}$$

The first term is the desired AM wave with amplitude sensitivity $k_a = \frac{4}{\pi A_c}$.

The unwanted terms are removed from the load voltage $v_2(t)$ by means of a BPF.

Detection of AM waves:-

- The process of detection or demodulation provides a means of recovering the message signal from an incoming modulated wave. Detection is the inverse process of modulation.

(i) Square law detector (ii) envelope detector

Square law detector:-

- It is obtained by using a square law modulator for the purpose of detection.
- The transfer characteristic of a nonlinear device is given by

$v_2(t) = a_1 v_1(t) + a_2 v_1^2(t)$

where $v_1(t)$ and $v_2(t)$ are the input and output voltages, respectively, and a_1 and a_2 are constants.

- When such a device is used for the demodulation of an AM wave,
- we have for the input

$$v_1(t) = A_c[1 + K_a m(t)] \cos 2\pi f_c t.$$

$$\therefore v_2(t) = a_1 A_c[1 + K_a m(t)] \cos 2\pi f_c t + a_2 A_c^2 [1 + K_a m(t)]^2 \cos^2 2\pi f_c t.$$

$$= a_1 A_c [1 + K_a m(t)] \cos 2\pi f_c t + \frac{1}{2} a_2 A_c^2 [1 + 2K_a m(t) + K_a^2 m^2(t)] [1 + \cos 4\pi f_c t]$$

- The desired signal namely $a_2 A_c^2 K_a m(t)$, is due to $a_2 v_1^2(t)$ term — hence the description "square law detector".
- This component can be extracted by means of a low pass filter.
- This is not the only contribution within the baseband spectrum, however, because the term $\frac{1}{2} a_2 A_c^2 K_a^2 m^2(t)$ will give rise to a plurality of similar freq components.
- The ratio of wanted signal to distortion is equal to $2/K_a m(t)$.
- To make this ratio large we limit the % modulation, i.e; we choose $|K_a m(t)|$ small compared with unity for all t.
- We conclude ∴ that distortionless recovery of the baseband signal $m(t)$ is possible only if the applied AM wave is weak (so as to justify the use of a square law i/p–o/p relation) and if the % modulation is very small.

Envelope detector:-

- An envelope detector is a simple and highly efficient device that is well suited for the demodulation of a narrowband AM wave. (i.e; f_c is large compared with W) for which the % modn is less than 100.
- Ideally an envelope detector produces an o/p signal that follows the envelope of the i/p signal waveform exactly; hence the name.
- Some version of this ckt is used in almost all commercial AM radio receivers.
- Fig. shows the circuit diagram of an envelope detector.
- It consists of a diode and a resistor-capacitor filter.
- on the +ve half cycle of the input signal, the diode is FBed and the capacitor C charges up rapidly to the peak value of the i/p signal.
- when the i/p signal falls below this value the diode becomes RBed and the capacitor C discharges slowly through the load res R_L.
- The discharging process continues until the next positive half cycle.
- when the i/p signal becomes greater than the voltage across the capacitor, the diode conducts again and the process is repeated.
- we assume that the diode is ideal, presenting zero impedance to current flow in the FBed region, and infinite impedance in the reverse biased region.
- we further assume that the AM wave applied to the envelope detector is supplied by a voltage source of internal impedance R_s.
- The charging time constant $R_s C$ must be short compared with the carrier period $\frac{1}{f_c}$. i.e; $R_s C \leq \frac{1}{f_c}$.
- Hence the capacitor C charges rapidly and thereby follows the applied voltage up to the positive peak when the diode is conducting.
- on the other hand the discharging time constant $R_L C$ must be long enough to ensure that the capacitor discharges slowly through the load resistor R_L between +ve peaks of the carrier wave, but not so long that the cap voltage will not discharge at the max rate of change of the modulating wave, i.e; $\frac{1}{f_c} \ll R_L C \ll \frac{1}{W}$

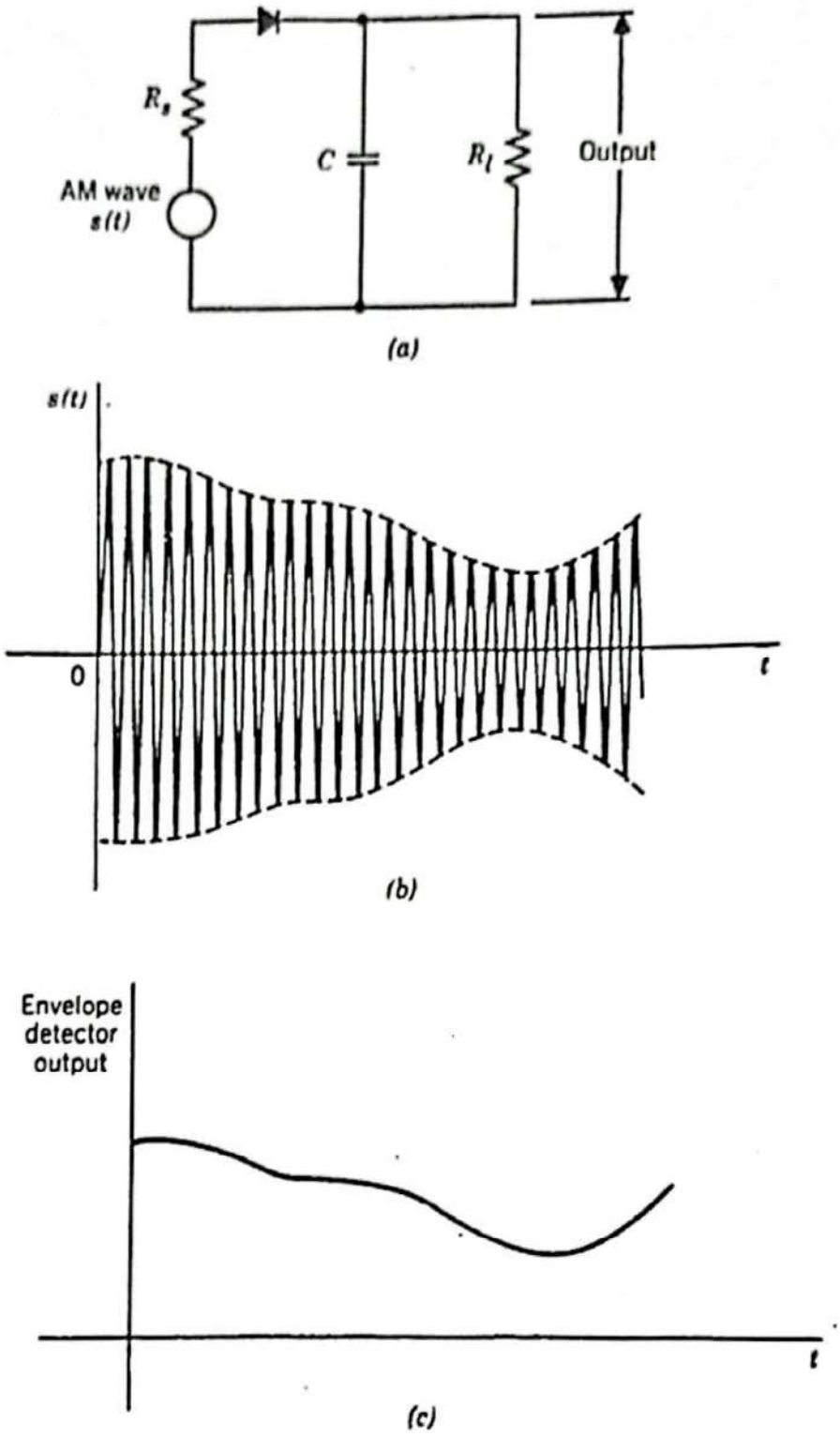


Figure
Envelope detector. (a) *Circuit diagram.* (b) *AM wave input.* (c) *Envelope detector output.*

- where W is the message bandwidth.
- The result is that the capacitor voltage or detector output is very nearly the same as the envelope of the AM wave.
- The detector o/p usually has a small ripple at the carrier freq fc, this ripple is easily removed by low pass filtering.

DSB MODULATION

Double side band suppressed carrier (DSB-SC) modulation:-

(*) The equation of AM wave is given by

$$S(t) = V_c \cos\omega_c t + \frac{mV_c}{2}\cos(\omega_c+\omega_m)t + \frac{mV_c}{2}\cos(\omega_c-\omega_m)t$$

(*) From this equation, it is obvious that the carrier component in AM wave remains constant in amplitude and frequency.

(*) The carrier of the amplitude modulated wave doesnot contain any information.

(*) In power calculations of AM signal,

$$P_t = P_{carrier} + P_{USB} + P_{LSB}$$

$$P_t = \frac{V_c^2}{2R} + \frac{m^2}{4}\left(\frac{V_c^2}{2R}\right) + \frac{m^2}{4}\left(\frac{V_c^2}{2R}\right)$$

$$P_t = \frac{V_c^2}{2R}\left(1 + \frac{m^2}{4} + \frac{m^2}{4}\right)$$

$$\boxed{P_t = P_c\left(1+\frac{m^2}{2}\right)}$$

(*) If the carrier is suppresed, then the total power transmitted in DSB-SC is the sum of the sidebands power.

$$P_t' = P_{SB} = P_{LSB} + P_{USB}$$

$$\boxed{P_t' = \frac{m^2}{2}\cdot P_c}$$

$$\% \text{ power saving} = \left(\frac{P_t - P_t'}{P_t}\right)\times 100 \quad \Leftarrow \quad \frac{P_t - P_{SB}}{P_t}\times 100$$

$$= \left[\frac{P_c\left(1+\frac{m^2}{2}\right) - \frac{m^2}{2}\cdot P_c}{P_c\left(1+\frac{m^2}{2}\right)}\right]\times 100$$

$$= \left[\frac{P_c\left(1+\frac{m^2}{2} - \frac{m^2}{2}\right)}{P_c\left(1+\frac{m^2}{2}\right)}\right]\times 100$$

$$= \left(\frac{2}{2+m^2}\right)\times 100$$

$$\boxed{\% \text{ power saving} = \frac{200}{2+m^2}}$$

(*) if $m=1$ ie., for 100% modulation,

$$\% \text{ power saving} = \frac{200}{2+1} = \frac{200}{3} = 66.667\%$$

→ Therefore, for 100% modulation, about 66.67% of the total power is required for transmitting the carrier which

doesnot contain any information.

(*) If the carrier is suppressed, only the sidebands power will remains and in this way, the saving of two-third (2/3) of power may be achieved at 100% modulation.

(*) Suppression of the carrier doesnot affect the baseband signal in any way. The resulting signal obtained by suppressing carrier for a modulated wave is called as double sideband suppressed carrier signal.

(*) Thus, in DSB-SC modulation, there is no carrier signal, only sideband signals are present.

Time domain description of DSB-SC :—

Let $v_m = V_m \cos\omega_m t$ and $v_c = V_c \cos\omega_c t$

modulation index (m) $= \frac{V_m}{V_c}$

$s(t) = V_c (1 + m\cos\omega_m t) \cos\omega_c t$

$s(t) = (V_c + mV_c \cos\omega_m t) \cos\omega_c t$

$s(t) = (V_c + V_m \cos\omega_m t) \cos\omega_c t \quad (\because m = \frac{V_m}{V_c})$.

$s(t) = (V_c + v_m) \cos\omega_c t$

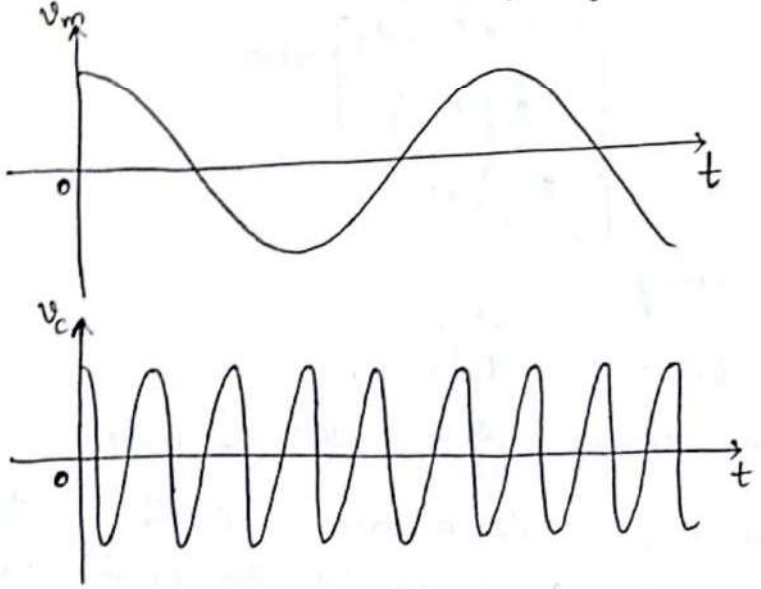
$s(t) = V_c \cos\omega_c t + v_m \cos\omega_c t$

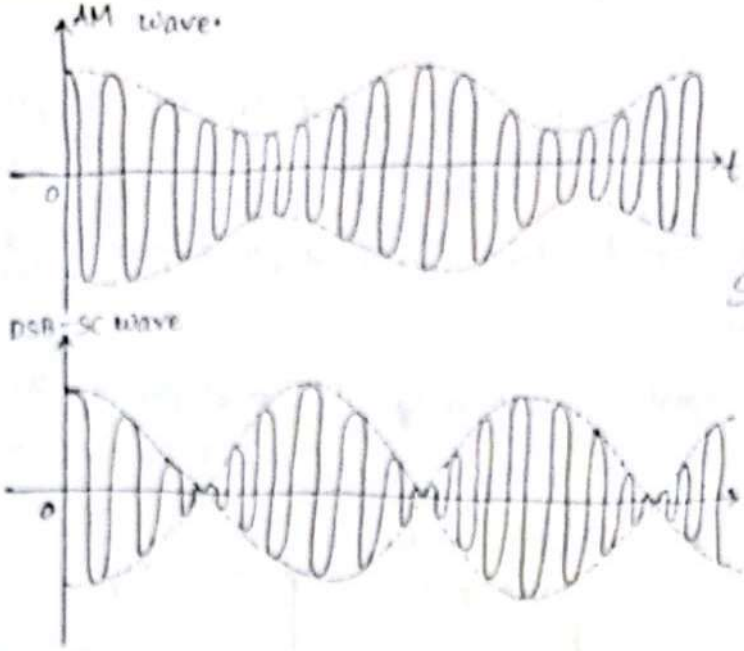
$s(t)$ for DSB-SC = $s(t)$ of AM wave $- V_c \cos\omega_c t$

$= v_m \cos\omega_c t$

$= x(t) \cos\omega_c t$

where $x(t)$ = message signal.





$s(t) = c(t)\, m(t)$
$= A_c \cos 2\pi f_c t\, m(t)$

The phase reversal occurs in DSB-SC when there is a zero crossing of the message signal.

Frequency domain description of DSB–SC:–

Let $x(t) = v_m = V_m \cos \omega_m t$

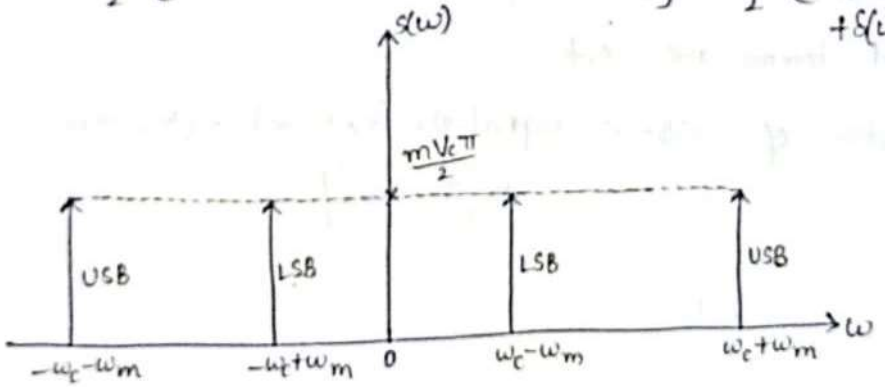
$x(\omega) = V_m \left[\pi \left(\delta(\omega-\omega_m) + \delta(\omega+\omega_m)\right)\right]$

$c(t) = v_c = V_c \cos \omega_c t$

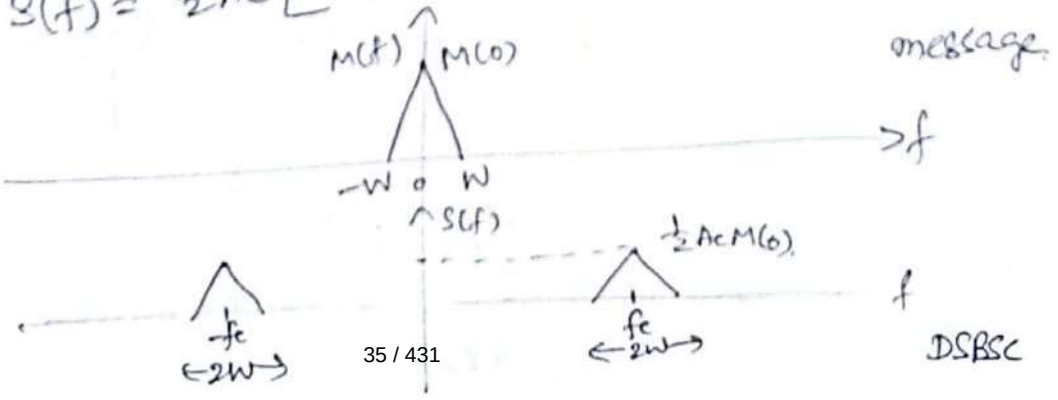
$c(\omega) = V_c \left[\pi(\delta(\omega-\omega_c) + \delta(\omega+\omega_c))\right]$

$s(t) = \frac{mV_c}{2} \cos(\omega_c+\omega_m)t + \frac{mV_c}{2} \cos(\omega_c-\omega_m)t$

$s(\omega) = \frac{mV_c}{2} \left[\pi(\delta(\omega-\omega_c-\omega_m) + \delta(\omega+\omega_c+\omega_m))\right] + \frac{mV_c}{2} \left[\pi\left(\delta(\omega-\omega_c+\omega_m) + \delta(\omega+\omega_c-\omega_m)\right)\right]$



$S(f) = \frac{1}{2} A_c \left[M(f-f_c) + M(f+f_c)\right]$

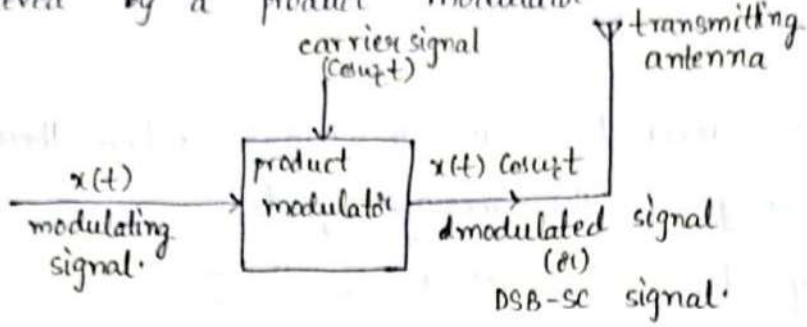


Product Modulator:—

The DSB-SC signal is represented by $x(t)\cos\omega_c t$.

(*) If $x(t)$ is taken as the modulating signal and $\cos\omega_c t$ is taken as carrier signal, then $x(t)\cos\omega_c t$ represents the modulated signal.

(*) The DSB-SC signal is obtained by multiplying the modulating signal $[x(t)]$ with carrier signal $[\cos\omega_c t]$. This is achieved by a product modulator.



Features of DSB-SC signal:—

(1) DSB-SC signal exhibits phase reversal at zero-crossing ie., whenever the modulating signal $[x(t)]$ crosses zero.

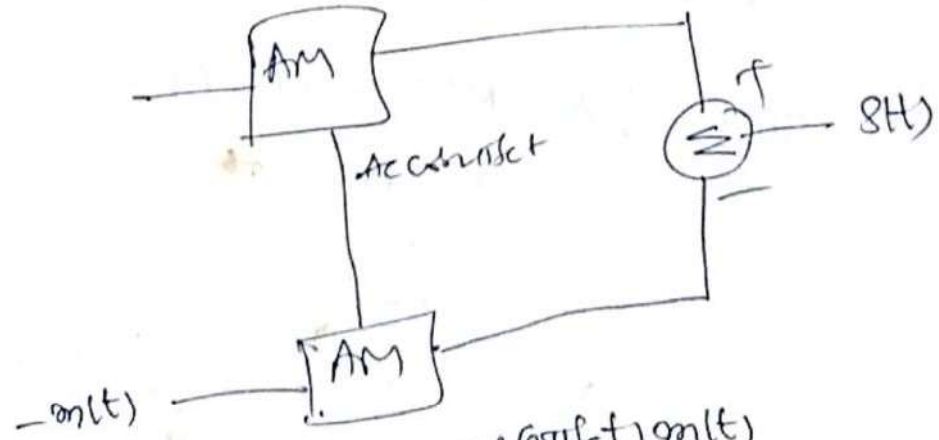
(2) Because of this, the envelope of DSB-SC modulated signal is different from the message signal.

(3) The impulses at $\pm\omega$ are missing which means that the carrier term is suppressed in the spectrum and only sideband terms are left.

(4) Bandwidth of DSB-SC signal $(B) = (\omega_c + \omega_m) - (\omega_c - \omega_m)$

$$\therefore \boxed{B = 2\omega_m}$$

Balanced Modulator



$$S(t) = 2K_a A_c \cos(2\pi f_c t)\, m(t)$$

Generation of DSBSC waves:-

- The expression for DSBSC signal is $s(t) = x(t)\cos\omega_c t$.
  where $x(t)$ = baseband signal ; $\cos\omega_c t$ = carrier signal.
- from this expression, we see that a DSBSC signal is basically the product of the modulating signal and carrier signal.
- A circuit used to generate a DSBSC signal is called a product modulator.

(i) Balanced modulator (ii) Ring modulator.

Balanced Modulators:-

- We know that a non linear device may be used to produce AM. i.e; one carrier and two side bands
- DSBSC signal contains only two side bands
- ∴ If two non linear devices are connected in a balanced mode so as to suppress the carriers, then only side bands will be left. i.e; DSBSC signal is generated.

∴ A **Balanced modulator** may be defined as a circuit in which two non linear devices are connected in a balanced mode to produce a DSBSC signal.

Balanced modulator using diodes:-

- A modulating signal $x(t)$ is applied to the two diodes through a centre tapped transformer with the carrier signal $\cos\omega_c t$.
- A non linear v-i relationship is given by $i = av + bv^2$.
- where v is the input voltage applied across a non linear device and i is the current through the non linear device.
- from the fig. the two i/p voltages v_1 and v_2 across the diodes are

$v_1 = \cos\omega_c t + x(t)$; $v_2 = \cos\omega_c t - x(t)$.

- for diode D_1, the non linear v-i relationship becomes

$i_1 = av_1 + bv_1^2$ i.e. $i_2 = av_2 + bv_2^2$

$\Rightarrow i_1 = a[\cos\omega_c t + x(t)] + b[\cos\omega_c t + x(t)]^2$

$= a\cos\omega_c t + a x(t) + b\cos^2\omega_c t + b x^2(t) + 2bx(t)\cos\omega_c t$

i.e. $i_2 = a[\cos\omega_c t - x(t)] + b[\cos\omega_c t - x(t)]^2$

$= a\cos\omega_c t - a x(t) + b\cos^2\omega_c t + b x^2(t) - 2bx(t)\cos\omega_c t$

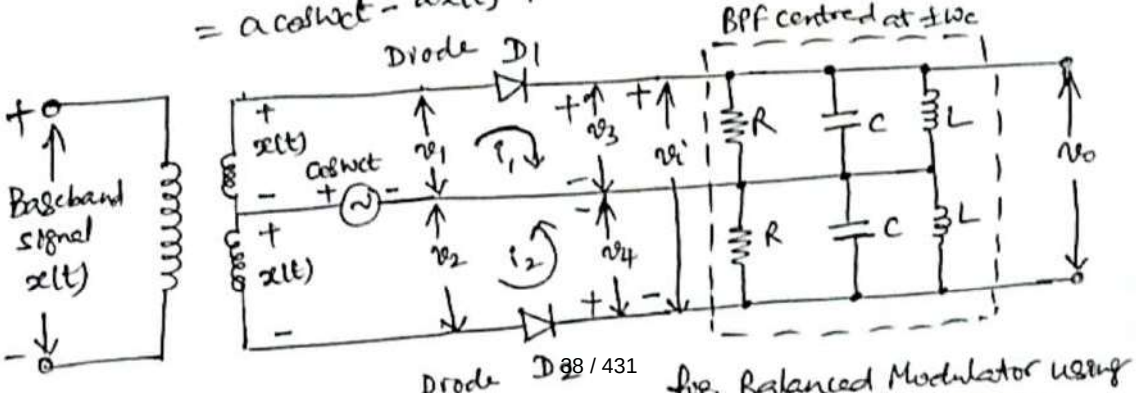


fig. Balanced Modulator using diodes

- The net voltage v_i' at the i/p of BPF is $(v_3 - v_4)$; $v_i' = v_3 - v_4$
- But $v_3 = i_1 R$; $v_4 = i_2 R$
- $\therefore v_i = i_1 R - i_2 R = R(i_1 - i_2)$
- $v_i = R[2a x(t) + 4b x(t) \cos \omega_c t] = \underline{2R[a x(t) + 2b x(t) \cos \omega_c t]}$
- This voltage v_i is the input to the BPF centered around $\pm \omega_c$.
- This BPF will pass a narrow band of freqs centred at $\pm \omega_c$ with a small bandwidth of $2\omega_m$ to preserve the sidebands.
- $\therefore$ The o/p of BPF (centred around $\pm \omega_c$) is given by

$$\boxed{v_0 = 4bR\, x(t) \cos \omega_c t = K \underline{x(t) \cos \omega_c t}.}$$

which is the expression for a DSBSC signal. where $K = 4bR$.

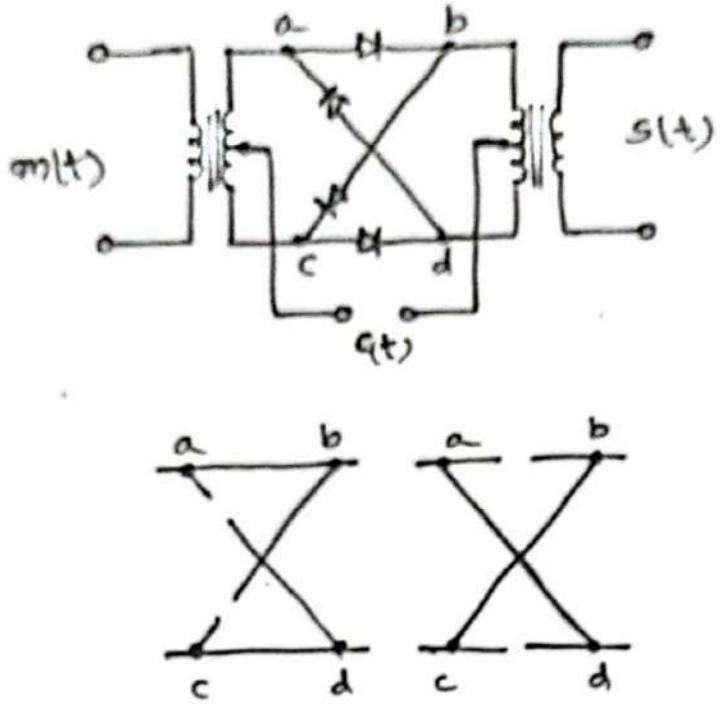
Ring Modulator:— [lattice (or) double balanced modlr]

- One of the most useful product modlr's for generating DSBSC modd. wave.
- The 4 diodes form a ring in which they all point in the same way.
- The diodes are controlled by a square wave carrier c(t) of frequency fc, which is applied by means of two center tapped transformers.
- We assume that the diodes are ideal and the transformers are perfectly balanced.
- When the carrier supply is +ve, the outer diodes are switched ON, presenting zero impedance, where as inner diodes are switched OFF, presenting infinite impedance, so that the modlr multiplies the message signal m(t) by +1.
- When the carrier supply is -ve, the situation becomes reversed and the modulator multiplies the message signal by -1
- Thus the ring modulator, in its ideal form, is a product modlr for a square wave carrier and message signal, for the case of a sinusoidal modulating wave.
- The square wave carrier c(t) can be expressed by a fourier series as $$c(t) = \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c t(2n-1)]$$
- The ring modulator o/p is therefore $s(t) = c(t)\, m(t)$.

$$\Rightarrow s(t) = \frac{4}{\pi} \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c t(2n-1)]\, m(t).$$

- We see that there is no o/p from the modulator at fc; that is the modulator o/p consists entirely of modlm products.

RING MODULATOR



(a) circuit diagram

fig1. Ring Modulator

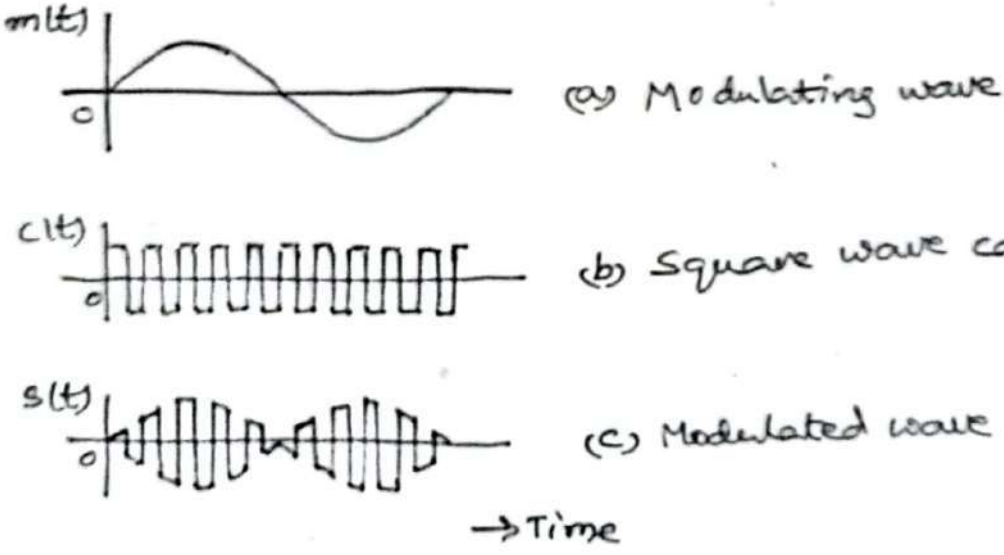
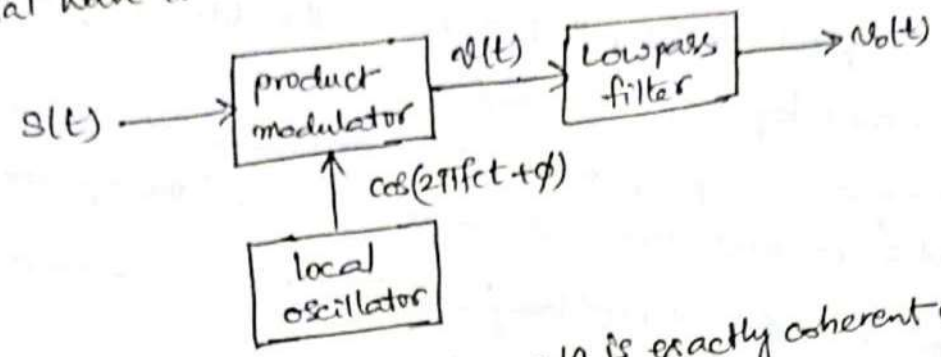


fig2. Wave forms illustrating the operation of the ring modulator. for a sinusoidal modulating wave.

Coherent detection of DSBSC modulated waves:-

- The message signal $m(t)$ is recovered from a DSBSC wave $s(t)$ by first multiplying $s(t)$ with a locally generated sinusoidal wave and then low-pass filtering the product.



- It is assumed that the L.O. o/p is exactly coherent or synchronized, in both frequency and phase with the carrier wave $c(t)$ used in the product modulator to generate $s(t)$.
- This method of demodulation is known as coherent detection or synchronous detection.
- Derivation of coherent detection as a special case of the more general demodulation process using a local oscillator signal of the same frequency but arbitrary phase difference ϕ, measured w.r.t the carrier wave $c(t)$.
- Thus denoting the L.O. signal by $\cos(2\pi f_c t+\phi)$ assumed to be of unit amplitude for convinience, the product modulator o/p is

$$v(t) = \cos(2\pi f_c t+\phi)\, s(t).$$
$$= A_c \cos(2\pi f_c t)\, m(t) \cos(2\pi f_c t+\phi)$$
$$= \underbrace{\tfrac{1}{2} A_c \cos\phi\, m(t)}_{\text{scaled version of message signal.}} + \underbrace{\tfrac{1}{2} A_c \cos(4\pi f_c t+\phi)\, m(t)}_{\text{unwanted term}}$$

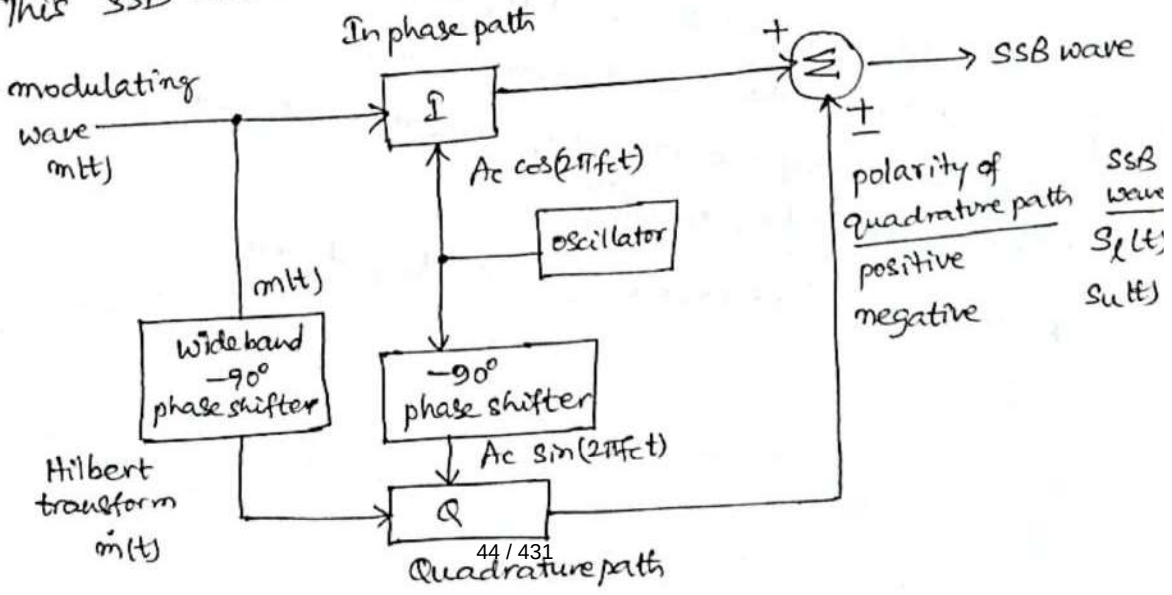
- The LPF removes the unwanted term in the product modulator o/p.
- The overall o/p $v_o(t) = \tfrac{1}{2} A_c \cos\phi\, m(t)$
- The demodulated signal $v_o(t)$ is therefore proportional to $m(t)$ when the phase error ϕ is a constant.

- The amplitude of this demodulated signal is maximum when $\phi = 0$, and is minimum (zero) when $\phi = \pm \frac{\pi}{2}$.
- The zero demodulated signal represents the quadrature null effect of the coherent detector.
- Thus the phase error ϕ in the L.O. causes the detector o/p to be attenuated by a factor = $\cos\phi$.
- As long as the phase error ϕ is constant, the detector o/p provides an undistorted version of the original message signal only.
- In practice, ϕ varies randomly with time, due to random variations in the communication channel.
- $\Rightarrow$ At the det o/p, $\cos\phi$ also varies randomly with 't', which is undesirable.
- $\therefore$ circuitry must be provided in the receiver to maintain the local oscillator in perfect synchronism, in both freq. and phase, with the carrier wave used to generate the DSBSC modd wave in the transmitter.
- The resulting increase in receiver complexity is the price that must be paid for suppressing the carrier wave to save transmitter power.

Phase discrimination method for generating an SSB modd wave:-

- The phase discrimination method of generating an SSB modulated wave involves two seperate simultaneous modl^n processes and subsequent combination of the resulting modl^n products as shown in fig.

- The system uses two product modulators, I and Q, supplied with carrier waves in phase quadrature to each other.
- The incoming base band signal m(t) is applied to product modl^r I, producing a modulated DSBSC wave that contains reference phase sidebands symmetrically spaced about carrier freq fc.
- The Hilbert transform $\hat{m}(t)$ of m(t) is applied to product modl^r Q, producing a DSBSC modd wave that contains sidebands having identical amplitude spectra to those of modulator I, but with phase spectra such that vector addition or subtraction of the two modulator outputs results in cancellation of one set of sidebands and reinforcement of the other set.
- The use of plus sign at the summing junction yields an SSB wave with only the lower sideband, whereas the use of a minus sign yields an SSB wave with only the upper sideband.
- In this way the desired SSB modulated wave is produced.
- This SSB modulator is also known as Hartley modulator.



Demodulation of SSB waves :-

- To recover the baseband signal $x(t)$ from the SSB wave $s(t)$ equal to $s_u(t)$ or $s_l(t)$, we have to shift the spectrum of $S_u(w)$ or $S_l(w)$ by the amount $\pm f_c$ so as to convert the transmitted side band back in to the baseband signal.
- This can be accomplished using coherent detection, which involves applying the SSB wave $s(t)$, together with a locally generated carrier $\cos 2\pi f_c t$, assumed to be of unit amplitude for convinience, to a product modulator and then low pass filtering the modulator o/p as shown in fig.

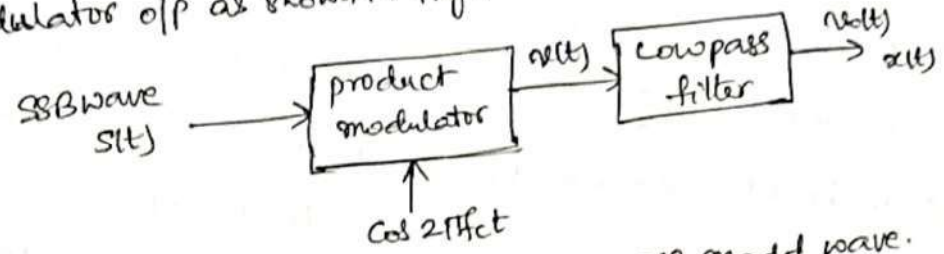


Fig. coherent detection of an SSB modd wave.

The product modulator o/p is given by

$$v(t) = s(t) \cdot \cos 2\pi f_c t$$

$$= \left[x(t)\cos\omega_c t \mp \hat{x}(t)\sin\omega_c t\right]\cos\omega_c t$$

$$= x(t)\cos^2\omega_c t \mp \hat{x}(t)\sin\omega_c t\cos\omega_c t$$

$$= \frac{x(t)}{2}(1+\cos 2\omega_c t) \mp \frac{\hat{x}(t)}{2}\sin 2\omega_c t$$

$$v(t) = \underbrace{\frac{1}{2}x(t)}_{\text{scaled message signal}} + \underbrace{\frac{1}{2}\left[x(t)\cos 2\omega_c t \mp \hat{x}(t)\sin 2\omega_c t\right]}_{\text{unwanted component}}$$

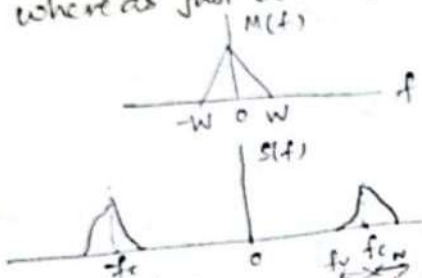
When $v(t)$ is passed through LPF, the unwanted component is removed.

[The first term is the desired message signal.
The combination of the remaining terms represents and SSB modd wave with a carrier freq of $2f_c$]

At the o/p of detector we get $v_0(t) = \frac{1}{2}x(t)$.

Vestigial Sideband (VSB) Modulation:-

- SSB modulation is well suited for the tx^{n} of voice because of the energy gap that exists in the spectrum of voice signals between zero and a few hundred hertz.
- When the message signal contains significant components at extremely low frequencies (as in the case of TV signals and wideband data), the upper and lower sidebands meet at the carrier freq.
- This means that the use of SSB modulation is inappropriate for the transmission of such message signals, because of the difficulty of isolating one sideband (from the other).
- This difficulty has been overcome by VSB modulation, which is a compromise between SSB and DSBSC modulation.
- In VSB modulation, one sideband is passed almost completely whereas just a trace, or vestige of the other sideband is retained.



(a) Spectrum of message signal

(b) Spectrum of VSB modulated wave containing a vestige of the LSB.

Frequency domain description:-

- Fig. illustrates the spectrum of a VSB modulated wave s(t) in relation to that of the message signal m(t), assuming that the lower sideband is modified in to the VSB.

* Specifically the transmitted vestige of the LSB compensates for the amount removed from the USB.

- The tx^{n} BW required by the VSB modulated wave is given by

$$B = W + f_v \qquad (f_c + f_m - (f_c - f_v) = f_m + f_v = W + f_v)$$

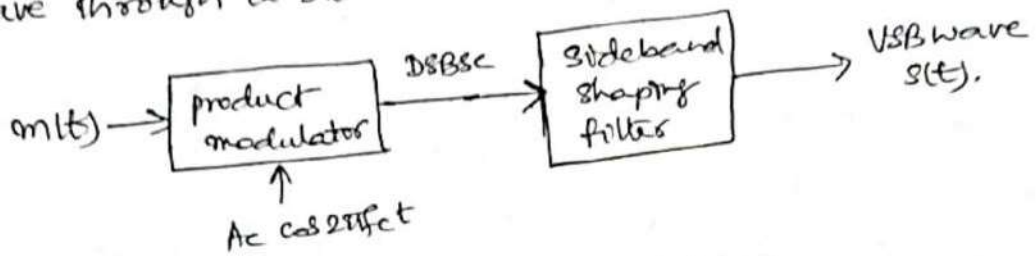
where W is the message BW and f_v is the width of the VSB.

- VSB modn has the virtue ( ) of conserving BW almost as efficiently as SSB modn, while retaining the excellent LF Baseband characteristics of DSB modulation.
- Thus VSB modn has become standard for the tx^{n} of TV and similar signals where good phase characteristics and tx^{n} of LF components

are important, but the BW required for DSB tx^n is unavailable or ~~unecono~~ uneconomical.

Generation of VSB modulated wave :-

- To generate a VSB modd wave, we pass a DSBSC modd wave through a sideband shaping filter as shown in fig.



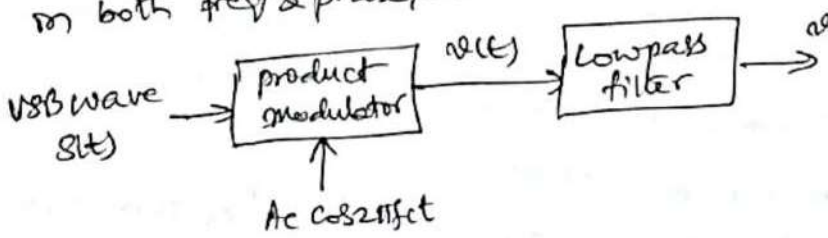
① Block diagram of VSB modulator.

- The exact design of this filter depends on the desired spectrum of the VSB modd wave.
- The relation b/n the transfer fn H(f) of the filter and the spectrum S(f) of the VSB modd wave S(f) is defined by

$$S(f) = \frac{Ac}{2}\left[M(f-fc) + M(f+fc)\right] H(f) \quad \text{---(1)}$$

where M(f) is the message spectrum.

- We wish to determine the specification of the filter transfer function H(f), so that S(f) defines the spectrum of the desired VSB wave s(t).
- This can be established by passing s(t) through a coherent detector and then determining the necessary condition for the detector output to provide undistorted version of the original message signal m(t).
- Thus multiplying s(t) by a locally generated (carrier) sine wave $\cos 2\pi fct$ which is synchronous with the carrier wave Ac cos2πfct in both freq & phase, as shown in fig, we get $v(t) = \cos 2\pi fct \cdot s(t)$ — (2)



② Block diagram of VSB demodulator.

- Transforming this relation into the freq domain gives the fourier transform of $v(t)$ as

$$V(f) = \frac{1}{2}\left[S(f-f_c)+S(f+f_c)\right] \quad -③$$

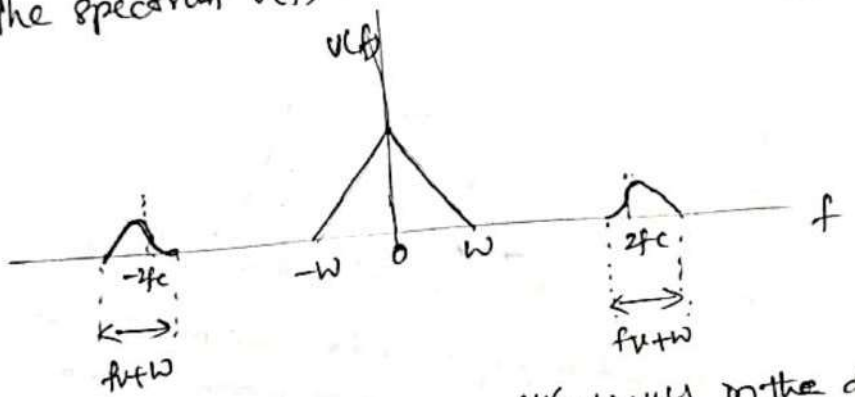
- ∴ Substitution of eq ① in ③ yields.

$$V(f) = \frac{1}{2}\cdot\frac{A_c}{2}\left[\{M(f-f_c-f_c)+M(f+f_c-f_c)\}H(f-f_c)+\{M(f-f_c+f_c)+M(f+f_c+f_c)\}H(f+f_c)\right]$$

$$V(f) = \frac{A_c}{4}\left[\{M(f-2f_c)+M(f)\}H(f-f_c)+\{M(f)+M(f+2f_c)\}H(f+f_c)\right]$$

$$\Rightarrow V_o(f) = \frac{A_c}{4}M(f)\left[H(f-f_c)+H(f+f_c)\right] + \frac{A_c}{4}\left[M(f-2f_c)H(f-f_c)+M(f+2f_c)H(f+f_c)\right] \quad -④$$

- The spectrum $V(f)$ is illustrated in fig.

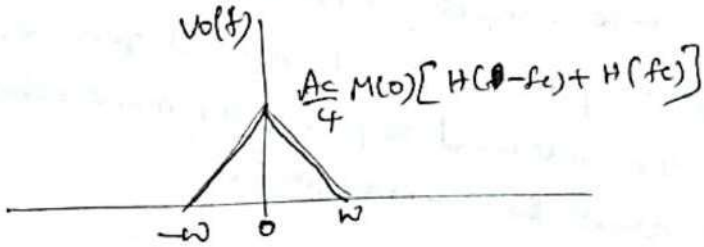


⑤ Spectrum of the product modⁿ o/p $v(t)$ in the demodulation

- The 2nd term in eq ④ represents a VSB wave corresponding to a carrier freq $2f_c$.
- This term is removed by the LPF in fig ② to produce an o/p $v_o(t)$, the spectrum of which is given by

$$V_o(f) = \frac{A_c}{4}M(f)\left[H(f-f_c)+H(f+f_c)\right] \quad -⑤$$

- The spectrum $V_o(f)$ is illustrated in fig ④



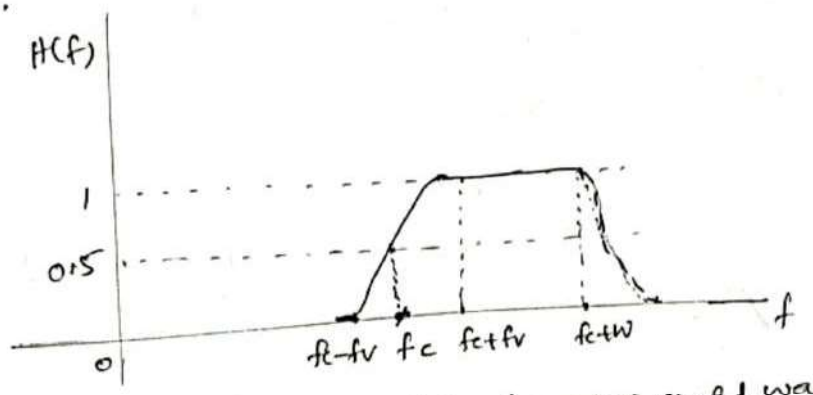
④ Spectrum of the demodulated signal $v_o(t)$.

- For a distortionless reproduction of the original baseband signal $m(t)$ at the coherent detector o/p, we require $V_o(f)$ to be a scaled version of $M(f)$.
- This means, i.e. that the transfer fn $H(f)$ must satisfy the condition,

$$H(f-f_c) + H(f+f_c) = 2H(f_c) \quad \text{---} \quad (6)$$

where $H(f_c)$ is a constant.

- With the message spectrum $M(f)$ assumed to be essentially zero outside the interval $-W \le f \le W$, we need to satisfy Eq (6) only for the values of f in this interval.
- The requirement of (6) is satisfied by using a filter with a freq response $H(f)$ such as that shown in fig (5) for positive frequencies.



(5) freq response of sideband shaping filter for a VSB modulated wave containing a vestige of LSB. only the +ve freq portion is shown.

- The response is normalised so that $H(f)$ falls to one half at the carrier freq f_c.
- The cutoff portion of this response around f_c exhibits odd symmetry in the sense that inside the transition interval defined by $f_c - f_v \le f \le f_c + f_v$ the sum of the values of $H(f)$ at any two freqs equally displaced above and below f_c is unity.

Such a filter is much less elaborate than that required if one sideband is to be completely suppressed.

- In general, to preserve the baseband spectrum, the phase response of the sideband shaping filter must exhibit odd symmetry about the carrier freq f_c.

- Specifically it must be linear over the freq intervals

$$f_c - f_v \le |f| \le f_c + W$$

and its value at the freq "f_c" has to equal zero or an integer multiple of 2π radians.

- The effect of this linear phase char. is merely to introduce a constant delay in the recovery of the message signal $m(t)$ at the receiver o/p.
- The freq response of fig pertains to a VSB modd wave containing a vestige of the LSB.
- In the situation depicted here, control over the freq response of the sideband shaping filter need only be exercised over the band $f_c - f_v \le |f| \le f_c + W$.
- This is the reason for showing the freq response of the sideband shaping filter in the fig for $f > f_c + W$ as a dashed line.

Time domain description:-

[Let $s(t)$ denote a VSB modd wave containing a vestige of the lower sideband.
This modd wave may be viewed as the o/p of a sideband shaping filter produced in response to a DSBSC signal $(m(t) A_c \cos 2\pi f_c t)$.
The filter has a tr. fn $H(f)$ as shown in fig (5) above.]
This can be obtained by using canonical form of the bandpass signal

$$s(t) = s_c(t)\cos 2\pi f_c t - s_s(t)\sin 2\pi f_c t \quad \text{(DSB signal)} \quad -(7)$$

$$S(f) = S(f-f_c) + S(f+f_c) \quad -W \le f \le W$$
$$= 0 \quad \text{elsewhere.} \quad \Big\} -(8)$$

$$S_s(f) = j[S(f-f_c) - S(f+f_c)] \quad -W \le f \le W$$
$$= 0 \quad \text{elsewhere.} \quad \Big\} -(9)$$

Substituting eq (1) in eq (8)

$$S_c(f) = \frac{A_c}{2}\left[M(f-f_c-f_c) + M(f+f_c-f_c)\right] H(f-f_c)$$
$$+ \frac{A_c}{2}\left[M(f-f_c+f_c) + M(f+f_c+f_c)\right] H(f+f_c)$$
$$= \frac{A_c}{2} M(f)\left[H(f-f_c) + H(f+f_c)\right] \quad -(10)$$

Assuming the transfer function $H(f)$ of the VSB filter satisfies the eq (6) with $H(f_c)=0.5$, we obtain from eq (10)

$$S_c(f) = \frac{1}{2} A_c M(f) \cdot 2H(f_c)$$

$$= \frac{1}{2} A_c M(f) \cdot (2)(0.5)$$

$$S_c(f) = \frac{1}{2} A_c M(f) \quad — (11)$$

$$\Rightarrow s_c(t) = \frac{1}{2} A_c m(t) \quad — (12)$$

$S_s(f)$ is obtained by substituting (1) in (9)

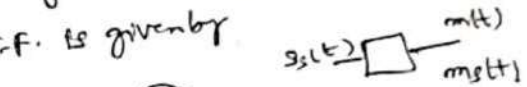
$$S_s(f) = j\left[\frac{A_c}{2}\{M(f-f_c-f_c)+M(f+f_c-f_c)\}H(f+f_c)\right.$$

$$\left. - \frac{A_c}{2}\{M(f-f_c+f_c)+M(f+f_c+f_c)\}H(f+f_c)\right]$$

$$S_s(f) = j\frac{A_c}{2} M(f)\left[H(f-f_c) - H(f+f_c)\right] \quad — (13)$$

From eq (13) it is clearly seen that we may generate the quadrature component $s_s(t)$, except for a scaling factor, by passing message signal through a filter whose T.F. is given by

$$H_s(f) = j\left[H(f-f_c) - H(f+f_c)\right] \quad — (14)$$



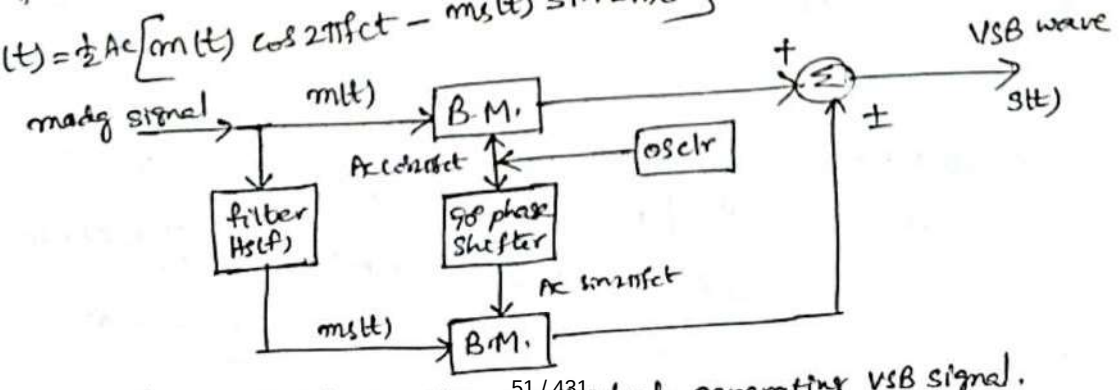
The freq response of the filter is shown below. assuming that $H(f)$ is defined as in fig. below.

Let $m_s(t)$ denote the o/p of this filter in response to $m(t)$. Thus we may express the quadrature component of the VSB signal as

$$s_s(t) = \frac{1}{2} A_c m_s(t) \quad — (15)$$

Substituting the values of $s_c(t)$ and $s_s(t)$ from eqs 12 & 15 respectively in eq (7) we obtain the time domain representation of VSB signal as

$$s(t) = \frac{1}{2} A_c\left[m(t)\cos 2\pi f_c t - m_s(t)\sin 2\pi f_c t\right] \quad — (16)$$



Phase discrimination method of generating VSB signal.

AM: advantages, disadvantages and applications.

advantages:

Easier to implement

Demodulation can be done using few components and a circuit

The receiver used for AM is very cheap.

disadvantages:

Inefficient power usage

Requires large bandwidth

Noise interference is highly noticeable.

applications:-

AM is used in broadcasting tx^n over short, medium, and long wave bands

AM is used in the VHF tx^{n}s for many air borne applications (ground to air radio comms., 2 way radio links for ground staff personnel.)

AM (SSB) is used for HF radio links or point to point HF links.

Virtues and limitations of AM.

Virtues:-

Modulation and demodulation is easy.
Modⁿ is done simply in the tx^r using a switching modⁿ or a square law modulator
Demodⁿ is done easily in the receiver using an envelope detector or a square law detector
AM system is relatively cheap to build

Limitations:-

1. AM is wasteful of power: The carrier wave is completely independent of the information bearing signal or base band signal m(t). The txⁿ of the carrier wave ∴ represents a waste of power.

2. AM is wasteful of Bandwidth: The upper and lower side bands of an AM wave are uniquely related to each other by virtue of their symmetry about the carrier freq. Hence given the amplitude and phase spectra of either sideband, we can uniquely determine the other. This means that in so far as the txⁿ of information is concerned, only one sideband is necessary, and the communication channel ∴ needs to provide only the same bandwidth as the base band signal.

In light of this, AM is wasteful of bandwidth as it requires a txⁿ BW = twice the message bandwidth.

A sinusoidal carrier is $c(t) = 100\cos(2\pi \times 10^5 t)$ is AMed by a sinusoidal voltage $m(t) = 50\cos(2\pi \times 10^3 t)$ up to a modⁿ depth of 50%. cal. the freq. and amplitude of each sideband.

freq of side bands $= 15^5 + 10^3$

$15^5 - 10^3$

amplitude of each side band $= \frac{mV_c}{2} = \frac{0.5 \times 100}{2} = 25$

A given AM broadcast station transmits a total power of 60 kW when the carrier is modulated by a sinusoidal signal with a modⁿ index of 0.707. cal. (i) carrier power (ii) t_x^n η (iii) peek amplitude of the carrier assuming the antenna to be represented by a $(50+j0)\Omega$.

$P_t = 60kW \qquad m_a = 0.707$

$P_t = P_c(1+\frac{m^2}{2})$

(i) $P_c = \frac{P_t}{1+\frac{m^2}{2}} = \frac{60k}{1+0.25} = 48\,KW$

(ii) $\eta = \frac{P_t - P_c}{P_t} = \frac{60k - 48k}{60k} = \frac{12}{60} = 20\%$

(iii) $\frac{A_c^2}{2\times 50} = 48k \qquad A_c = \sqrt{48k \times 100} = 2.19\,kV$

A certain Tx^r radiates 9KW with the carrier unmodulated and 10.125 KW with the carrier modulated by sinusoidal wave. cal. the modⁿ index. If another sine wave modulates the carrier to a depth of 40%, det the total radiated power.

$P_c = 9KW$ $\quad P_t = 10.125KW$ $\quad P_t = P_c\left(1+\frac{m^2}{2}\right)$

$$m = \sqrt{\left(\frac{P_t}{P_c}-1\right)2} = \left(\left(\frac{10.125}{9}-1\right)2\right)^{\frac{1}{2}} = 0.5$$

multitone modⁿ index $= \sqrt{0.5^2+0.4^2} = \underline{0.64}$

$$P_t = 9K\left(1+\frac{0.64^2}{2}\right) = 10.24\ KW.$$

A sinusoidal carrier voltage of freq 10KHz is amplitude modulated by 2 sine waves given by $V_1(t) = 5\cos(2000\pi t)$ and $V_2(t) = 3\cos 5000\pi t$. Assume the carrier amplitude as 10V. Find the freq. of sidebands, modⁿ indices and draw the spectrum of this multitone modulation.

$f_c = 10KHz.$

$V_1(t) = 5\cos(2000\pi t)$ $\quad f_{m_1} = \frac{2000\pi}{2\pi} = 100\ Hz.$

$V_2(t) = 3\cos(5000\pi t)$ $\quad f_{m_2} = \frac{5000\pi}{2\pi} = 250\ Hz.$

$f_{USB} = 10K+100,\ 10K+250$

$= 10.1\ KHz,\ 10.25\ KHz$

$f_{LSB} = 10K-100,\ 10K-250$

$= 9.9\ KHz,\ 9.75 KHz$

$m_{a_1} = \frac{5}{10} = 0.5,\quad m_{a_2} = \frac{3}{10} = 0.3.$

A broadcast AM Tx^r radiated 50KW of carrier power. what will be the radiated power at 85% modulation?

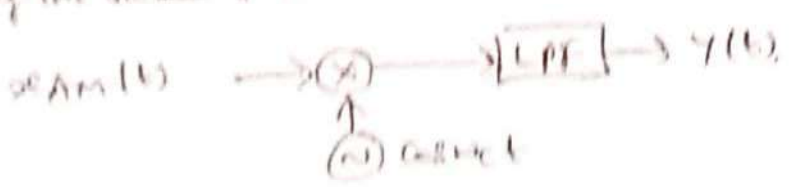
$$P_t = 50K\left(1+\frac{0.85^2}{2}\right) = 68kW.$$

A 400 W carrier is modulated on a depth of 75%. cal. P_t in modd wave in the following forms of AM (i) DSBSC (ii) SSBSC.

$$P_{DSBSC} = P_c\frac{m^2}{2} = 400\times\frac{0.75)^2}{2} = 112.5W.$$

$P_{SSBSC} = 56.25W. \Leftarrow \frac{P_{DSBSC}}{2}.$

Show that a synchronous demodulator shown in fig. below can demodulate an AM signal $s_{AM}(t) = [A + m(t)]\cos\omega_c t$ regardless of the value of A.



$$d(t) = s_{AM}(t)\cos\omega_c t$$
$$= [A + m(t)]\cos\omega_c t \cdot \cos\omega_c t$$
$$= [A + m(t)](1 + \cos 2\omega_c t)$$
$$= \frac{1}{2}[A + m(t)] + \frac{1}{2}[A + m(t)]\cos 2\omega_c t$$

$d(t)$ when passed through LPF $\Rightarrow y(t) = \frac{1}{2}m(t) + \frac{1}{2}A$

blocking capacitor suppresses the dc term $\frac{1}{2}A$.

$$\Rightarrow y(t) = \frac{1}{2}m(t).$$

Unit III Angle Modulation

Basic concepts:-

- Previously we investigated the effect of slowly varying the amplitude of a sinusoidal carrier wave in accordance with the baseband information bearing signal.
- There is another method of modulating a sinusoidal carrier wave, namely, angle modulation in which either the phase or frequency of the carrier wave is varied according to the message signal.
- In this method of modulation the amplitude of the carrier wave is maintained constant.
- The general form of modulated wave in angle modulation is

$$s(t) = A_c \cos[\theta(t)] \quad \text{—— (1)}$$

- where the carrier amplitude A_c is maintained constant, and the angular argument $\theta(t)$ is varied by a message signal $m(t)$.
- The mathematical form of this variation is determined by the type of angle modulation of interest.
- In any event, a complete oscillation occurs whenever $\theta(t)$ changes by 2π radians.
- If $\theta(t)$ increases monotonically with time, the average frequency in hertz, over an interval from t to $t+\Delta t$, is given by

$$f_{\Delta t}(t) = \frac{\theta(t+\Delta t) - \theta(t)}{2\pi \Delta t} \quad \text{—— (2)}$$

- We define the instantaneous frequency of the angle modulated wave $s(t)$ by

$$f_i(t) = \lim_{\Delta t \to 0} f_{\Delta t}(t)$$

$$= \lim_{\Delta t \to 0} \left[\frac{\theta(t+\Delta t) - \theta(t)}{2\pi \Delta t} \right]$$

$$= \frac{1}{2\pi} \frac{d}{dt} \theta(t) \quad \text{—— (3)}$$

- Thus according to eq(1) we may interpret the angle modd wave $s(t)$ as a rotating phasor of length A_c and angle $\theta(t)$.

- The angular velocity of such a phasor is $d\theta(t)/dt$, in accordance with eq ③.
- In the simple case of an unmodulated carrier, the angle $\theta(t)$ is $\theta(t) = 2\pi f_c t + \phi_c$
- and the corresponding phasor rotates with a constant angular velocity equal to $2\pi f_c$.
- The constant ϕ_c is the value of $\theta(t)$ at $t=0$.
- There are an infinite no. of ways in which the angle $\theta(t)$ may be varied in some manner with the message signal
- However we will consider only two commonly used methods, Phase modulation and frequency modulation, as next defined:

1. Phase modulation (PM) is that form of angle modulation in which the angular argument $\theta(t)$ is varied linearly with the message signal $m(t)$, as shown by

$$\theta(t) = 2\pi f_c t + k_p m(t) \quad \text{— ④}$$

- The term $2\pi f_c t$ represents the angular argument of the unmodulated carrier, and the constant k_p represents the phase sensitivity of the modulator, expressed in radians per volt. This assumes that $m(t)$ is a voltage waveform.
- For convinience we have assumed in eq ④ that the angular argument of the unmodulated carrier ($2\pi f_c t$) is zero at $t=0$.
- The phase modulated wave $s(t)$ is thus described in the time domain by $s(t) = A_c \cos\left[2\pi f_c t + k_p m(t)\right]$ — ⑤

2. Frequency modulation (FM) is that form of angle modulation in which the instantaneous frequency $f_i(t)$ is varied linearly with the message signal $m(t)$, as shown by

$$f_i(t) = f_c + k_f m(t) \quad \text{— ⑥}$$

The term f_c represents the freq. of the unmodulated carrier, and the constant k_f represents the frequency sensitivity of the modulator expressed in hertz per volt.

This assumes that m(t) is a voltage w.f.

- Integrating (6) wrt time and multiplying the result by 2π we get

$$\theta(t) = 2\pi f_c t + 2\pi K_f \int_0^t m(t)\,dt \quad -(7)$$

- where for convinience we have assumed that the angular argument of the unmodd carrier wave is zero at t=0.
- The frequency modulated wave is therefore described in the time domain by $s(t) = A_c \cos\left[2\pi f_c t + 2\pi K_f \int_0^t m(t)\,dt\right] - (8)$
- A consequence of allowing the angular argument $\theta(t)$ to become dependent on the message signal m(t) as in eq (4) or on its integral as in eq (7) is that the zero crossings of a PM wave or FM wave no longer have a perfect regularity in their spacing; zero crossings refer to the instants of time at which a w.f. changes from a -ve to a +ve value or vice-versa.
- This is one important feature that distinguishes both PM and FM waves from an AM wave.
- Another important difference is that the envelope of an PM or FM wave is constant (equal to the carrier amplitude), whereas the envelope of an AM wave is dependent on the message signal.
- Comparing eq (5) with (8) reveals that an FM wave may be regarded as a PM wave in which the modulating wave is $\int_0^t m(t)\,dt$ in place of m(t).
- This means that an FM wave can be generated by first integrating m(t) and then using the result as the i/p to a phase modulator.

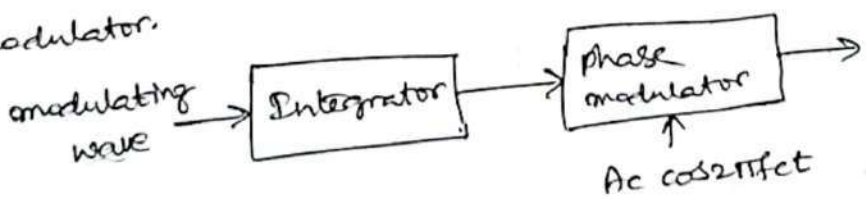


fig (a)

- Conversely, a PM wave can be generated by first differentiating m(t) and then using the result as the i/p to a freq modulator, we may thus deduce all the properties of PM waves from those of FM waves and vice-versa.

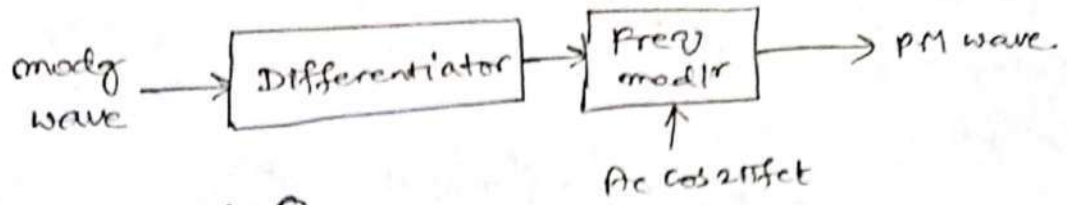


fig (b)

fig. The relationship between FM & PM.

(a) Scheme for generating an FM wave by using a phase modlr

(b) a PM freq

Frequency Modulation:-

The FM wave $s(t)$ defined by (8)

$$s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right]$$

is a nonlinear function of the modulating wave $m(t)$.

Hence, FM is a nonlinear modⁿ process.

Consequently, Unlike AM, the spectrum of an FM wave is not related in a simple manner to that of the modg wave.

Single tone frequency modulation:-

Consider a sinusoidal modg wave defined by

$$m(t) = A_m \cos(2\pi f_m t) \quad \text{—— (9)}$$

The instantaneous freq of the resulting FM wave equals

$$f_i(t) = f_c + k_f m(t)$$

$$= f_c + k_f A_m \cos(2\pi f_m t)$$

$$f_i(t) = f_c + \Delta f \cos(2\pi f_m t) \quad \text{—— (10)}$$

where $\Delta f = k_f A_m$ —— (11)

The quantity Δf is called the frequency deviation, representing the maximum departure of the instantaneous frequency of the FM wave from the carrier frequency f_c.

A fundamental char. of an FM wave is that the freq deviation Δf is proportional to the amplitude of the modg wave and is independent of the modulation freq.

Using eq (10) the regular argument $\theta(t)$ of the FM wave is obtained as $\theta(t) = 2\pi \int_0^t f_i(t)\,dt$

(3) $\Rightarrow f_i(t) = \frac{1}{2\pi}\frac{d}{dt}\theta(t)$

$2\pi f_i(t) = \frac{d}{dt}\theta(t)$

$\theta(t) = 2\pi\int_0^t f_i(t)\,dt$

$$\theta(t) = 2\pi\left[f_c t + \frac{\Delta f}{2\pi f_m}\sin(2\pi f_m t)\right]$$

$$\theta(t) = 2\pi f_c t + \frac{\Delta f}{f_m}\sin(2\pi f_m t) \quad \text{—— (12)}$$

The ratio of the freq deviation Δf to the modⁿ freq f_m is commonly called the modulation index of the FM wave.

It is denoted by β $\therefore \beta = \frac{\Delta f}{f_m}$ — (13)

and $\theta(t) = 2\pi f_c t + \beta \sin(2\pi f_m t)$ — (14)

From (14) we see that, in a physical sense, the parameter β represents the phase deviation of the FM wave, that is, the max. departure of the angular argument $\theta(t)$ from the angle $2\pi f_c t$ of the unmodd carrier.

Spectrum analysis of sinusoidal FM wave:-

The FM wave for sinusoidal modn is given by (*) using eq(14)

$$s(t) = A_c \cos\left[2\pi f_c t + \beta \sin(2\pi f_m t)\right] \quad — (15)$$

$$s(t) = A_c \, Re\left[\exp(j2\pi f_c t + j\beta \sin 2\pi f_m t)\right]$$

$$s(t) = A_c \cos(2\pi f_c t)\cos\left[\beta \sin(2\pi f_m t)\right] - A_c \sin(2\pi f_c t) \sin\left[\beta \sin(2\pi f_m t)\right] — (15)$$

$$s(t) = A_c \cos\left[\beta \sin(2\pi f_m t)\right] \cos(2\pi f_c t)$$

$$- A_c \sin\left[\beta \sin(2\pi f_m t)\right] \sin(2\pi f_c t) \quad — (16)$$

From Eq (16) The inphase and quadrature components of the FM wave $s(t)$ for the case of sinusoidal modn are as follows:

$$s_I(t) = A_c \cos\left[\beta \sin(2\pi f_m t)\right] \quad — (17)$$

$$s_Q(t) = A_c \sin\left[\beta \sin(2\pi f_m t)\right] — (18)$$

Hence, the complex envelope of the FM wave equals

$$\tilde{s}(t) = s_I(t) + j\, s_Q(t) = A_c \exp\left[j\beta \sin(2\pi f_m t)\right] — (19)$$

The complex envelope $\tilde{s}(t)$ retains complete information about the modulation process.

Indeed, we may readily express the FM wave $s(t)$ in terms of the complex envelope $\tilde{s}(t)$ by writing $s(t)$ as

$$s(t) = Re\left[A_c \exp(j2\pi f_c t + j\beta \sin(2\pi f_m t)\right]$$

$$s(t) = Re\left[\tilde{s}(t) \exp(j2\pi f_c t)\right] \quad — (20)$$

From eq (19) we see that the complex envelope is a periodic function of time, with a fundamental frequency equal to the modn freq f_m.

We may ∴ expand $\tilde{s}(t)$ in the form of a complex fourier series as

$$\tilde{s}(t) = \sum_{n=-\infty}^{\infty} c_n \exp(j2\pi n f_m t) \quad -(21)$$

where the complex fourier coefficients c_n equals

$$c_n = f_m \int_{-1/2f_m}^{1/2f_m} s(t) \exp(-j 2\pi n f_m t)\, dt$$

$$c_n = f_m A_c \int_{-1/2f_m}^{1/2f_m} \exp\left[j\beta \sin(2\pi f_m t) - j 2\pi n f_m t \right] dt \quad -(22)$$

$t = \frac{x}{2\pi f_m}$, $dt = \frac{dx}{2\pi f_m}$, $\frac{1}{2f_m} = \frac{\pi}{\pi}$

Let $x = 2\pi f_m t$ — (23)

$$(22) \Rightarrow c_n = \frac{A_c}{2\pi} \int_{-\pi}^{\pi} \exp\left[j(\beta \sin x - nx) \right] dx \quad -(24)$$

The integral on the right side of eq (24) is recognized as the nth order Bessel function of the first kind and the argument β.

This fn is commonly denoted by the symbol $J_n(\beta)$.

$$\therefore J_n(\beta) = \frac{1}{2\pi} \int_{-\pi}^{\pi} \exp\left[j(\beta \sin x - nx) \right] dx \quad -(25)$$

$$\therefore (24) \Rightarrow c_n = A_c J_n(\beta) \quad -(26)$$

Substituting (25) in (21)

$$\tilde{s}(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \exp(j2\pi n f_m t) \quad -(27)$$

Substituting (27) in (20)

$$s(t) = A_c \, \mathrm{Re}\left[\sum_{n=-\infty}^{\infty} J_n(\beta) \exp\left[j2\pi (f_c + n f_m) t \right] \right] \quad -(28)$$

Interchanging the order of summation and evaluating the real part of the right side of (28), we get

$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos\left[2\pi (f_c + n f_m) t \right] \quad -(29)$$

This is the desired form for the fourier series representation of the single tone FM wave $s(t)$ for an arbitrary value of β. The discrete spectrum of $s(t)$ is obtained by taking the fourier transforms of both sides of (29).

$$\therefore S(f) = \frac{Ac}{2} \sum_{n=-\infty}^{\infty} J_n(\beta)\left[\delta(f_c - fc - nfm) + \delta(f + fc + nfm)\right] - (30)$$

In the fig shown Bessel function $J_n(\beta)$ versus modulation index β for $n = 0, 1, 2, 3, 4$ are plotted.

These plots show that for fixed n, $J_n(\beta)$ alternates between +ve and -ve values for increasing β and that $|J_n(\beta)|$ approaches zero as β approaches infinity.

Note also that for fixed β, we have

$$J_{-n}(\beta) = \begin{cases} J_n(\beta) & n \text{ even} \\ -J_n(\beta) & n \text{ odd} \end{cases} \quad - (31)$$

Accordingly, we need only plot or tabulate $J_n(\beta)$ for +ve values of order n.

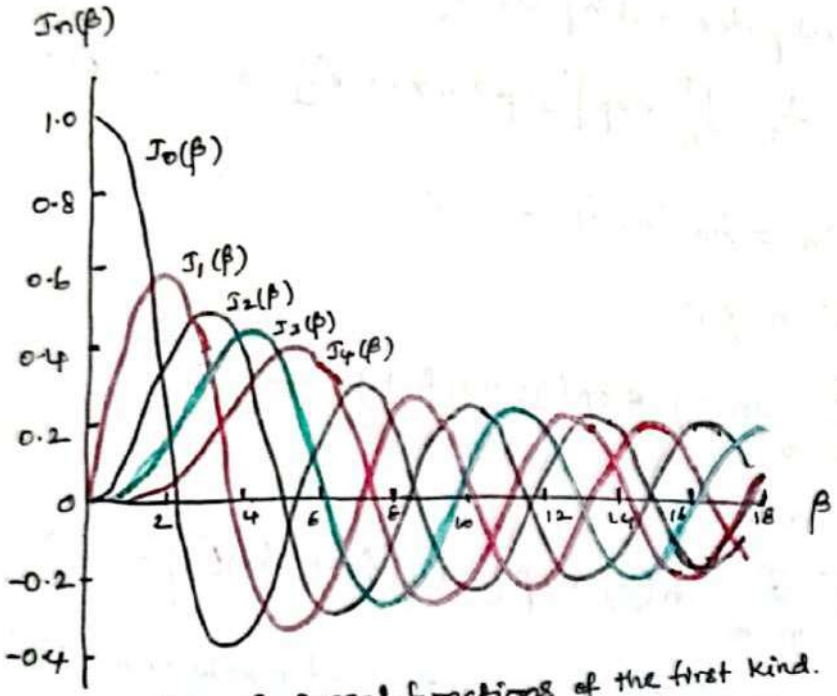
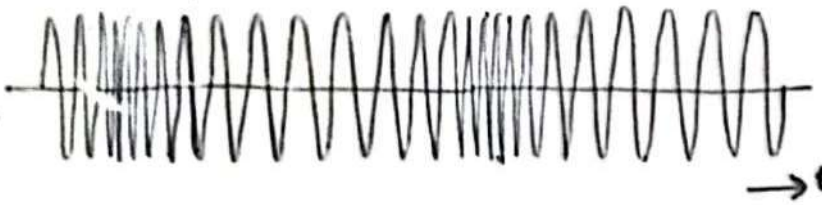
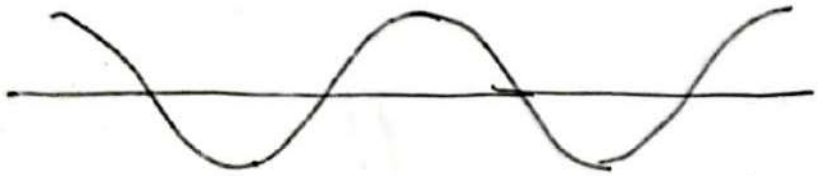


Fig. Plots of Bessel functions of the first kind.

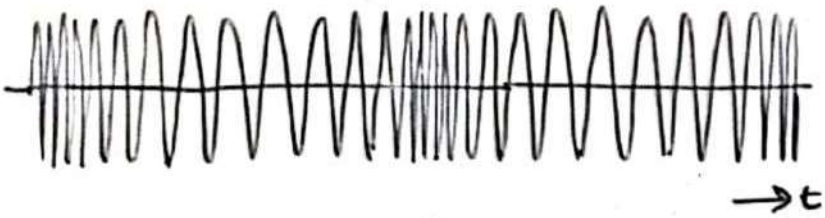
(a) sinusoidal modg wave m(t)



(b) Freq. modd wave



(c) derivative of m(t) w.r.t t



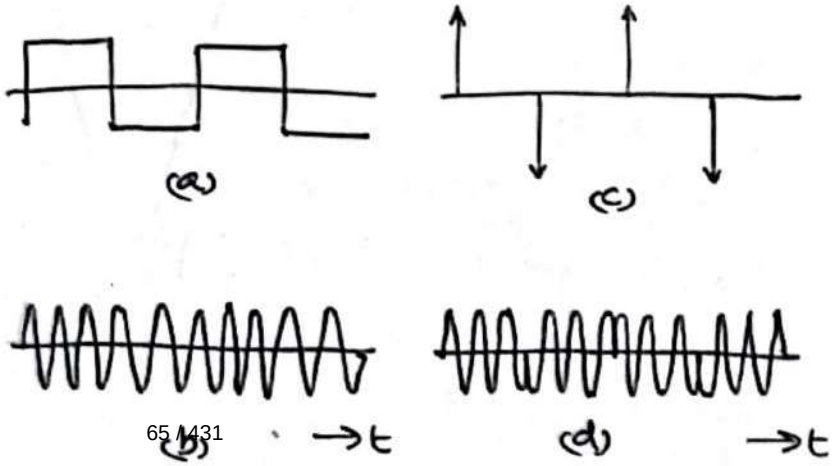
(d) phase modd wave.

(a) square modg wave m(t)

(b) FMd wave

(c) der. of m(t) w.r.t. t.

(d) phase modd wave



Narrow Band FM:-

For small values of the modl^n index β compared to one radian, the FM wave assumes a narrowband form consisting essentially of a carrier, an upper side freq component, and a lower side freq component.

This property follows from the fact that for small values of β, we have $J_0(\beta) \cong 1$

$$J_1(\beta) \cong \frac{\beta}{2}$$

$$J_n(\beta) \cong 0 \quad n > 1. \quad \text{— (32)}$$

The approximations indicated in eq (32) are closely justified for values of the modl^n index defined by $\beta < 0.3$ rad.

Thus substituting eq (32) in (29) we get

$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta) \cos\left[2\pi(f_c + nf_m)t\right]$$

$$s(t) = A_c \cos 2\pi f_c t + \frac{\beta A_c}{2} \cos\left[2\pi(f_c + f_m)t\right] - \frac{\beta A_c}{2} \cos\left[2\pi(f_c - f_m)t\right] \quad \text{— (33)}$$

This eq shows that for small β, the FM wave $s(t)$ may be closely approximated by the sum of a carrier of amplitude A_c, an upper side freq component of amplitude $\frac{\beta A_c}{2}$, and a lower side freq component of amplitude $\frac{\beta A_c}{2}$ and a phase shift equal to 180° (represented by the minus sign in (33))

An FM wave so characterized is said to be narrow-band.

Wide band FM:-

For large values of β compared to one rad, the FM wave (in theory) contains a carrier and an infinite no. of side freq components located symmetrically around the carrier.

This property is a restatement of eq (29) with no approxns made.

An FM wave thus defined is said to be wideband.

Note that the amplitude of the carrier component contained in a wideband FM wave varies with β in accordance with $J_0(\beta)$

Constant average power:-

The envelope of an FM wave is constant, so that the average power of such a wave dissipated in a 1-ohm resistor is also constant.

This property follows directly from the def. of FM wave (9). Specifically, the FM wave $s(t)$ defined in (9) has a constant envelope equal to A_c.

Accordingly the average power dissipated by $s(t)$ in a 1-ohm resistor is given by $P = \frac{1}{2} A_c^2$ — (34)

The result may also be derived from eq (29)

In particular we note from the series expansion of eq (29) that the average power of a single tone FM wave $s(t)$ may be expressed in the form of a corresponding series as

$$P = \frac{1}{2} A_c^2 \sum_{n=-\infty}^{\infty} J_n^2(\beta) \quad \text{— (35)}$$

Next, we note that $\sum_{n=-\infty}^{\infty} J_n^2(\beta) = 1$ — (36)

Thus substituting (36) in (35) we get the result given in eq (34).

Transmission Bandwidth of FM waves:

- Theoretically an FM wave contains an infinite no of sidefreqs so that the BW required to transmit such a signal is infinite.
- In practice, we find that the FM wave is effectively limited to a finite no of significant side freqs compatible with a specified amount of distortion.
- We may therefore specify an effective BW requirement for the tx^n of an FM wave.
- Consider first the case of an FM wave generated by a single tone modg wave of freq fm.
- In such an FM wave, the side freqs that are seperated from the carrier freq fc by an amount greater than the freq deviation Δf decrease rapidly toward zero, so that the BW always exceeds the total freq excursion, but nevertheless (inspite of that) is limited.
- Specifically, for large values of the modulation index β, the BW approaches (and is only slightly greater than) the total frequency excursion $2\Delta f$.
- On the otherhand, for small values of the modulation index β, the spectrum of the FM wave is effectively limited to the carrier freq fc and one pair of side frequencies at $f_c \pm f_m$, so that the BW approaches $2f_m$.
- We thus define a rule for the transmission Bandwidth of an FM wave generated by a single tone modulating wave of freq fm as

$$B = 2\Delta f + 2f_m$$
$$= 2\Delta f\left(1 + \frac{2f_m}{2\Delta f}\right)$$
$$= 2\Delta f\left(1 + \frac{1}{\beta}\right)$$

This relation is known as Carson's rule.

• For a more accurate assessment of the BW requirement of an FM wave, we may use a definition based on retaining the max. no. of significant side freqs with amplitudes all greater than some specified value.

• A convenient choice for this value is 1% of the unmodd carrier amplitude.

• We may thus define the 99% BW of an FM wave as the seperation between the two freqs beyond which none of the side freqs is greater than 1% of the carrier amplitude obtained when the modⁿ is removed.

• That is, we define the txⁿ BW as $2n_{max} f_m$, where f_m is the modulation freq and n_{max} is the max. value of the integer n that satisfies the requirement $|J_n(\beta)| > 0.01$

• The value of n_{max} varies with β and can be detd readily from the tabulated values of the bessel fn $J_n(\beta)$.

• Table shows the total no. of significant side freqs (including both the upper & lower side freqs) for different values of β, calculated on the 1% basis just explained.

(β) (modⁿ index)	(no. of significant side freqs) ($2n_{max}$)
0.1	2
0.3	4
0.5	4
1.0	6
2.0	8
5.0	16
10.0	28
20.0	50
30.0	70.

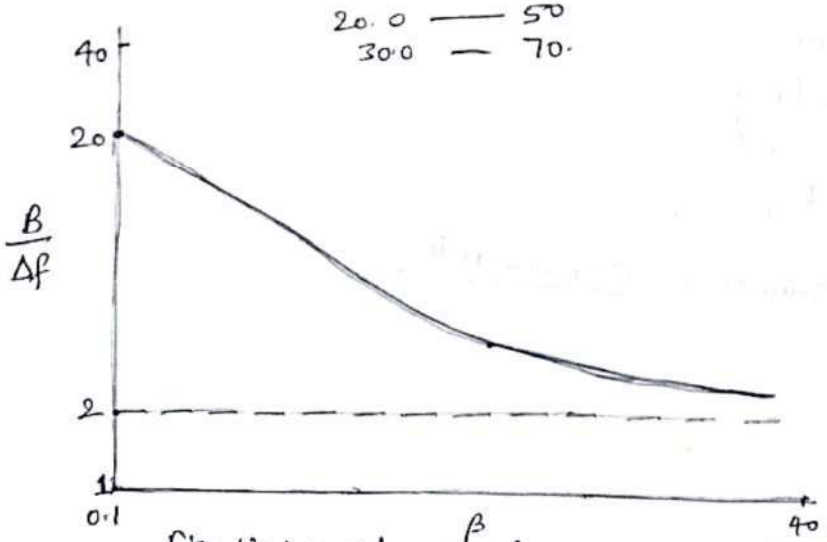


Fig Universal curve for evaluating the 99% BW of an FM wave

- The tx^n BW 'B' calculated using this procedure can be presented in the form of a universal curve by normalizing it wrt the freq deviation Δf, and then plotting it versus β.
- This curve is shown in fig, which is drawn as a best fit through the set of points obtained by using the table.
- from the curve, we note that as β is increased, the BW occupied by the significant side freqs drops toward that over which the carrier freq actually deviates.
- This means that small values of β are relatively more extravagant, (exceeding reasonable limits) in tx^n BW than are the larger values of β.

Consider next an arbitrary mlt) with its highest freq component denoted by W.

- The BW required to transmit an FM wave generated by this mlt) is estimated by using a worst case tone modl^n analysis
- Specifically we first determine the so called deviation ratio D, defined as the ratio of the freq deviation Δf, which corresponds to the max possible amplitude of the mlt), to the highest modl^n freq W; these conditions represent the extreme cases possible.
- The deviation ratio D plays the same role for nonsinusoidal modl^n that the modl^n index β plays for the case of sinusoidal modl^n.
- Then replacing β by D and replacing f_m by W, we use carson's rule or the universal curve to obtain a value for the tx^n BW of an FM wave.

$$\beta = \frac{\Delta f}{f_m} \qquad D = \frac{\Delta f}{W}$$

* From a practical view point, carson's rule some what underestimates the BW requirement of an FM sys., where as using the universal curve yields a some what conservative (of an estimate) result.
- Thus the choice of a tx^n BW that lies b/n the bounds provided by these two rules of thumb is acceptable for most practical purposes.

Generation of FM waves:-

There are essentially two basic methods of generating FM waves.

(1) Indirect FM

(2) Direct FM.

(1) The modg wave is first used to produce a NBFM wave, and freq. multiplication is next used to increase the freq deviation to the desired level.

(2) The carrier freq is directly varied in accordance with the incoming message signal.

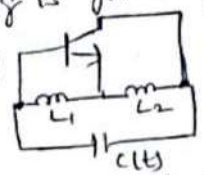
Direct FM:-

* The instantaneous freq of the carrier wave is varied directly in accordance with the message signal by means of a device known as voltage controlled oscillator.

- One way of implementing such a device is to use a sinusoidal oscl having a relatively high Q freq determining n/w and to control the oscillator by incremental variation of the reactive components.
- An ex: of this scheme is shown in fig. showing a Hartley oscl.
- We assume that the capacitive component of the freq determining n/w consists of a fixed capacitor shunted by a voltage variable capacitor.
- The resultant capacitance is represented by C(t) in fig.
- A voltage variable capacitor, commonly called a varactor or varicap is one whose capacitance depends on the voltage applied across its electrodes.
- The variable voltage capacitance may be obtained, for example, by using a p-n junction diode that is biased in the reverse direction, the larger the reverse voltage applied to such a diode, the smaller the transition capacitance of the diode.
- The freq of oscln of the Hartley oscillator of fig is given by

$$f_i(t) = \frac{1}{2\pi\sqrt{(L_1+L_2)\,C(t)}} \qquad \text{—— (1)}$$

fig. hartley oscl



where C(t) is the total capacitance of the fixed capacitor and the variable voltage capacitor, and
L_1 and L_2 are the two inductances in the freq determining n/w.

- Assume that for a modg wave m(t) the capacitance C(t) is expressed as follows:

$$C(t) = C_0 - K_c\, m(t) \qquad \text{—— (2)}$$

where C_0 is the total capacitance in the absence of modⁿ, and K_c is the variable capacitor's sensitivity to voltage change.

Substituting (2) in (1) we get

$$f_i(t) = \frac{1}{\left(2\pi\sqrt{(L_1+L_2)(C_0 - k_c\, m(t))}\right)^{1/2}}$$

$$= \frac{1}{2\pi\sqrt{C_0(L_1+L_2)}\ \sqrt{1-\frac{k_c}{C_0}m(t)}}$$

$$\boxed{f_i(t) = \frac{f_0}{\sqrt{1-\frac{k_c}{C_0}m(t)}} = f_0\left(1-\frac{k_c}{C_0}m(t)\right)^{-1/2}} \quad -③$$

where f_0 is the unmodd freq of oscln $f_0 = \frac{1}{2\pi\sqrt{C_0(L_1+L_2)}}$ —④

Provided that the max. change in capacitance produced by the modg wave is small compared with the unmodd capacitance C_0, we may approximate ③ as follows.

$$f_i(t) = f_0\left(1-\frac{k_c}{C_0}m(t)\right)^{-1/2}$$

$$\cong f_0\left(1+\tfrac{1}{2}\frac{k_c}{C_0}m(t)\right)$$

$$f_i(t) \cong f_0\left(1+\frac{k_c}{2C_0}m(t)\right) \quad —⑤$$

$$= f_0 + \frac{f_0 k_c}{2C_0}m(t).$$

Define $k_f = \frac{f_0 k_c}{2C_0}$ —⑥

$\Rightarrow f_i(t) = f_0 + k_f\, m(t)$ —⑦

where k_f is the resultant freq sensitivity of the modulator.

Disadvantages of direct FM;-

① Carrier freq is not obtained from a highly stable oscillator. ∵ generation of carrier signal is directly affected by modg signal. $m(t)$ directly controls tank ckt of carrier genr.

② Non linearity of the varactor diode produces a freq variation due to the harmonics of the modg signal and ∴ the FM signal is distorted.

- fc is not obtained from a highly stable oscl'r.
- It is ∴ necessary to provide some auxiliary means by which a very stable freq generated by a crystal will be able to control fc
- One method of effecting this control is shown below

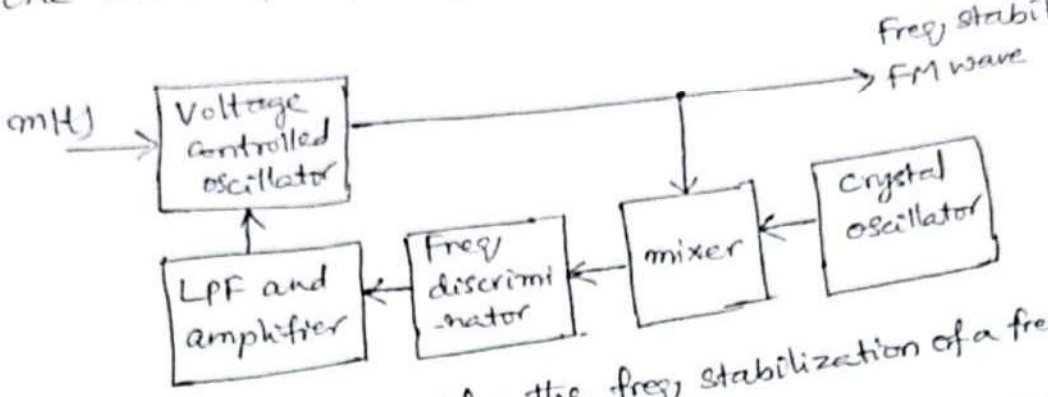


fig A feedback scheme for the freq stabilization of a freq modl'r.

- The o/p of the FM genr is applied to a mixer together with the o/p of a crystal controlled oscl'r, and the difference freq term is extracted.
- The mixer o/p is applied to a freq discr. and then low pass filtered.
- A freq discr is a device whose o/p voltage has an instantaneous amplitude that is proportional to the instant freq of the FM wave applied to its i/p.
- When the FM tx'r has exactly the correct fc, the LPF o/p is zero
- However, the deviations of the tx'r carrier freq from its assigned value will cause the discr-filter combination to develop a dc o/p voltage with a polarity determined by the sense of the tx'r freq drift.
- The dc voltage after suitable amplification, is applied to VCO in such a way as to modify the freq of the oscl'r in a direction that tends to restore the fc to its required value.

Indirect FM:-

- The message signal $m(t)$ is first integrated and then used to phase modulate a crystal controlled oscillator.
+ crystal controlled osc. provides freq. stability.
+ To minimize the distortion in the phase modulator, the max. phase deviation or modulation index β is kept small.

This results in Narrow-Band FM signal.

The narrow band FM signal is next multiplied in frequency by means of a frequency multiplier to produce the desired wide band FM signal.

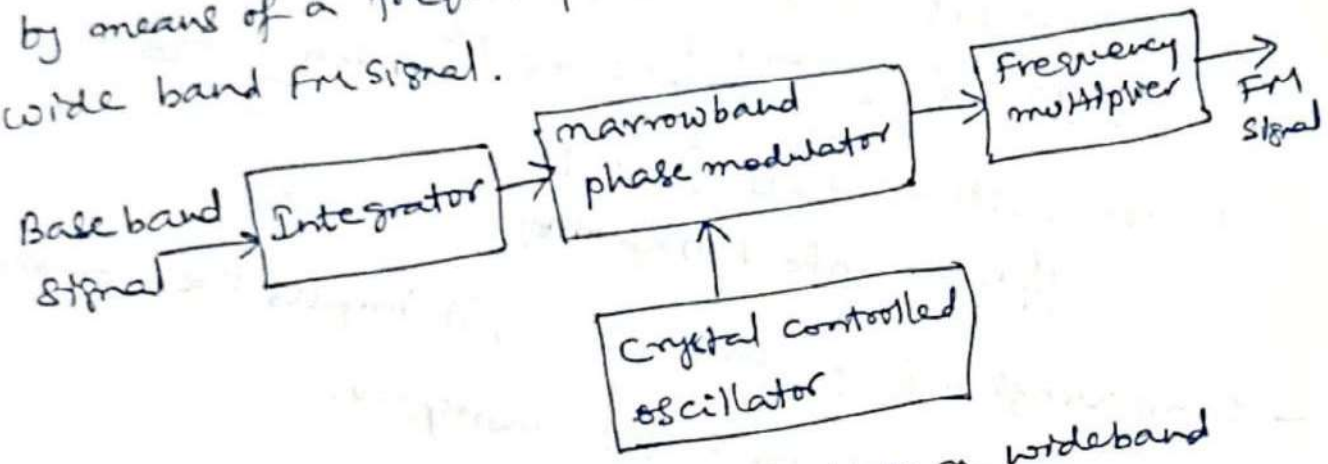


fig Indirect method of generating a wideband FM signal

A freq. multiplier consists of a memoryless non linear device followed by a Band pass filter.

memoryless non linear device = no energy storage elements

The i/p – o/p relation of such a device is

$$v(t) = a_1 s(t) + a_2 s^2(t) + \dots + a_n s^n(t) \quad \text{--- (1)}$$

where $a_1, a_2, \dots a_n$ are the coefficients determined by the operating point of the device. n is the highest order of non linearity.

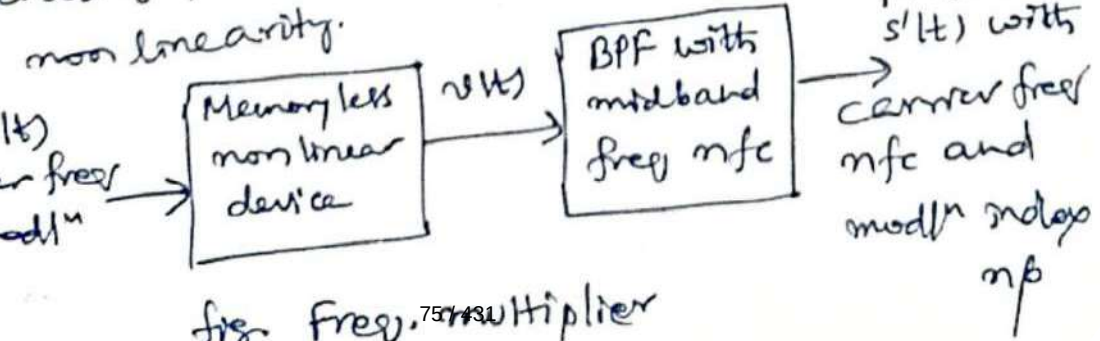


fig Freq. multiplier

- The i/p $s(t)$ is an FM signal defined by

$$s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f \int_0^t m(t)\,dt\right]$$

whose instantaneous freq is $f_i(t) = f_c + k_f m(t)$. — (2)

- The midband freq of the BPF is set equal to nf_c, where f_c is the carrier freq of the incoming FM signal $s(t)$
- The BPF is designed to have a bandwidth equal to n times the transmission B.W. of $s(t)$.
- After BP filtering of the non linear devices o/p $v(t)$, the new FM signal given by

$$s'(t) = A_c' \cos\left[2\pi n f_c t + 2\pi n k_f \int_0^t m(t)\,dt\right]$$ — (3)

whose instantaneous freq is

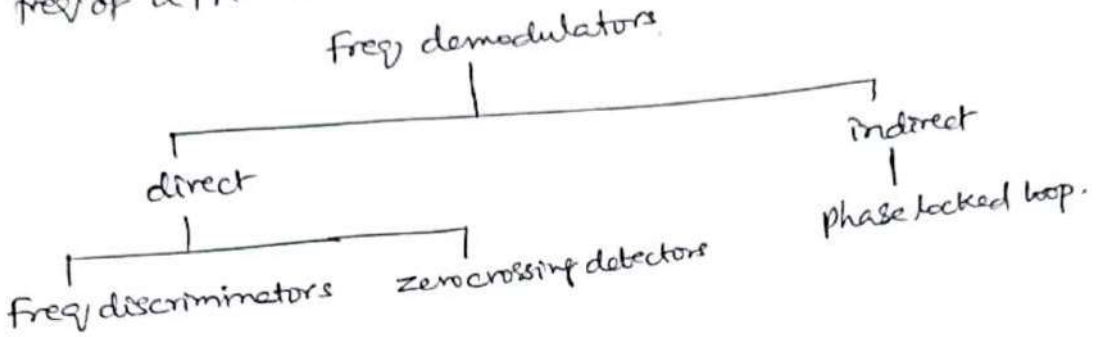
$$f_i'(t) = nf_c + nk_f m(t)$$ — (4)

- comparing eqs (4) and (2), it implies the non linear processing dev acts as a freq multiplier.

Demodulation of FM waves:-

- Freq demodⁿ is the inverse process of FM.
- It enables the original modg wave to be recovered from a fmᵈ wave.

⊙ Freq demodⁿ may be treated as a two port device that produces an o/p signal with amplitude directly proportional to the instant freq of a fmd wave, that is used as the i/p signal.



FM demodulators

The function of a freq to amp changer or FM demodⁿ is to extract original modg voltage from the freq modd voltage.

The detection should be done efficiently and linearly.

The detector ckt should be insensitive to amp changes and should not be too critical in its adjustment and operation.

FM detector performs the detection in two steps

(i) it converts the FMd voltage into corresponding AMd voltage using one or more tuned ckts

(ii) it rectifies this amplitude modd voltage in linear diode det to extract the original modⁿ freq voltage.

Types of FM detector

FM detector or discriminator are of 3 types

(i) Slope detector (or) Single tuned ckt discriminator

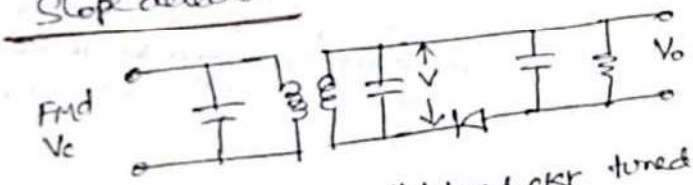
(ii) Balanced slope detector (or) Stagger tuned discriminator.

(iii) Phase diff discriminator.

(a) centre tuned discriminator or foster seely discriminator or phase shift discr

(b) Ratio detector.

Slope detector



ckt

→ It makes use of a tuned ckt tuned to a freq slightly different from the centre freq of the FM carrier, followed by a linear rectifier with filter section.

operation:-

→ As the instantaneous carrier freq varies in accordance with the variation of instantaneous modn voltage, the response of the tuned ckt varies correspondingly as shown in fig below.

→ The amplitude of the V_o then varies in accordance with the variation of carrier freq.

→ This AMd voltage may then be rectified in diode detector to extract the modn freq voltage.

→ If the response of the single tuned ckt is a straight line, then the

demodⁿ freq voltage is identical with the original analog voltage.

→ But the response curve is not linear and hence the detected Vo gets distorted i.e. it contains harmonic terms.

→ This is one draw back.

→ another draw back is: It does not reject amp variations.

→ Thus if the applied V_f is doubled, the dc o/p and the o/p voltage also gets doubled.

inefficient, It is linear over a limited range difficult to adjust

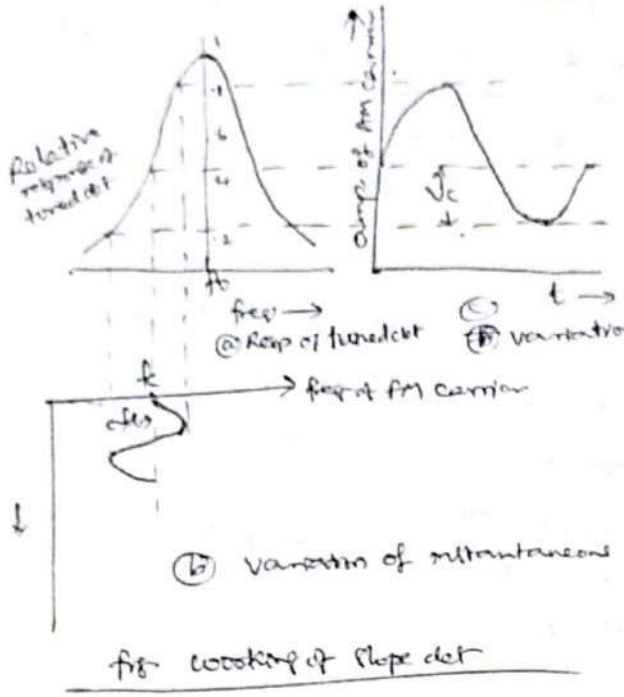


fig working of Slope det

Balanced Slope detector (Travis detector), (Triple tuned discriminator)

Two slope detectors are connected back to back.

→ It is an extension of the slope det. and eliminates the distortion. (present in the latter)

→ It is fed from an i/p tuned ckt, tuned to the carrier centre freq fc.

→ It utilizes two tuned ckts A and B, each associated with its own rectifier diode and o/p load ckt.

→ The tuned ckt A is associated with rectifier diode DA and o/p load ckt consisting of RA and CA (similar to T.ckt B)

RA = RB; CA = CB.

→ ckt A is tuned to a freq f_1 which is greater than the carrier freq fc by a small freq interval Δf.

→ ckt B is tuned to a freq f_2 below the carrier centre freq by Δf.

Let V_A and V_B the voltages response of tuned ckts A and B respectively.

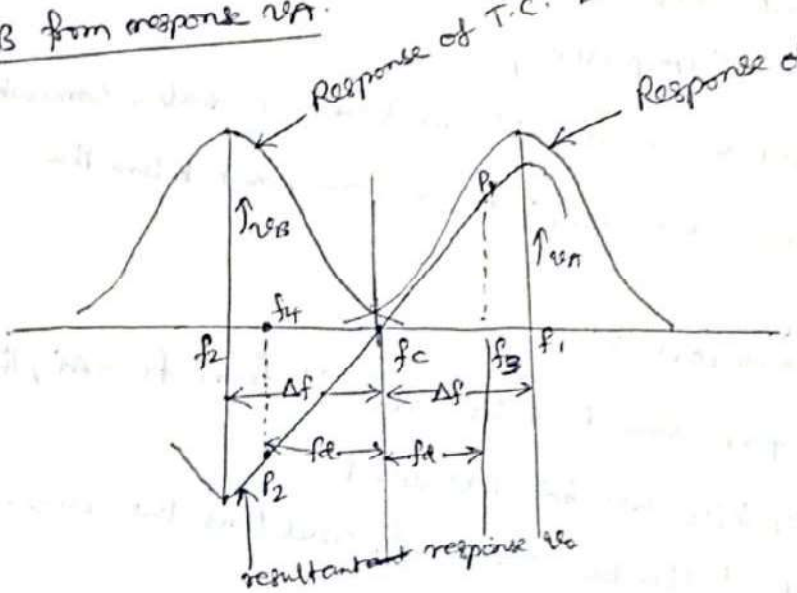
→ These voltages are rectified by diodes D_A & D_B respectively and the the RF components of the rectified currents are by passed by cap C_A and C_B respectively.

→ The voltage of v_A and v_B developed across the load resistors R_A and R_B respectively are then nothing but the variations in amplitudes of voltages VA and VB. → Then the response of the tuned ckt plotted against freq ~~resp~~ represents the detected o/p voltage v_A as shown in fig

→ ||||y the response of T.C. B plotted against freq represents the detected o/p voltage v_B.

→ The ckt is so arranged that the detected voltage v_A and v_B oppose each other so that the resultant o/p voltage $v_o = v_A - v_B$.

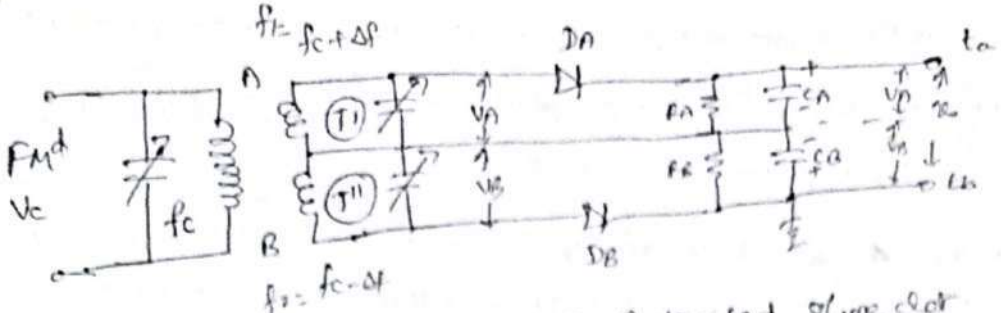
→ The resultant response v_o is also plotted in fig. by subtracting the response v_B from response v_A.



$f_4 \rightarrow f_3$ useful range
$f_1 = f_c + \Delta f$
$f_2 = f_c - \Delta f$

→ It is seen that the curve of the resultant v_o is almost a straight line in the range $P_1 P_2$ and it cuts the freq axis at the carrier centre freq fc.

→ If now a FM voltage of carrier centre freq fc and freq deviation equal to or less than fd ($= f_3 - f_c = f_c - f_4$) is applied to the i/p of this discriminator then the rectified o/p voltage is proportional to the instantaneous freq variation and hence is proportional to the instantaneous value of the modg voltage.



Basic ckt of Balanced Slope detector

Drawbacks

1. tricky to align
2. amp limiting is not provided
3. Linearity is not good enough.

Operation (Kennedy)

→ Let fc be the IF to which the ckt is tuned, and let fc+Δf and fc−Δf be the resonant freqs of the upper secondary and lower secondary ckts T' and T'' respectively.

→ When fin is instantaneously = fc ; voltage across T' will have a value somewhat less than the maximum available, ∵ fc is some what below the resonant of T'.

→ A similar condition exists across T''.

→ In fact, ∵ fc is just far from fc+Δf as it is from fc−Δf, the voltages applied to D1 & D2 will be identical.

→ The dc o/p voltages will also be identical, and thus the detector o/p will be zero.

→ Consider the inst. freq fin = fc+Δf ; ∵ T' is tuned to this freq, o/p of D1 is quite large. o/p of D2 will be very small, ∵ fc+Δf is quite a long away from fc−Δf

→ Similarly if fin (inst) = fc−Δf o/p of D2 will be a large −ve voltage, and that of D1 is a small +ve voltage.

→ If the fin goes outside the range, the o/p will fall.

→ When the inst. freq is b/n these 2 extremes the o/p will have some intermediate value.

Phase discriminator (or) Centre tuned discriminator (or) Foster-Seely discriminator.

→ It is possible to obtain the same S shaped response curve from a ckt in which the primary and secondary windings are both tuned to the centre freq. of the incoming signal.

→ This is desirable because it greatly simplifies alignment (both tuned ckts tuned to fc. and also because the process yields far better linearity than slope detection. Better linearity, ∵ ckt relies less on freq response and more on pri-sec phase relation.

drawback:- It does not provide amplitude limiting

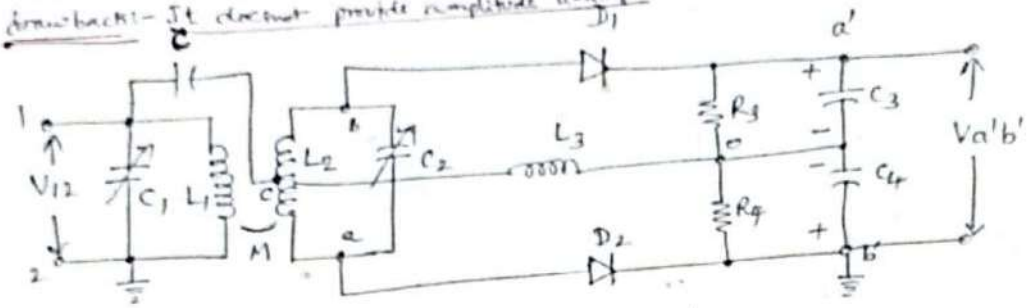


fig. phase discriminator

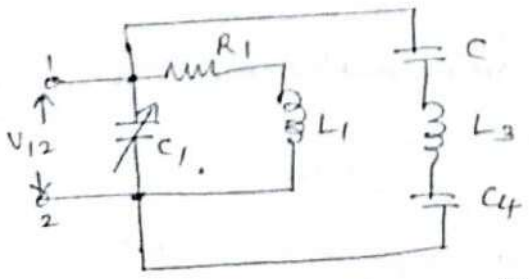
→ The diode and load arrangement is same as in balanced slope detector; because such an arrangement is satisfactory

→ The resistances forming the load are made much larger than the capacitive reactances.

→ The ckt composed of C, L_3 and C_4 is effectively placed across the primary winding

→ The voltage applied to each diode = primary voltage + corresponding half secondary voltage. $\left(\overbrace{V_{12}+(V_d)}^{\text{pri-sec}} + V_{ca}\right)$

fig. Discriminator primary voltage



The voltage across L_3; $V_L = \frac{Z_{L_3}}{Z_c + Z_{C_4} + Z_{L_3}} \cdot V_{12}$

$$= \frac{j\omega L_3}{j\omega L_3 - j\left(\frac{1}{\omega c} + \frac{1}{\omega c_4}\right)} \cdot V_{12}$$

L_3 is an RF choke and is purposely given a large reactance

$\therefore X_{L_3} > X_C$ or X_{C_4}

$\Rightarrow V_L \approx V_{12}$

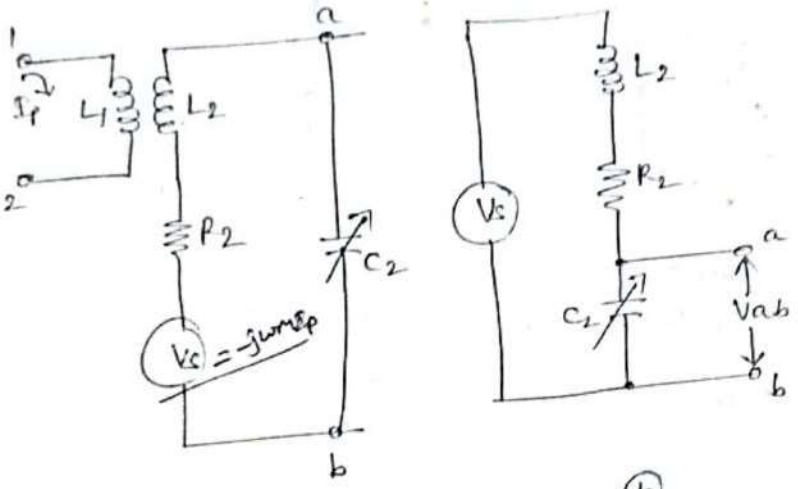
Voltage across the RF choke = applied primary voltage.

I_p is given by $\Rightarrow I_p = \frac{V_{12}}{j\omega L_1}$

[A voltage is induced in series with the secondary as a result of I_p]

$V_s = \pm j\omega M I_p$ (sign depends on the direction of winding)

It is simpler here to take the connection giving -ve mutual inductance.



(a) (b)

fig: Discriminator secondary ckt & voltages (a) primary, secondary relations (b) secondary redrawn.

$$V_s = -j\omega M I_p = -\frac{j\omega M V_{12}}{j\omega L_1} = -\frac{M}{L_1} V_{12}$$

The voltage across the secondary winding, V_{ab} is given by

$$V_{ab} = V_s \cdot \frac{Z_{C_2}}{Z_{C_2} + Z_{L_2} + R_2} = \frac{-jX_{C_2}(-V_{12} M/L_1)}{R_2 + j(X_{L_2} - X_{C_2})}$$

$$V_{ab} = \frac{jM}{L_1} \frac{V_{12} X_{C_2}}{R_2 + jX_2}$$ where $X_2 = X_{L_2} - X_{C_2}$. and may be +ve, -ve or zero. depending on the freq.

The total voltages applied to D_1 and D_2 are V_{ao} and V_{bo} respectively.

$$V_{ao} = V_{ac} + V_L = \frac{1}{2} V_{ab} + V_{12}$$

$$V_{bo} = V_{bc} + V_L = -V_{ac} + V_L = -\frac{1}{2} V_{ab} + V_{12}$$

As predicted the voltage applied to each diode is sum of the primary voltage and the corresponding half secondary voltage.

The dc o/p voltages cannot be calculated exactly because the diode drop is unknown.

However, we know that each will be proportional to the peak value of the RF voltage applied to the respective diode.

$V_{a'b'} = V_{ab} - V_{bb} \propto V_{ao} - V_{bo}$

VP&VS
fin=fc
90° condition
<90° ... fin>fc
>90°-time ce

(a) When the input freq fin is instantaneously equal to fc.

$X_2 \Rightarrow 0$ (resonance i.e. $X_{L_2} = X_{C_2}$)

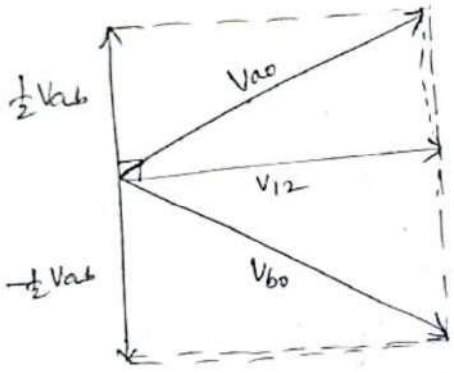
$$\Rightarrow V_{ab} = \frac{jM}{L_1} \frac{V_{12} X_{C_2}}{R_2} = \frac{V_{12} X_{C_2} M \angle 90^\circ}{R_2 L_1}$$

it follows that the secondary voltage V_{ab} leads the applied primary voltage by 90°.

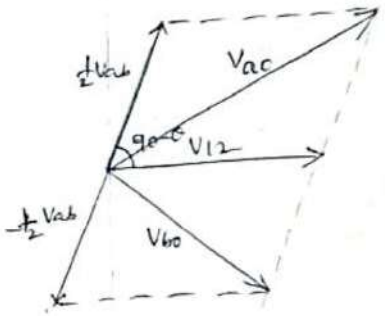
Thus $\frac{1}{2} V_{ab}$ will lead V_{12} by 90°,

and $-\frac{1}{2} V_{ab}$ will lag V_{12} by 90°.

It is now possible to add diode input voltages vectorially.



(a) fin = fc



(b) fin > fc

It is seen that ∵ $V_{ao} = V_{bo}$, the discriminator o/p is zero, when fin = fc ; i.e. no o/p for no modulation.

(b) When fin is greater than fc.

Now $X_{L_2} > X_{C_2} \Rightarrow X_2$ is +ve.

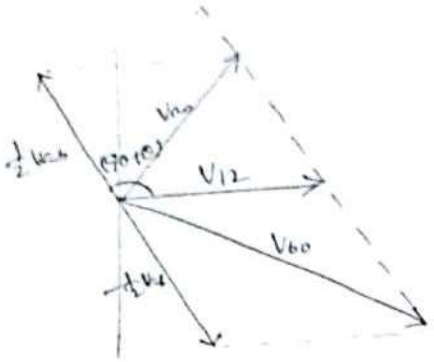
$$V_{ab} = \frac{jM}{L_1} \frac{V_{12} X_{C_2}}{R_2 + jX_2} = \frac{V_{12} X_{C_2} M \angle 90^\circ}{L_1 |Z_2| \angle \theta^\circ} = \frac{V_{12} X_{C_2} M}{L_1 |Z_2|} \angle (90 - \theta)^\circ.$$

$\Rightarrow V_{ab}$ leads V_{12} by less than 90° so that $-\frac{1}{2} V_{ab}$ must lag V_{12} by more than 90°

It is apparent from the vector diagram that V_{ao} is now greater than V_{bo} the discriminator o/p is +ve.

iii) when $f_{in} < f_c$; X_2 will be -ve, and the angle of Z_2 will also be -ve.

Thus V_{ab} will lead V_{12} by more than 90°, and V_{cc} will be smaller than V_{bb} and the o/p voltage $V_{a'b'} = -ve$.



© $f_{in} < f_c$

fig. phase discriminator phasor diagrams

If the freq. response is plotted for the phase discriminator, it will follow the required 'S' shape as shown in fig.

As f_{in} moves further and further away from the center, the disparity bet the 2 diode i/p voltages becomes greater and greater.

The o/p of discriminator will increase up to the limits of the useful range.

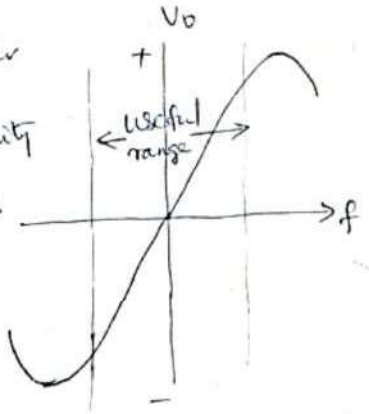
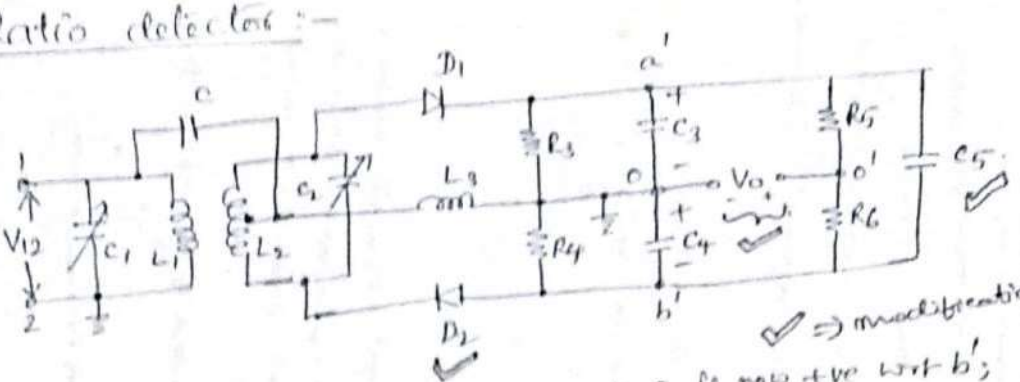


fig. discriminator response.

The limits correspond roughly to the half power points of the discriminator tuned transformer.

Beyond these points, the diode i/p voltages are reduced because of the freq response of the transformer, so that the overall o/p falls.

Ratio detector :-



⇒ modifications

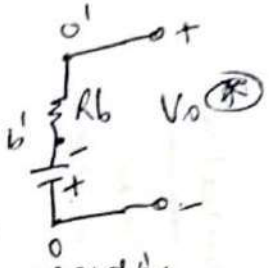
operation :- with diode D_2 reversed, o is now +ve wrt b'; so that Va'b' is now a sum voltage, rather than difference voltage

In practice $R_5 = R_6$

$$V_o = V_{b'o'} - V_{b'o} = \frac{V_{a'b'}}{2} - V_{b'o} = \frac{V_{a'o} + V_{b'o}}{2} - V_{b'o} = \frac{V_{a'o} - V_{b'o}}{2}$$

$V_{o'b'} - V_{b'o'} + V_{a'o}$

(*) V_o ∝ the diff. bet the individual o/p voltages.



Amplitude limiting

If the i/p voltage V_{12} is const and has been so for some time, C_5 has been able to charge up to the potential existing between a' and b'. ∴ this is a dc voltage if V_{12} is const, there will be no current either flowing in to charge the capacitor or flowing out to discharge it. (i.e. Zi of C_5 is ∞)

The total load impedance for the two diodes is ∴ the sum of R_3 and R_4.

∵ $R_3, R_4 < R_5, R_6$.

If V_{12} tries to increase, C_5 will tend to oppose any rise in V_b.

This operation not only depends on large time constant, but there is another reason for it.

As soon as the input voltage rises, extra diode current flows in to C_5 charging it. The voltage Va'b' remains const ∵ it is not possible for the voltage across the cap to change instantaneously

The situation now is that the current in the diodes load has increased. But the voltage across the load has not changed. ⇒ load impedance has decreased.

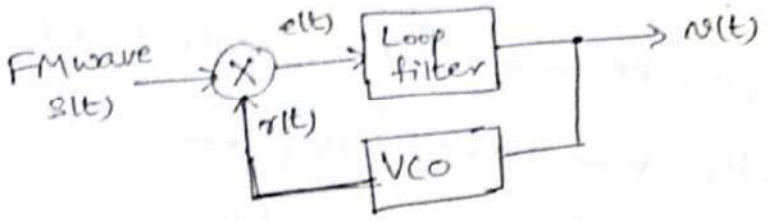
⇒ The secondary of ratio detector transformer is more heavily damped.

Q falls, gain of amplifier driving the Ratio detector also reduces

This counteracts the initial rise in input voltage.

Phase Locked Loop (PLL):-

- The PLL is a -ve feedback sys. that consists of 3 major components (i) multiplier (ii) loop filter (iii) VCO. connected together in the form of a feed back loop as shown in fig ①



fig① Phase locked loop.

- The VCO is a sine wave generator whose freq is det^d by a voltage applied to it from an external source.
- In effect, any freq modl^r may serve as a VCO.

→ We assume that initially we have adjusted the VCO so that when the control voltage is zero, two conditions are satisfied.

(1) the freq of the VCO is precisely ~~at~~ set at the unmodd carrier freq fc, and

(2) the VCO o/p has a 90° phase shift w.r.t. the unmodd carrier wave.

- Suppose that the i/p signal applied to the PLL is an FM wave defined by $s(t) = A_c \sin[2\pi f_c t + \phi_1(t)]$ —①

where Ac is the carrier amplitude.

- With a modg wave m(t), we have $\phi_1(t) = 2\pi K_f \int_0^t m(t)\,dt$ —②

where Kf is the freq sensitivity of the freq modulator.

- Let the VCO o/p be defined by $r(t) = A_v \cos[2\pi f_c t + \phi_2(t)]$ —③

where Av is the amplitude.

- With a control voltage v(t) applied to the VCO i/p, we have

$$\phi_2(t) = 2\pi K_v \int_0^t v(t)\,dt \quad —④$$

where Kv is the freq sensitivity of the VCO, measured in hertz per volt.

→ The incoming FM wave s(t) and the VCO o/p r(t) are applied to the multiplier, producing two components:

(1) A HF component represented by $k_m A_c A_v \sin[4\pi f_c t + \phi_1(t) + \phi_2(t)]$; $\left[\sin A \cos B = \frac{1}{2}[\sin(A+B) + \sin(A-B)]\right]$

(2) A LF component represented by $K_m A_c A_v \sin[\phi_1(t) - \phi_2(t)]$

where K_m is the multiplier gain measured in Volt^{-1}.

- The HF component is eliminated by the low pass action of the filter and VCO.

$\Rightarrow$ ∴ discarding the HF component, the i/p to the loop filter is given by $e(t) = K_m A_c A_v \sin[\phi_e(t)]$ — (5)

where $\phi_e(t)$ is the phase error defined by

$$\phi_e(t) = \phi_1(t) - \phi_2(t) = \phi_1(t) - 2\pi K_v \int_0^t v(t)\,dt \quad -(6)$$

- The loop filter operates on its input $e(t)$ to produce the o/p

$$v(t) = \int_{-\infty}^{\infty} e(\tau)\, h(t-\tau)\, d\tau \quad -(7)$$

where $h(t)$ is the impulse response of the filter.

- Using eqs (5) through (7) to relate $\phi_e(t)$ and $\phi_1(t)$ and differentiating wrt time, we obtain

$$\frac{d\phi_e(t)}{dt} = \frac{d\phi_1(t)}{dt} - 2\pi K_o \int_{-\infty}^{\infty} \sin[\phi_e(\tau)]\, h(t-\tau)\, d\tau \quad -(8)$$

where K_o is a loop parameter defined by $K_o = K_m K_v A_c A_v$ — (9)

- Eq (8) suggests the representation or model of fig 2(a) shown below.
- In this model we have also included the relationship between $v(t)$ and $e(t)$ as represented by eqs (5) & (7)
- we see that the B.D. of the model resembles fig (1) PLL.
- The multiplier is replaced by a subtractor and a sinusoidal non linearity, and the VCO by an integrator.
- The loop parameter K_o plays an important role in the operation of a PLL.
- It has the dimensions of freq, this follows from eq (8), where we observe that the amplitudes A_c and A_m are both measured in volts. and the multiplier gain K_m is measured in Volt^{-1}.

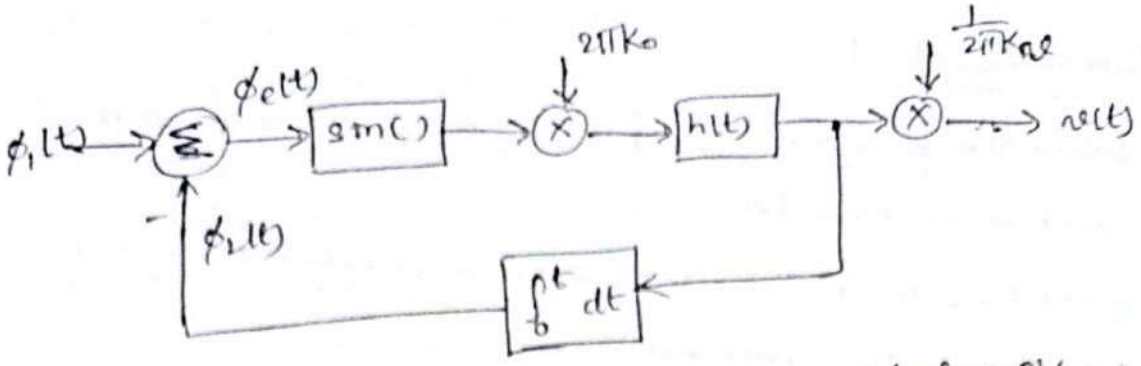


Fig 2 @ Non linear model of a PLL.

(*)

$$\frac{d\phi_e(t)}{dt} = \frac{d\phi_1(t)}{dt} - 2\pi K_{vc} \frac{d}{dt} \int_0^t v(t)\, dt$$

$$= \frac{d\phi_1(t)}{dt} - 2\pi K_{vc}\, v(t) = \frac{d\phi_1(t)}{dt} - 2\pi K_{vc} \int_{-\infty}^{\infty} e(\tau) h(t-\tau)\, d\tau$$

$$= \frac{d\phi_1(t)}{dt} - 2\pi K_{vc} \int_{-\infty}^{\infty} K_m A_c A_{vc} \sin[\phi_e(t)]\, h(t-\tau)\, d\tau$$

$$= \frac{d\phi_1(t)}{dt} - 2\pi K_{vc} K_m A_c A_{vc} \int_{-\infty}^{\infty} \sin[\phi_e(\tau)]\, h(t-\tau)\, d\tau$$

$$\boxed{\frac{d\phi_e(t)}{dt} = \frac{d\phi_1(t)}{dt} - 2\pi K_o \int_{-\infty}^{\infty} \sin[\phi_e(\tau)]\, h(t-\tau)\, d\tau} \quad -(8)$$

where $K_o = K_m K_{vc} A_c A_{vc}$; is a loop parameter. $-(1)$

Linearized Model:

When the phase error $\phi_e(t)$ is zero, the phase locked loop is said to be in phase lock.

When $\phi_e(t)$ is at all times small compared with one radian, we may use the approximation

$$\sin[\phi_e(t)] \cong \phi_e(t) \quad \text{— (10)}$$

which is accurate to within 4% for $\phi_e(t) < 0.5$ rad. In this case the loop is said to be near phase lock and the sinusoidal nonlinearity of fig (2) may be disregarded. Thus we may represent the PLL by the linearized model shown in fig 2(b).

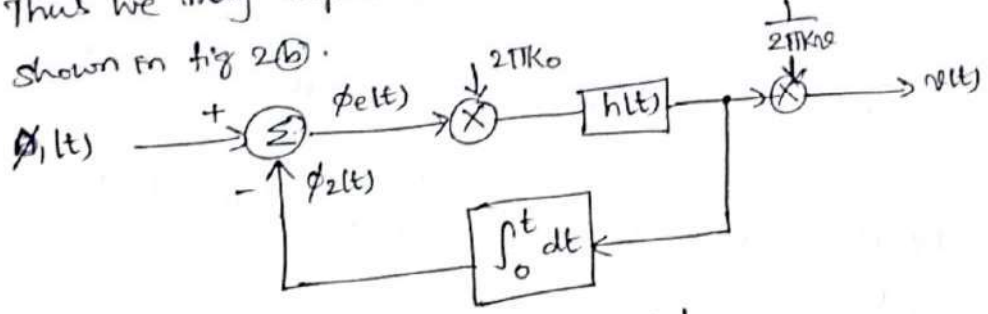


fig (2b) Linearized model.

According to this model, the phase error $\phi_e(t)$ is related to the i/p phase $\phi_1(t)$ by the integro differential equation! [Ref eq 8, b] (*)

$$\frac{d\phi_e(t)}{dt} + 2\pi K_0 \int_{-\infty}^{\infty} \phi_e(\tau)\, h(t-\tau)\, d\tau = \frac{d\phi_1(t)}{dt} \quad \text{— (11)}$$

Transforming (11) into freq domain →

$$j\omega \phi_e(\omega) + 2\pi K_0 [\phi_e(\omega) H(\omega)] = j\omega \phi_1(\omega)$$

$$\phi_e(\omega)\left[j\omega + 2\pi K_0 H(\omega)\right] = j\omega \phi_1(\omega)$$

$$\phi_e(\omega) = \frac{j\omega \phi_1(\omega)}{j\omega + 2\pi K_0 H(\omega)} = \frac{\phi_1(\omega)}{1 + \frac{2\pi K_0 H(\omega)}{j\omega}}$$

$$\phi_e(\omega) = \frac{\phi_1(\omega)}{1 + \frac{2\pi K_0}{2\pi f}\frac{H(\omega)}{j}} = \frac{\phi_1(\omega)}{1 + K_0 \frac{H(\omega)}{jf}}$$

$$\rightarrow \boxed{\phi_e(f) = \frac{\phi_1(f)}{1 + K_0 \frac{H(f)}{jf}} = \frac{1}{1 + K_0 \frac{H(f)}{jf}} \cdot \phi_1(f) = \frac{1}{1 + L(f)} \phi_1(f)} \quad \text{— (12)}$$

where $L(f) = \frac{K_0 H(f)}{jf}$ — (13) where $H(f)$ is the transfer function of the loop filter.

$L(f)$ is called open loop transfer fn. of PLL.

Suppose that for all values of f inside the base band we make the magnitude of $L(f)$ very large compared with unity, then from (12) we find that $\phi_e(f)$ approaches zero.

i.e; the phase of the VCO becomes asymptotically equal to the phase of the incoming wave, and phase lock is thereby established.

from fig 2(b) $v(t) = \frac{1}{2\pi K_v} 2\pi K_o \int_{-\infty}^{\infty} \phi_e(\tau') h(t-\tau') d\tau'$

$$V(f) = \frac{K_o}{K_v} H(f) \phi_e(f) \quad \text{— (14)}$$

$$V(f) = \frac{K_o}{K_v} \frac{jf}{K_o} L(f) \phi_e(f) = \frac{jf}{K_v} L(f) \phi_e(f) \quad \text{— (15)}$$

$$V(f) = jf/K_v \, L(f) \left\{ \frac{1}{1+L(f)} \phi_1(f) \right\} \quad \because \text{from (12)}$$

$$V(f) = \frac{(jf/K_v) L(f)}{1+L(f)} \phi_1(f) \quad \text{— (16)}$$

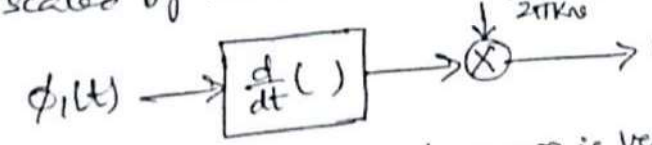
Again, when we make $|L(f)| \gg 1$ we may approximate (16) as

$$V(f) = (jf/K_v) \phi_1(f) \quad \text{— (17)}$$

$$= \frac{1}{K_v} jf \phi_1(f) \cdot \frac{2\pi}{2\pi} = \frac{1}{2\pi K_v} (j2\pi f \phi_1(f))$$

$$\Rightarrow \boxed{v(t) = \frac{1}{2\pi K_v} \frac{d}{dt} \phi_1(t)} \quad \text{— (18)}$$

Thus, provided the magnitude of $L(f)$ is very large for all freqs of interest, the PLL may be modeled as a differentiator with its output scaled by the factor $1/2\pi K_v$ as shown in fig 2(c).



2(c) simplified model when the loop gain is very large compared to 1.

(A) fig 2(c) provides the basis of using PLL as a freq demodulator. When the i/p signal is an FM wave like $s(t) = A_c \sin[2\pi f_c t + \phi_1(t)]$, the phase $\phi_1(t) = 2\pi K_f \int_0^t m(t)\, dt$, where $m(t)$ is modg wave.

$$\therefore v(t) = \frac{1}{2\pi K_v} \frac{d}{dt} \left(2\pi K_f \int_0^t m(t)\, dt \right) = \frac{K_f}{K_v} m(t) \quad \text{— (19)}$$

That is, the o/p $v_2(t)$ of the PLL is approximately the same (except for the scale factor K_f/K_{VS}) as the original message signal $m(t)$, and the freq demodⁿ is accomplished. (performed)

Pre emphasis :-

- In FM, the noise has a greater effect on the higher modulating frequencies.
- This effect can be reduced by increasing the value of modulation index (mf) for higher modulating frequencies (fm)
- This can be done by increasing the deviation Δf and Δf can be increased by increasing the amplitude of modulating signal at higher modulating frequencies.
- Thus if we boost the amplitude of higher frequency modulating signals artificially then it will be possible to improve the noise immunity at higher modulating frequencies.
- The artificial boosting of higher modulating frequencies is called as pre emphasis.
- (Boosting of higher frequency modulating signal) This is achieved by using the pre emphasis circuit.

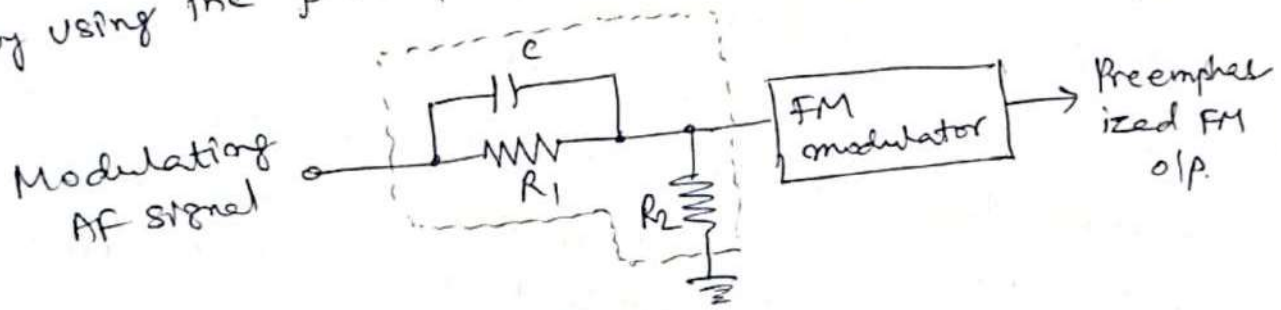


fig. Pre emphasis ckt

- Pre emphasis ckt is basically a HPF.
- The modulating AF signal is passed through a high pass RC filter, before applying to the FM modulator.
- As fm increases, reactance of C decreases and modulating voltage applied to FM modulator goes on increasing.
- The boosting is done according to this pre arranged curve.

fig. Pre emphasis characteristics.

- The amount of preemphasis in FM tx^n and sound tx^n in TV has been standardized at 75 μs.
- Pre emphasis is carried out at the transmitter.
- Frequency for the RC high pass $m/w = 2122$ Hz.

De emphasis:

- The artificial boosting given to the higher modulating freq in the process of preemphasis is nullified or compensated at the receiver by a process called de-emphasis.
- Artificially boosted Hf signals are brought to their original amplitude using de emphasis ckt.
- 75 μs deemphasis ckt is standard. It is a LPF.
- 75μs deemphasis corresponds to a freq response curve that is 3 dB down at a freq whose RC time constant is 75 μs.
- Demodulated FM is applied to deemphasis ckt, with increase in fm the reactance of C goes on decreasing and the o/p of deemphasis ckt will also reduce.

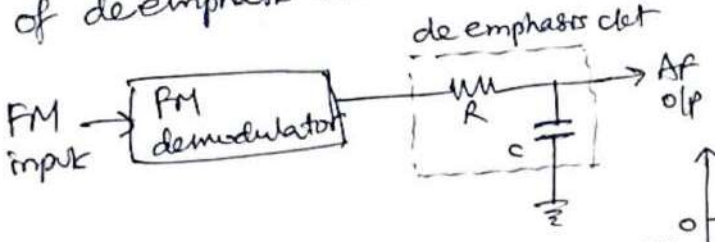


fig. deemphasis ckt.

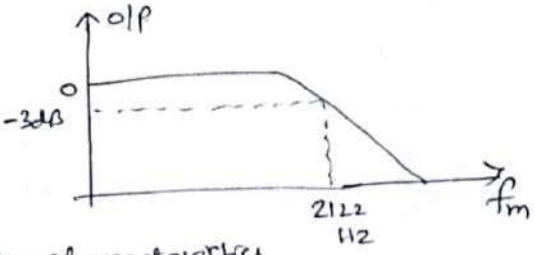


fig. characteristics

Comparison of FM and AM:-

	FM	AM
1.	Amplitude of FM wave is constant. It is independent of modl^n index.	Amplitude of AM wave is not constant It depends on modulation index
2.	The no. of sidebands having significant amplitudes depends on modulation index	The no. of side bands in AM is constant and equal to 2.
3.	Bandwidth of FM $= 2(\Delta f + f_m)$ BW depends on modulation index.	Bandwidth of AM $= 2 f_m$. It is independent of modulation index
4.	Bandwidth is large	Bandwidth is less than FM BW.
5.	Transmitted power remains constant. It is independent of modulation index.	Transmitted power is not constant It depends on modulation index.
6.	All the transmitted power is useful.	Carrier power and one sideband power are useless.
7.	The information is contained in the frequency variation of the carrier.	The information is contained in the amplitude variation of the carrier.
8.	The value of modulation index may be less than 1 or greater than 1.	The value of modulation index is always between 0 and 1.
9.	The amplitudes of carrier and sidebands vary with the modulation index and can be calculated with Bessel functions.	The amplitudes of the sidebands depend on the modulation index, and is always less than the amplitude of the carrier.
10.	Space wave propagation is used for FM transmission. So, radius of transmission is limited to line of sight.	Ground wave and sky wave propagations are used for AM transmission. $\therefore$ Larger area is covered
11.	Hence, it is possible to operate several transmitters on same frequency.	It is not possible to operate several tx's on the same freq.
12.	FM Tx^n and Rx^n equipment are more complex.	AM equipment are less complex.
13.	FM receivers are immune to noise.	AM receivers are not immune to noise.

A 25MHz carrier is modulated by a 400Hz audio sine wave. If the carrier voltage is 4V. and the maximum deviation is 10KHz. Write the eq of this modulated wave for (i) FM and (ii) PM. If the modulating freq is now changed to 2KHz, all else remaining const, Give the new eq for (iii) FM & (iv) PM.

Eq. for FM $e = E_c \cos(\omega_c t + m_f \sin \omega_m t)$

$= 4 \cos\left(25M \times 2\pi t + \left(\frac{10K}{400}\right) \sin(2\pi 400)t\right)$

$e_{FM} = 4 \cos(1.57 \times 10^8 t + 25 \sin 2514t) = e_{PM}$.

$\Delta f = k_f A_m$.

f_m : 400 → 2KHz 5 times increased. $\frac{\Delta f}{f_m} = \frac{10K}{2K} = 5$

$\beta = \frac{\Delta f}{f_m}$ ⇒ decreases 5 times. $2\pi f_m = 2\pi \times 2K = 4\pi \times 10^3$

β_{PM} = const. $e_{FM} = 4 \cos(1.57 \times 10^8 t + 5 \sin 4\pi \times 10^3 t)$

∵ It doesnot depend on f_m. $e_{PM} = 4 \cos(1.57 \times 10^8 t + 25 \sin 2514t)$

A modulating signal $5 \cos(2\pi 15 \times 10^3)t$, angle modulates a carrier $A \cos \omega_c t$. (i) find the modulation index & BW for both FM & PM. (ii) Det the change in BW & modln index for both FM & PM, if the modg signal freq reduces to 5KHz.

i) $V_m = 5V$ $f_m = 15$ KHz. Assume $k_f = k_p = 15$ KHz/V.

for FM $\Delta f = k_f \cdot V_m = 15KHz/V \times 5V = 75$ KHz

$m_f = \frac{\Delta f}{f_m} = \frac{75K}{15K} = 5$.

$BW = 2(\Delta f + f_m) = 2(75K + 15K) = 180KHz$.

for PM $\Delta f = k_p V_m f_m = 15KHz/V \times 5 \times 15K = 1125MHz$.

$BW = 2(\Delta f + f_m) = 2(1125M + 15K) \cong 2250MHz$.

$m_p = k_p \cdot V_m = 15KHz/V \cdot 5V = 75000$.

ii) $f_m = 5$KHz. $\Delta f = k_f V_m = 75KHz$ $m_f = \frac{\Delta f}{f_m} = \frac{75K}{5K} = 15$

$BW = 2(75K + 5K) = 160KHz$.

PM: $\Delta f = k_p V_m f_m = 15KHz/V \times 5 \times 5K = 375$ MHz.

$BW = 2(\Delta f + f_m) = 2(375M + 5K) \cong 750$ MHz.

$m_p = k_p V_m = 75000$.

Unit 4 : Transmitters & Receivers

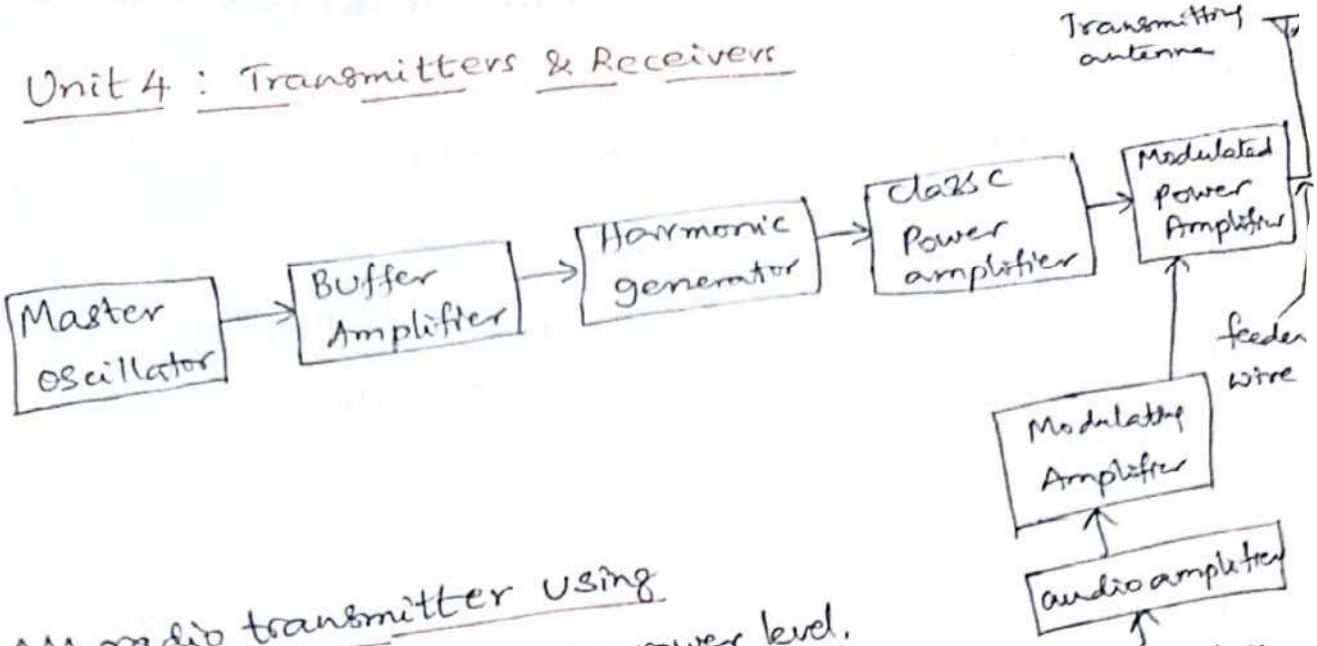


fig. AM radio transmitter using modulation at high carrier power level.

(or)

fig. AM transmitter (high level modulation)

Master oscillator :- It generates oscillations of desired frequency with high constancy of freq.

- The generated freq. is required to remain constant within close limits inspite of variations in the supply voltage, ambient temp, temp. of components of load.
- Further freq. variations with time and age of the tube or transistor are to be avoided.

Buffer Amplifier or Isolating amplifier :-

If the master oscillator directly drives a harmonic generator, then power is drawn from the M.O. This results in loading of M.O, which inturn causes freq. variation.

∴ B.A is placed b/n M.O. and H.G.

Harmonic generator :-

M.O. generates a voltage at a freq. which is a submulti-ple of carrier frequency.

H.G.
(It) consists of freq. doublers & triplers.
H.G. increases the carrier freq.

Class C power amplifier:

RF voltage generated by M.O. has very small power. The power level is rised to high value in a chain of class C amplifiers.

Modulated amplifier:-

Modulation of carrier by audio modulating voltage from modulating amplifier is done in this stage.

Modulating amplifier:-

Modulating signal is further amplified in this stage.

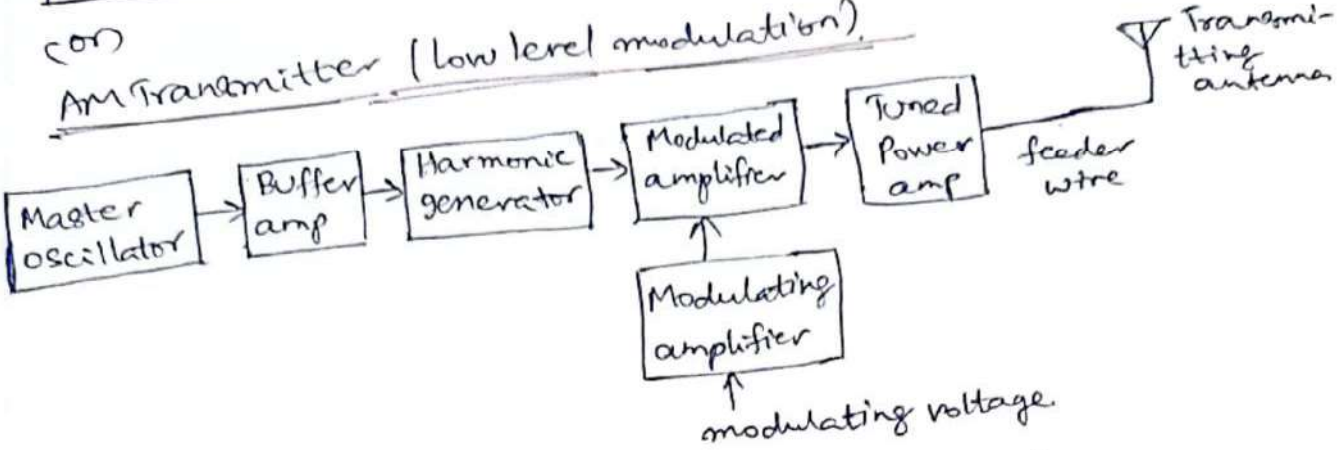
High level modln:-

The carrier voltage is modulated at the highest power level. Hence it is called high power level modulation system.

AM radio transmitter using modulation at low carrier power level.

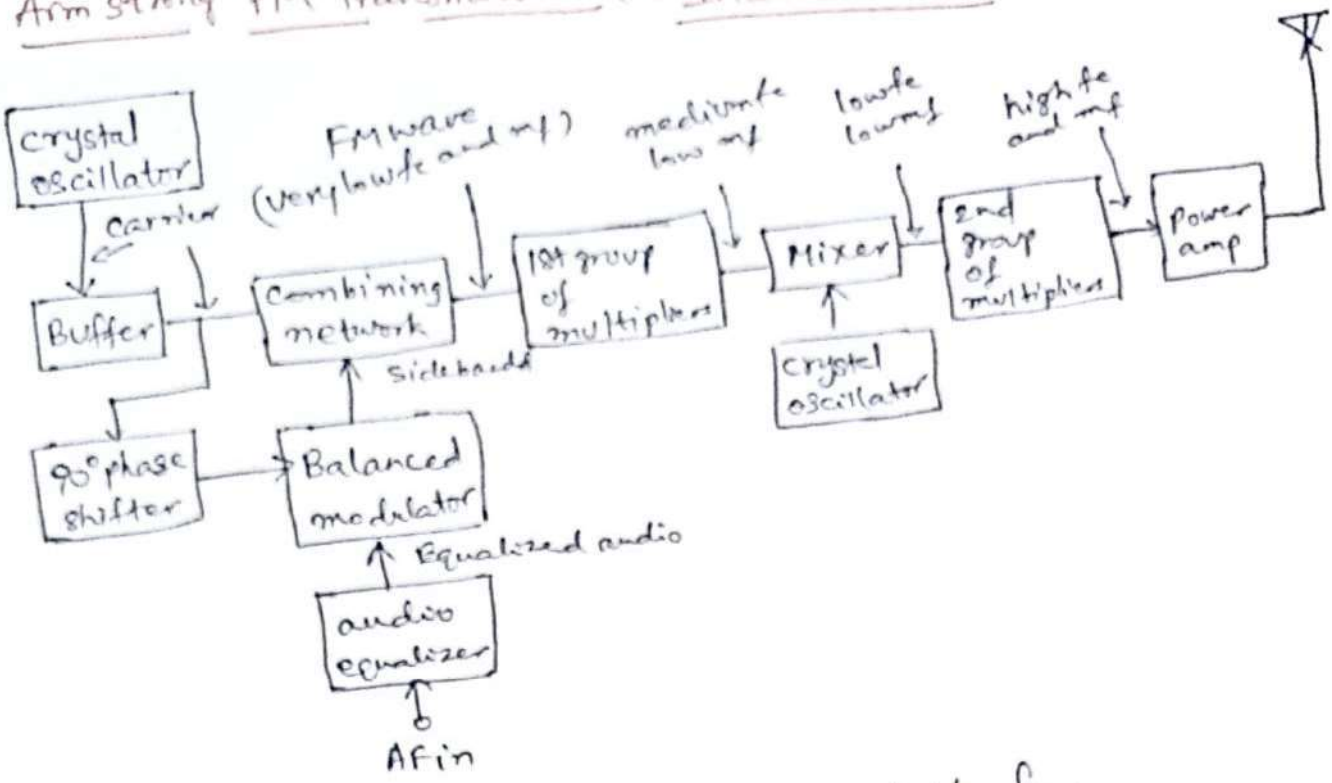
(or)

AM Transmitter (low level modulation).



- The carrier is modulated at low power level.
- Carrier power is subsequently rised to the desired level in tuned power amplifiers.

Armstrong FM Transmitter (or) Indirect FM Tx.



- Crystal oscillator is used to provide stable fc.
- Carrier voltage is fed to combining n/w through buffer.
- Phase shifted carrier is fed to balanced modulator.
- Equalized audio signal is also fed to balanced modulator.
- The carrier and side bands are combined in the combining network.
- The o/p of combining n/w is an FM wave.
- The phase deviation of this FM wave is very low, to avoid distortion.
- This FM wave is fed to 1st group of multipliers to increase center freq and deviation.
- At this stage fc is medium and Δf is low.
- The o/p of 1st group of multipliers is mixed with crystal oscillator freq to reduce/change center freq only.
- Δf remains same at this stage.
- 2nd group of multipliers rise the fc and Δf to the desired and maximum values respectively.
- After power amplification in the next stage, the o/p is fed to the transmitting antenna.

RADIO RECEIVERS

Introduction

A radio tx^r tx^t s or radiates a modd carrier voltage. This modd carrier is picked up by the antenna of the radio receiver. The signal so received is very weak. Hence generally this signal is first amplified in an RF amp stage of the radio receiver. Further since the signal is accompanied by lot of unwanted signals (noise) at adjacent freqs, it must be selected and the noise be rejected. finally the RF carrier must be demodd to get back the original modulating signal. Also ∵ the detected signal is usually weak, it has got to be amplified in one or more stages of audio amplifiers.

(*) A radio receiver is an electronic equipment which picks up the desired signal, rejects the unwanted signal, amplifies the desired signal, demodulates the carrier to get back the original modn freq signal.

* Radio Receivers are classified according to the type of traffic they are designed to handle.
Traffic ; AM, FM, TV, Comm :- for code Mgmt / Code telegraph
Telegraph
SW Telephone
Radar

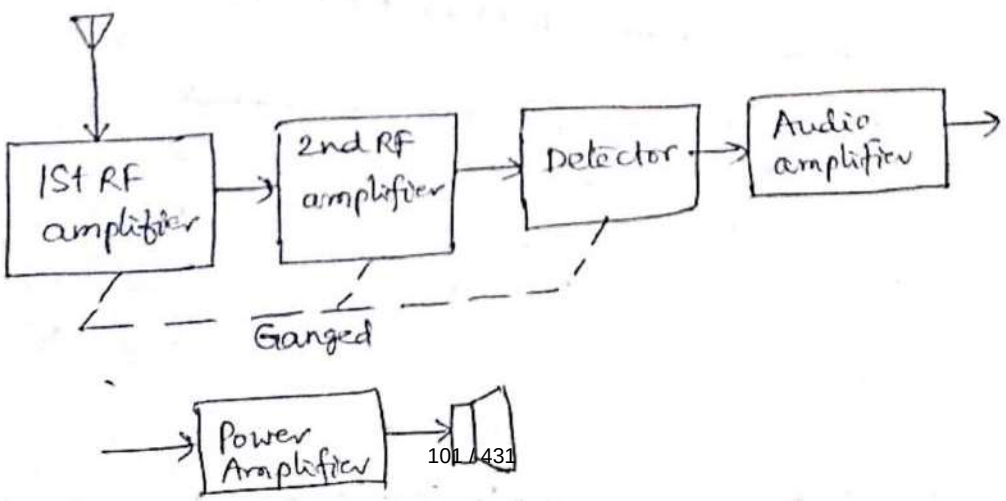
Types of Radio Receivers

① Tuned Radio frequency Receiver
② Superheterodyne Receiver
} have any real practical or commercial significance.

SHR is used to a large extent

TRF Receiver

Until shortly before World War II, most radio rx^r's were of TRF type.



TRF receiver is a simple logical receiver.

It is an improved version over the previously used – crystal, regenerative, Superregenerative receivers

Two or three RF amplifiers, all tuning together were employed to select and amplify the incoming frequency and simultaneously to reject all others.

After the signal was amplified to a suitable level, it was demoded and fed to the ~~the~~ HS, after being passed through the appropriate audio amplifying stages.

Advantages

Simple to design and align at broadcast freqs.

Disadvantages

Problems

Freq. instability (Due to High gain)

Variation in B.W. over the tuning range.

Insufficient adjacent freq. rejection ~~(at HF)~~

Insufficient selectivity at HF ; due to use of single tuned ckts

Consider a tuned ckt reqd to have a BW of 10KHz at fs = 535KHz.

$$Q = \frac{f}{\Delta f} = \frac{535\ KHz}{10KHz} = 53.5$$

At 1640KHz $\frac{1640}{10} = 164$.

Various losses dependent on freq will prevent so large an ↑ in Q.

and Q = 120 $\Delta f = \frac{f}{Q} = \frac{1640}{120} = 13.7\ KHz$, ensuring the r_x^r will pick up adjacent stns, inaddition to the stn to which it is tuned.

on SW Band $Q = \frac{36.5MHz}{10KHz} = 3.65 \times 10^3 = 3650$. Q = impossible.

Superheterodyne Receiver

heterodyne = mix

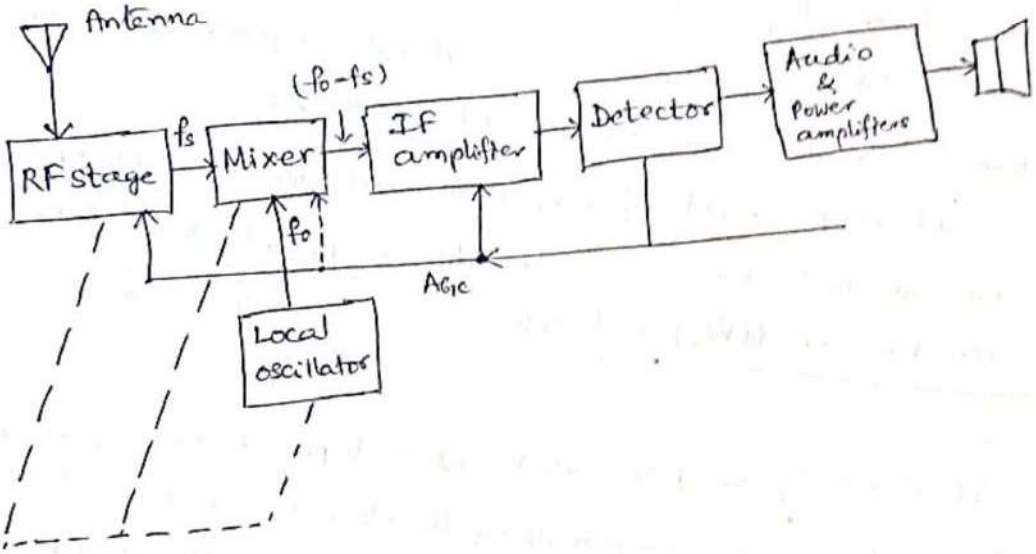
Superheterodyne → freq conversion takes place one or more times before detection.

In practice, the name Superheterodyne is applied to receivers in which only one freq conversion takes place before detection.

Double S.H.R :- [Triple detection Rx]

The Receiver in which freq conv. takes place twice before det. is called a double S.H.R. or Triple detection Rx.

Block diagram



Ganged tuning

Antenna :- It intercepts the EM waves. Voltages induced in the ant are fed to the i/p ckt of the Rx through a feeder wire.

RF amplifier :-

The RF stage is normally a wide band RF amp tunable from approx. 540 KHz - 1650 KHz (Standard commercial AM band). It is mechanically tied to the L.O. to ensure precise tuning chars.

RF ckts are used to select the wanted freq, to reject interference such as the image freq and (especially at HFs) to reduce the noise-fig. of Rx.

The desired freq signal is amplified by the RF stage.

~~Mixer~~ Local oscillator :- It is a variable oscillator capable of generating a signal from 0.995 MHz to 2.105 MHz

Mixer :- The o/p signal of RF stage is (f_s) combined with a predetermined L.O. signal in the mixer stage. (During this stage a class C non linear device processes the signals, producing the sum, difference, and original signals.)

Ganged tuning:-
A constant freq difference is maintained between the L.O. and the RF ckts, normally through capacitance tuning, in which all the capacitors are ganged together and operated in unison by one control knob.

IF amplifier
The signal from the mixer is fed to the IF (Intermediate freq) amplifier. It is a very-narrow bandwidth class A device capable of selecting a freq of 0.455 KHz ± 3KHz and rejecting all others. IF signal o/p is an amplified composite of the modd RF from the t_x^r in combination with RF from the L.O.

IF amp. consists of large no. of (6) double tuned ckts operating at const. specified chosen freq. and hence provides most of the gain (and ∴ sensitivity) and B.W. requirements of the R_x^r.

Detector
It eliminates one of the side bands still present and seperates the RF from the audio components of the other side band.
The RF is filtered to ground, and audio is supplied or fed to the audio stages for amplification.

Audio & power amplifiers
These will provide additional amplification

Loud speaker
Amplified audio o/p voltage of audio power amplifier is fed to loudspeaker; which reproduces the original programme.

L.O. is tuned to corresponding IF
640KHz + 455 KHz = 1095KHz.

f_s = 640 KHz
f_o = 1095KHz
$f_s + f_o$ = 1735 MHz, 455KHz

Radio receiver characteristics:-

Sensitivity :-

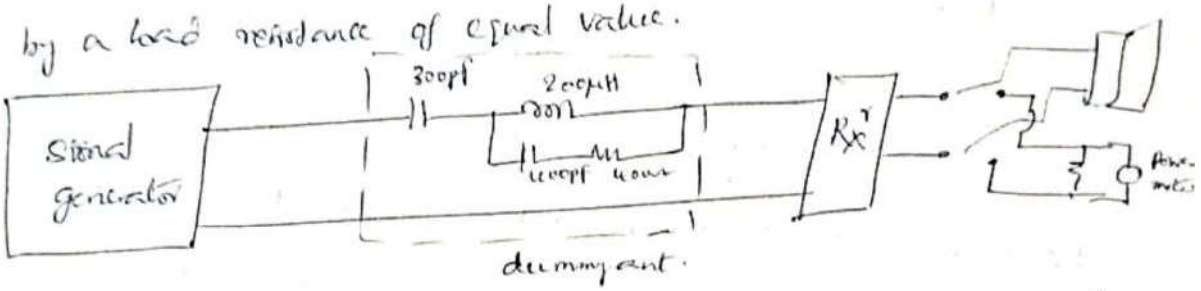
Def:- The sensitivity of a radio receiver is its ability to amplify weak signals

It is often defined in terms of the voltage that must be applied to the receiver input terminals to give a standard output power measured at the o/p terminals.

[For AM broadcast receivers, several of the relevant quantities have been standardized]

Thus 30% modulation by a 400 Hz sine wave is used, and the signal is applied to the receiver through a standard coupling n/w known as a dummy antenna.

The standard o/p is 50 mW. and for all types of Rx's the L.S. is replaced by a load resistance of equal value.

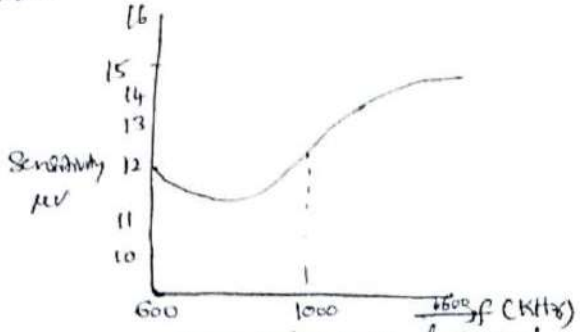


Sensitivity is often expressed in μV. or in dB below 1V and measured at 3 points along the tuning range.

The sensitivity varies over the tuning range

In terms of signal o/p power it is described in dB below 1mW or dBm.

The most important factor determining the sensitivity of a SHARP are gain of the IF amp. and that of RF amp.

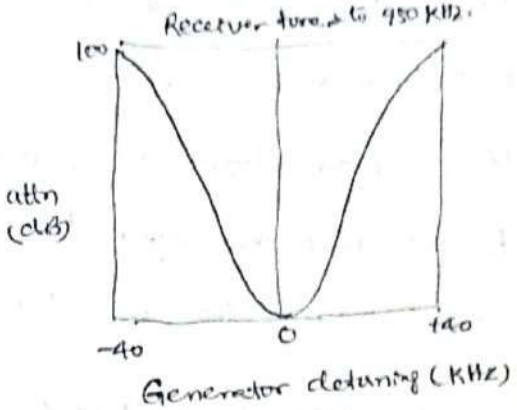


Sensitivity curve for good domestic receiver

Selectivity :-

It is the ability to reject unwanted signals.

Selectivity curve shows the attn that the rx offers to signals at freqs near to one to which it is tuned.



Selectivity is measured at the end of a sensitivity test with ~~the~~ conditions the same as for sensitivity, except that now the freq of the generator is varied to either side of the freq to which the rx is tuned.

The o/p of the rx naturally falls, ∴ the attn is now incorrect

The V/p must be increased until the o/p is the same as it was originally.

The ratio of the voltage required at resonance to voltage required when the generator is tuned to the rx's freq is calculated at a no. of points and then plotted in dB to give a curve,

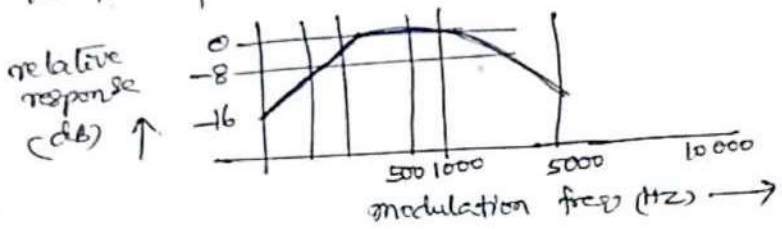
Selectivity is determined by the response of the IF section, with the mixer and RF amp i/p ckts playing a small but significant part.

Fidelity:- It is the ability of a radio receiver to reproduce correctly different modulating freq components in an i/p signal.

Fidelity of a receiver represents variation of the o/p with modulation freq, when the o/p load impedance is resistive.

A receiver must be able to reproduce the incoming signal without distortion; then such a receiver is said to have good fidelity.

fidelity at lower modl^n freqs is determined primarily by the LF. char. of the Af amp, at the higher modl^n freqs the fidelity is effected by the higher freq char of Af amplifier.



Signal to noise ratio (or) Noise figure :-

It is a measure of the extent to which the noise appearing in the receiver output in the absence of a signal is greater than the noise that would be present if the receiver is perfect receiver from the point of view of generating the minimum possible noise.

$$\text{Noise figure} = \frac{\text{Signal to noise power ratio of ideal receiver}}{\text{actual signal to noise power ratio of receiver o/p.}}$$

(Pa) When an SHR is tuned to 555 KHz its I.F. provides the mixer with an i/p at 1010KHz. What is the image freq? The antenna of this Rx is connected to the mixer via a tuned ckt whose loaded Q = 40. What will be the image freq rejection ratio for the calc image freq!

$f_s = 555$ KHz $f_i = f_o - f_s = 1010 - 555 = 455$ KHz

$f_o = 1010$ KHz

$f_{si} = f_s + 2f_i$

$= 555 + (2 \times 455) = 1465$ KHz.

$Q = 40$.

$\alpha = \sqrt{1+Q^2\rho^2} = \sqrt{1+40^2 \times (2.2608)^2} = 90.43$ Any

$\rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} = \frac{1465}{555} - \frac{555}{1465} = 2.2608.$

(Pb) Cal. the image rejection of a receiver having an RF amp and IF of 450 KHz. If the Q's of relevant coils are 65 at an incoming freq of (a) 1200 KHz (b) 20MHz

$f_i = 450$ KHz

$Q = 65$

(a) $f_s = 1200$ KHz

$f_{si} = f_s + 2f_i = 1200 + 900 = 2100$ KHz

$\rho = \frac{2100}{1200} - \frac{1200}{21} = 1.1786$

$\alpha = \sqrt{1+65^2 \times 1.1786^2} = 76.616$

(b) $f_s = 20$MHz

$f_{si} = 20M + (2 \times 450K) = 20.9$ MHz

$\rho = \frac{20.9}{20} - \frac{20}{20.9} = 0.0881$

$\alpha = \sqrt{1+65^2 \times 0.0881^2} = 5.813$

Adjacent channel selectivity (Double spotting)

Same SW stn is picked up at two nearby points on the receiver dial.

It is caused by poor front end selectivity. i.e. inadequate image frequency rejection.

Lack of selectivity is harmful because a weak stn may be masked by the reception of a nearby strong station at the spurious point on the dial.

[Double spotting may be used to calculate the IF of an unknown receiver, ∵ the spurious point on the dial is precisely 2fi below the correct freq.]

An improvement in image freq rejection will produce a corresponding reduction in double spotting.

A TRF Rx^r is tuned to a 1000KHz AM radio broadcast signal by a variable tuned ckt with 1KHz BW. Find the BW when the receiver is returned to 1550KHz and 550 KHz. Det the recovered Baseband.

$$Q = \frac{fc}{BW} = \frac{1000K}{1K} = 1000.$$

BW. of Rx^r. (i) $BW = \frac{fc}{Q} = \frac{1550K}{1000} = 1550$

(ii) $BW = \frac{550K}{1000} = 550$

recovered baseband range is from 550 Hz - 1550 Hz.

Find the tuning range necessary for the oscillator capacitor in a MW SHR which tunes over the range of signals from 530 KHz to 1650 KHz. and uses an IF of 455 KHz if the oscl frey is (i) higher than signal frey (ii) lower than signal frey.

IF = 455 KHz

(i) $f_{LO} > f_s$

$f_{LO} - f_s = 455\,KHz$

$f_{LO} = f_s + 455\,KHz = f_s + f_{IF}$

tuning range = (530+455) KHz to (1650+455) KHz

= 985 KHz to 2105 KHz.

(ii) $f_{LO} < f_s$

$f_s - f_{LO} = 455\,KHz$

$f_{LO} = f_s - 455\,KHz = f_s - f_{IF}$

tuning range = (530 − 455) KHz to (1650 − 455) KHz

= 75 KHz to 1195 KHz.

A superheterodyne receiver having an RF amplifier and an IF of 450 KHz is tuned to 15 MHz. cal. the Q's of the RF and mixer input tuned ckts, both being the same, if the receiver's image rejection is to be 120.

IF = 450 KHz $f_s = 15\,MHz$. $\alpha = 120$.

$f_{si} = f_s + 2f_i = 15M + 2 \times 450K = 15.9\,MHz$.

$\rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} = \frac{15.9}{15} - \frac{15}{15.9} = 0.117$

$\alpha = \sqrt{1+Q^2\rho^2} \Rightarrow Q = \sqrt{\frac{\alpha^2-1}{\rho^2}} =$ 1034

Calculate the image rejection of a receiver having an RF amplifier and an IF of 450 KHz, IF Q of the coils is 65, at an incoming freq of (i) 1200 KHz, (ii) 20 MHz.

$IF = 450\,KHz$, $Q = 65$

(i) $f_s = 1200\,KHz$

$f_{si} = f_s + 2 \cdot IF = 1200 + 2(450) = 2100\,KHz$

$\rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}} = \frac{2100}{1200} - \frac{1200}{2100} = 1.17$

$IR = \sqrt{1 + Q^2\rho^2} = \sqrt{1 + (65^2 \times 1.17^2)} = 76.57$.

(ii) $f_{si} = 20M + 900K = 20.9\,MHz$

$\rho = \frac{20.9M}{20M} - \frac{20M}{20.9M} = 0.088$

$IRR = \sqrt{1 + \rho^2 Q^2} = \sqrt{1 + [0.088^2 \times (65^2)]} = 5.806$

Unit V : PULSE MODULATION TECHNIQUES

Sampling

Sampling of signals is the fundamental operation in signal processing.
A continuous time signal is first converted to discrete time signal by sampling process.
The sufficient no. of samples of the signal must be taken so that the original signal is represented in its samples completely.
Also it should be possible to recover or reconstruct the original signal completely from its samples.
The no. of samples to be taken depends on maximum signal freq. present in the signal.

Sampling theorem :-

A continuous time signal may be completely represented in its samples and recovered back if the sampling freq $f_s \geq 2f_m$.
where f_m = max. freq. present in the signal.

Nyquist rate :-

When the sampling rate becomes equal to $2f_m$ samples per second, it is called Nyquist rate
It is also called minimum sampling rate.
It is given by $f_s = 2f_m$.

Nyquist interval

Maximum sampling interval is called Nyquist interval.
It is given by $T_s = \frac{1}{2f_m}$ seconds.

Sampling Techniques.

(i) Instantaneous sampling (or) ideal sampling
(ii) Natural sampling } practical sampling methods
(iii) Flat top sampling

Instantaneous sampling :-
(Ideal sampling or impulse sampling)

- The sampling function is a train of impulses.
- $x(t)$ is the input signal to be sampled.
- The ckt shown to produce instantaneous sampling is known as switching sampler.
- The ckt simply consists of a switch.
- If we assume that closing time 't' of switch approaches zero, then the o/p $g(t)$ of this circuit will contain only instantaneous value of the input signal $x(t)$
- ∴ the width of the pulse approaches zero, the instantaneous sampling gives a train of impulses of height equal to the instantaneous value of $x(t)$ at the sampling instant.

train of impulses may be represented as

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

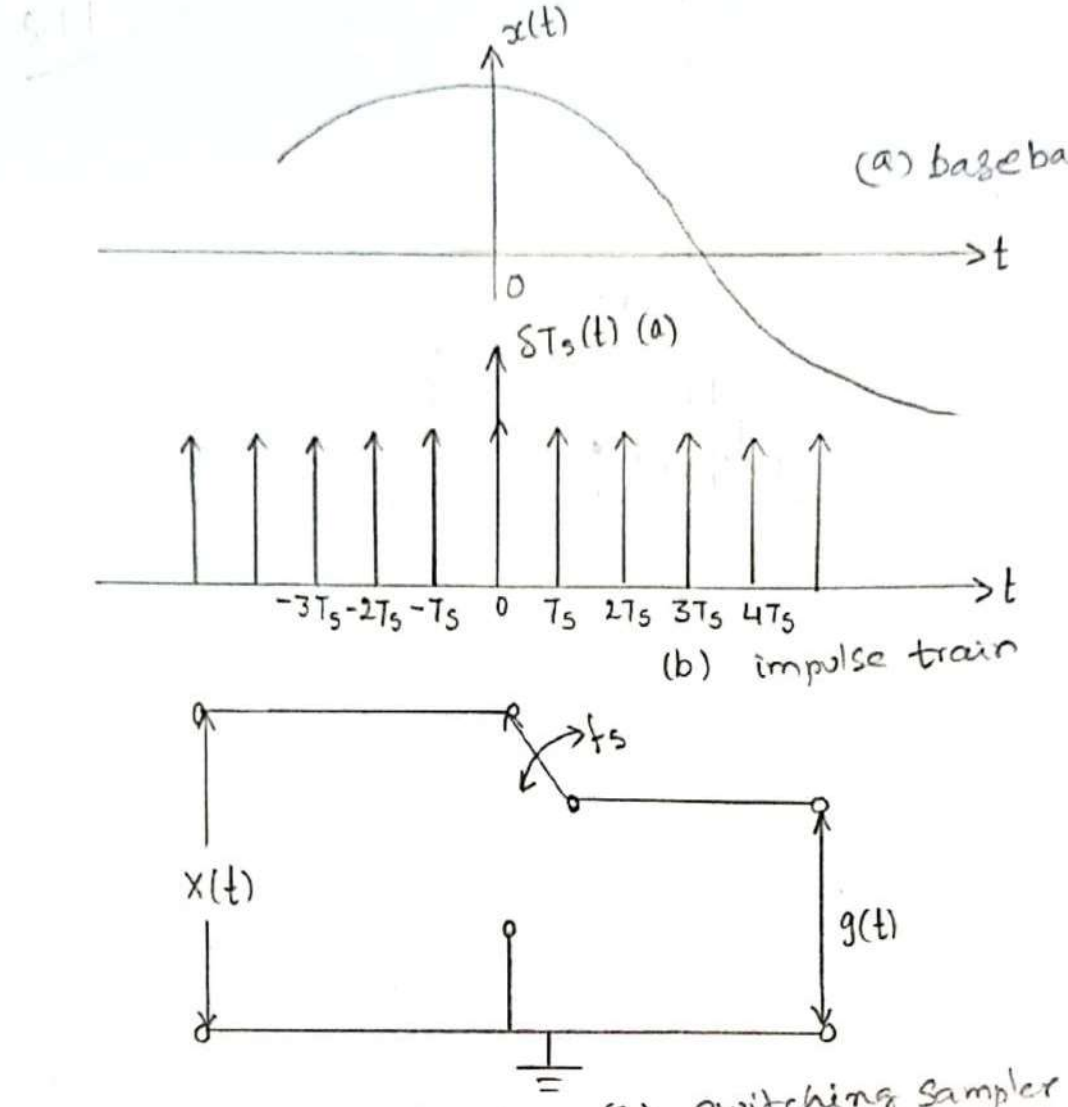
This is known as sampling function.

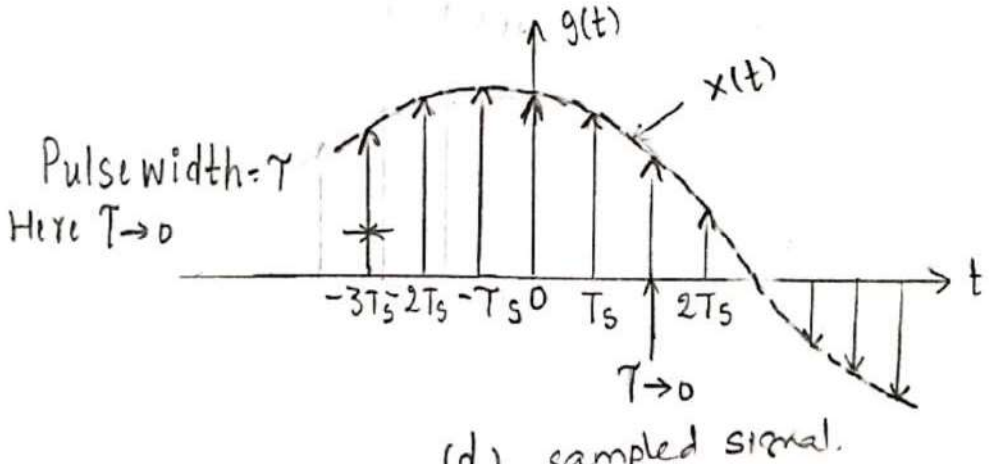
$$\text{Sampled signal } g(t) = x(t) \cdot \delta_{T_s}(t)$$

$$= x(t) \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t - nT_s)$$

Instantaneous sampling





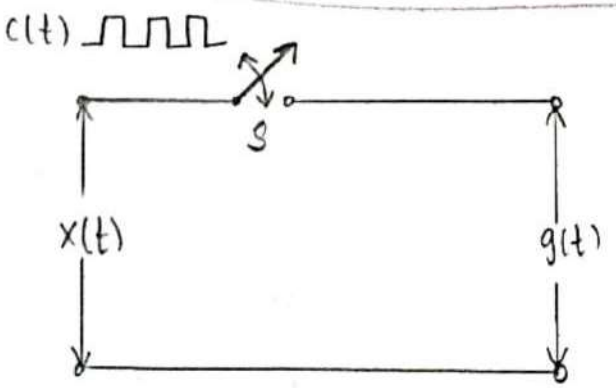
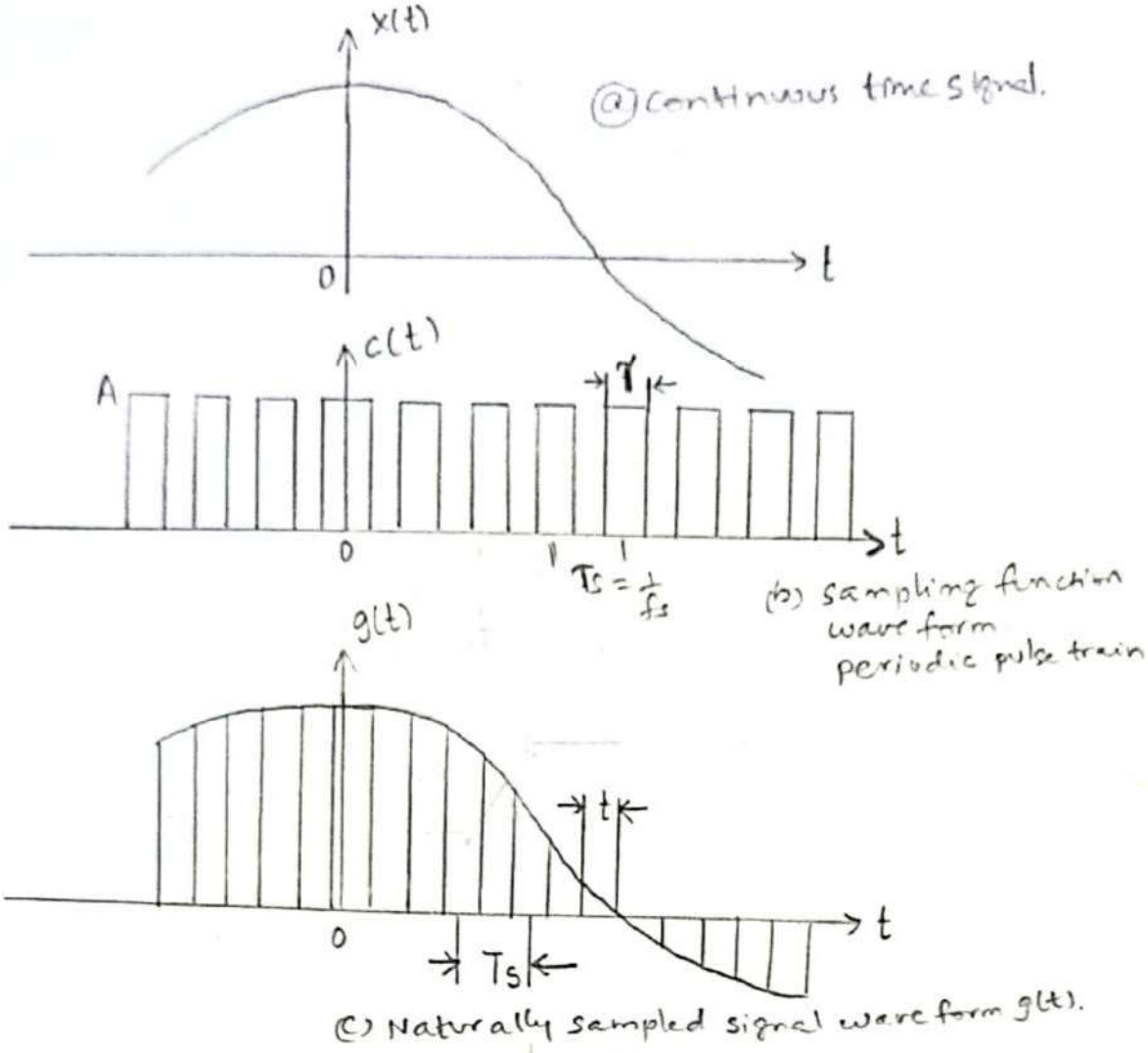


Fig: A functional diagram of Natural Sampler



(a) Continuous time signal.

(b) sampling function waveform periodic pulse train

(c) Naturally sampled signal waveform g(t).

Natural sampling.

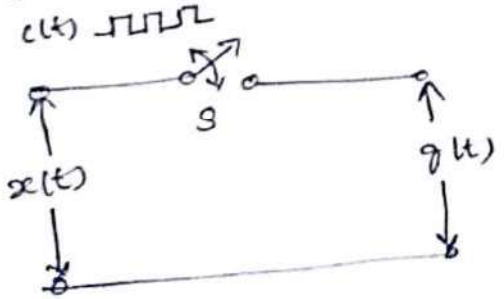
In natural sampling, the pulse has a finite width equal to τ.

Consider a continuous time signal x(t) to be sampled at the rate of f_s Hz.

f_s is higher than Nyquist rate.

Sampling function c(t) is a train of periodic pulses of width τ and frequency equal to f_s Hz

Fig. shows a natural sampler



$g(t) = c(t) \cdot x(t)$.

c(t) is high (A), s is closed $g(t) = x(t)$

low (0), s is open $g(t) = 0$

where A = amplitude of c(t)

Sampled signal $g(t) = c(t) \cdot x(t)$.

where c(t) = periodic train of pulse of width τ and freq f_s.

w.k.t Exponential fourier series for any periodic w.f is

$$c(t) = \sum_{n=-\infty}^{\infty} C_n e^{j2\pi n t / T_0}.$$

for the periodic pulse train $T_0 = T_s = \frac{1}{f_s}$ = period of $c(t)$.

$$f_0 = f_s = \frac{1}{T_0} = \frac{1}{T_s} = \text{freq. of } c(t)$$

$$\Rightarrow c(t) = \sum_{n=-\infty}^{\infty} C_n e^{j2\pi f_s n t} \quad \text{with } \frac{1}{T_0} = f_s.$$

$$C_n = \frac{A\tau}{T_0} \operatorname{sinc}(f_n \tau)$$

where $\tau = \tau' =$ pulsewidth

$f_n =$ harmonic freq.

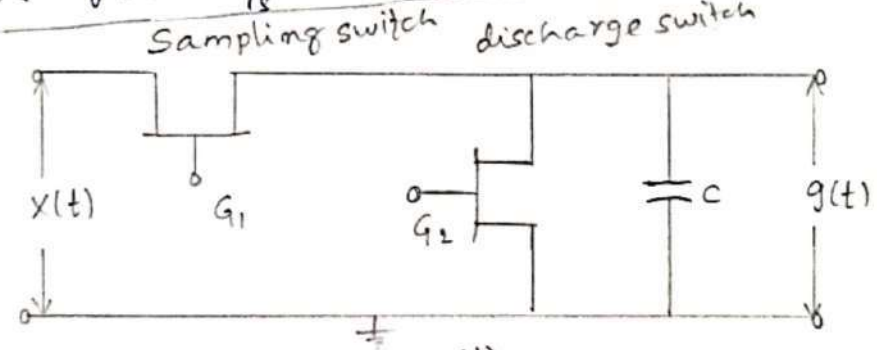
$f_n = n f_s$.

$f_s = \frac{f_n}{n}$

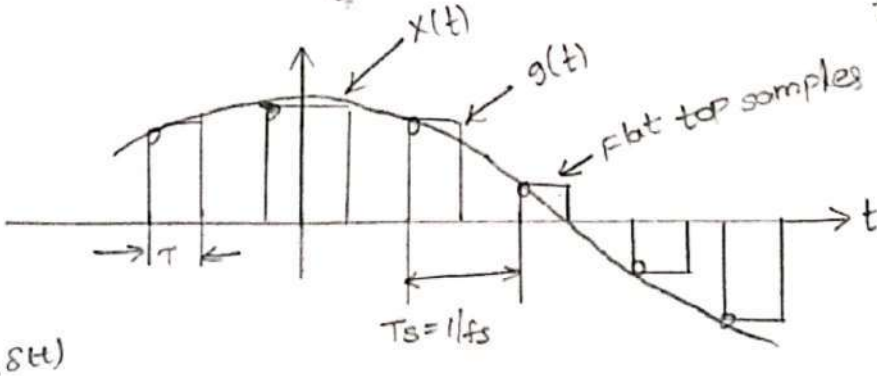
$$C_n = \frac{A\tau'}{T_s} \operatorname{sinc}(f_n \tau')$$

$$\therefore c(t) = \sum_{n=-\infty}^{\infty} \frac{A\tau'}{T_s} \operatorname{sinc}(f_n \tau') e^{j2\pi f_s n t}$$

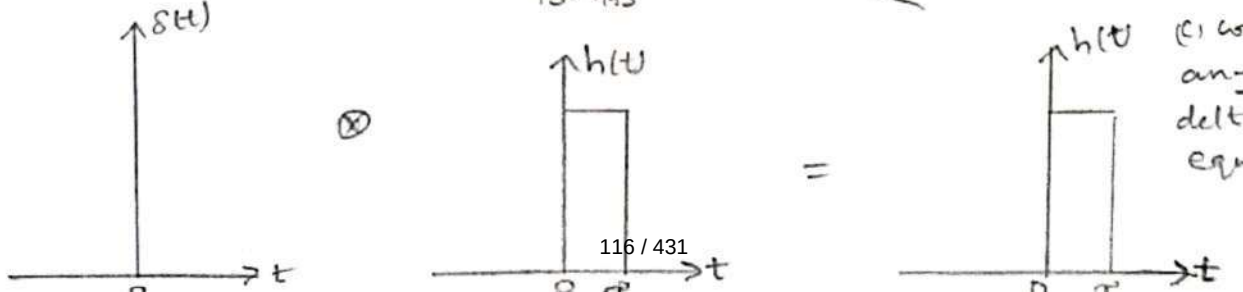
$$\therefore g(t) = \frac{A\tau'}{T_s} \sum_{n=-\infty}^{\infty} \operatorname{sinc}(f_n \tau') e^{j2\pi f_s n t} \cdot x(t).$$



(a) sample & hold ckt to generate flat top samples.



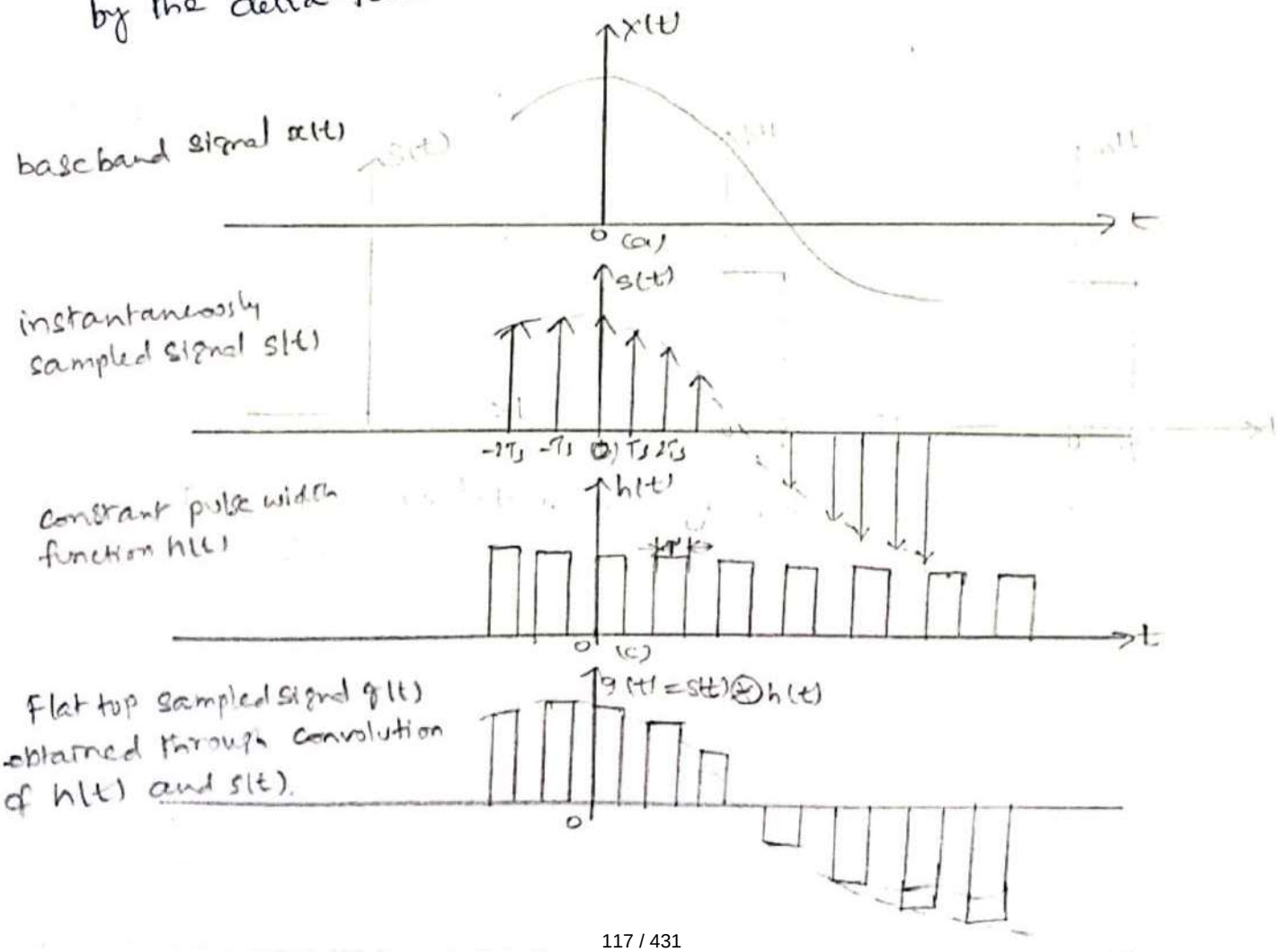
(b). w.f. of flat top sampling



(c) convol^n of any fn with delta fn is equal to that function.

Flat Top sampling or Rectangular sampling:

- The top of the samples remain constant and is equal to the instantaneous value of the baseband signal $x(t)$ at the start of sampling.
- width of each sample is τ and sampling rate is $f_s = \frac{1}{T_s}$.
- Fig@shows a sample and hold ckt used to generate flat top samples.
- fig (b) shows general w.f. of flat-top samples.
- It may be noted that only starting edge of the pulse represents instantaneous value of the baseband signal $x(t)$.
- The flat top pulse $g(t)$ is equivalent to convolution of instantaneous sample and a pulse $h(t)$.
- This means that width of the pulse in $g(t)$ is determined by width of $h(t)$ and the sampling instant is determined by the delta function.



- In fig (b) the starting edge of the pulse represents the point where baseband signal is sampled and width is determined by function $h(t)$.

$\therefore g(t) = s(t) \otimes h(t)$

from the property of delta function,

w.k.t. $f(t) \otimes \delta(t) = f(t)$.

This is used to obtain flat top samples.

- The above eq. is not directly applied but a modified form
- we are taking $s(t)$ in place of $\delta(t)$.
- $\delta(t)$ is a constant amplitude delta fn, where as $s(t)$ is a varying amplitude train of impulses.
- This means that we are taking $s(t)$ which is an instantaneously sampled signal and this is convolved with $h(t)$.
- on convolution of $s(t)$ and $h(t)$ we get a pulse whose duration is equal to $h(t)$, but amplitude is defined by $g(t)$.
- Train of impulses $\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$

$s(t) = x(t) \cdot \delta_{T_s}(t)$.

$s(t) =$

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t-nT_s)$$

$g(t) = s(t) \otimes h(t)$

$$= \int_{-\infty}^{\infty} s(\tau) h(t-\tau)\, d\tau$$

$$= \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t-nT_s)\, h(t-\tau)\, d\tau$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s) \int_{-\infty}^{\infty} \delta(t-nT_s)\, h(t-\tau)\, d\tau$$

w.k.t $\int_{-\infty}^{\infty} f(t)\, \delta(t-t_0) = f(t_0)$

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t-nT_s).$$

Pulse Modulation Techniques, types, definitions.

In Pulse modulation the carrier is a pulse train. Some characteristic of the pulse is varied according to the instantaneous value of the modulating signal.

Types of pulse modulation

(i) Pulse Amplitude Modulation (PAM)
(ii) Pulse Time Modulation (PTM)

PTM is divided in to two types.
(a) Pulse Width Modulation (PWM)
(b) Pulse Position Modulation (PPM)

Definitions:-

PAM :- Amplitude of the pulses of carrier pulse train is varied in accordance with the modulating signal.

PTM :- Timing of the pulses of the carrier pulse train is varied

PWM :- The width of the pulses of the carrier pulse train is varied in accordance with the modulating signal.

PPM :- Position of pulses of the carrier pulse train is varied

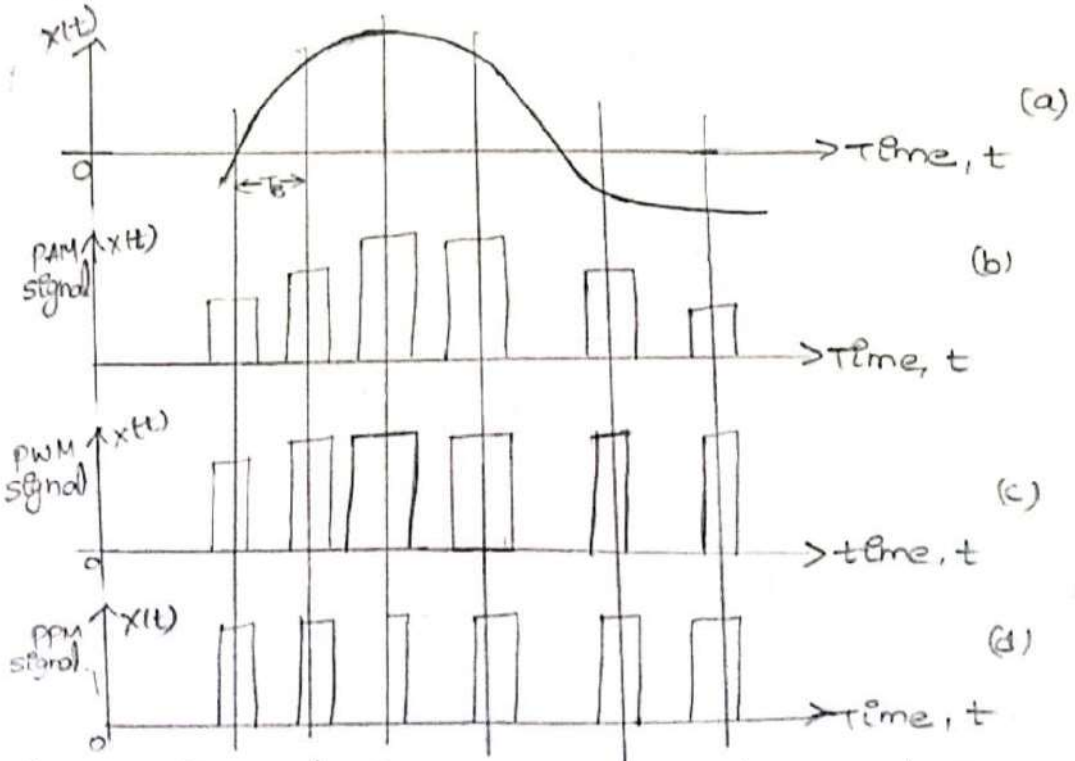


Fig: Different types of pulse analog modulation methods.

PULSE AMPLITUDE MODULATION (PAM)

- PAM is defined as that type of modulation in which the amplitudes of regularly spaced rectangular pulses vary according to the instantaneous value of the modulating signal.
- Pulses in PAM signal may be of flat top or natural type or ideal type.
- All the sampling methods discussed earlier are basically pulse amplitude modulation methods.

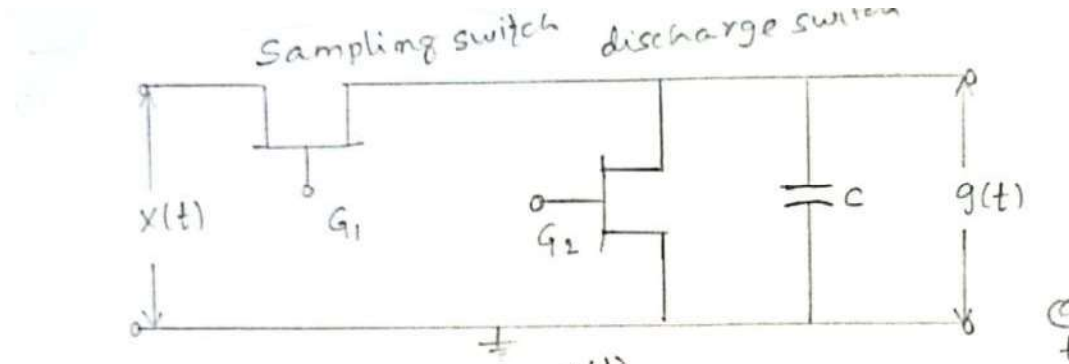
→ Flat top PAM is most popular and widely used. why?..

- During the transmission noise interfere with the top of transmitted pulses and this noise can be easily removed if the PAM pulse has flat top.
- In case of natural samples PAM signal, the pulse has varying top in accordance with the signal variation.
- When such a type of pulse is received at the receiver, it is always contaminated by noise.
- Then it becomes quite difficult to determine the shape of the top of the pulse and thus amplitude detection of the pulse is not exact. Due to this error are introduced in the received signal.

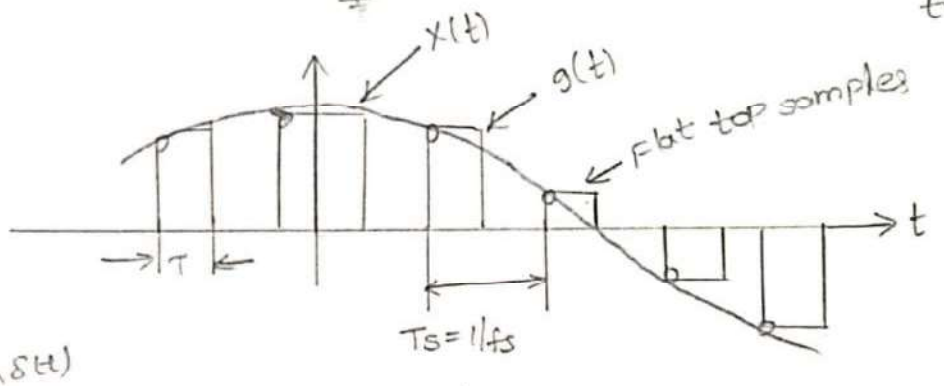
∴ Flat top sampled PAM is widely used.

Working Principle:

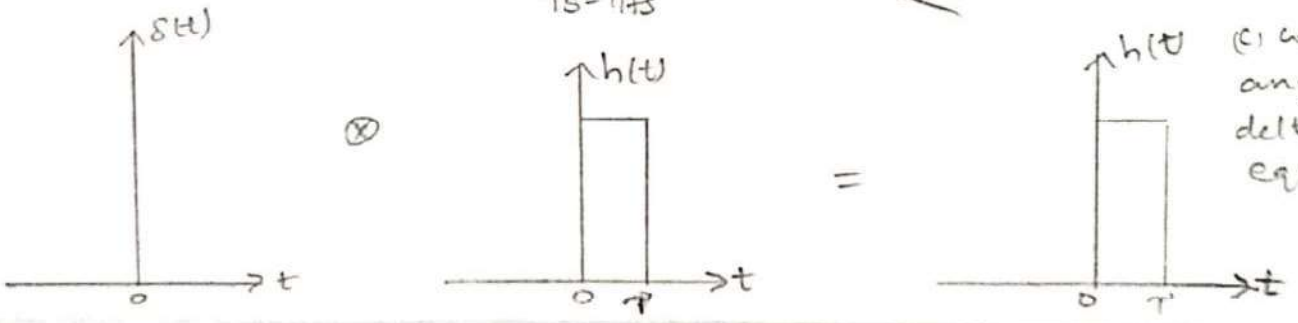
- A sample and Hold ckt shown in fig. is used to produce Flat top sampled PAM.
- The ckt consists of 2 FET switches and a capacitor.
- The sampling switch is closed for a short duration by a short pulse applied to the gate G_1 of the transistor.



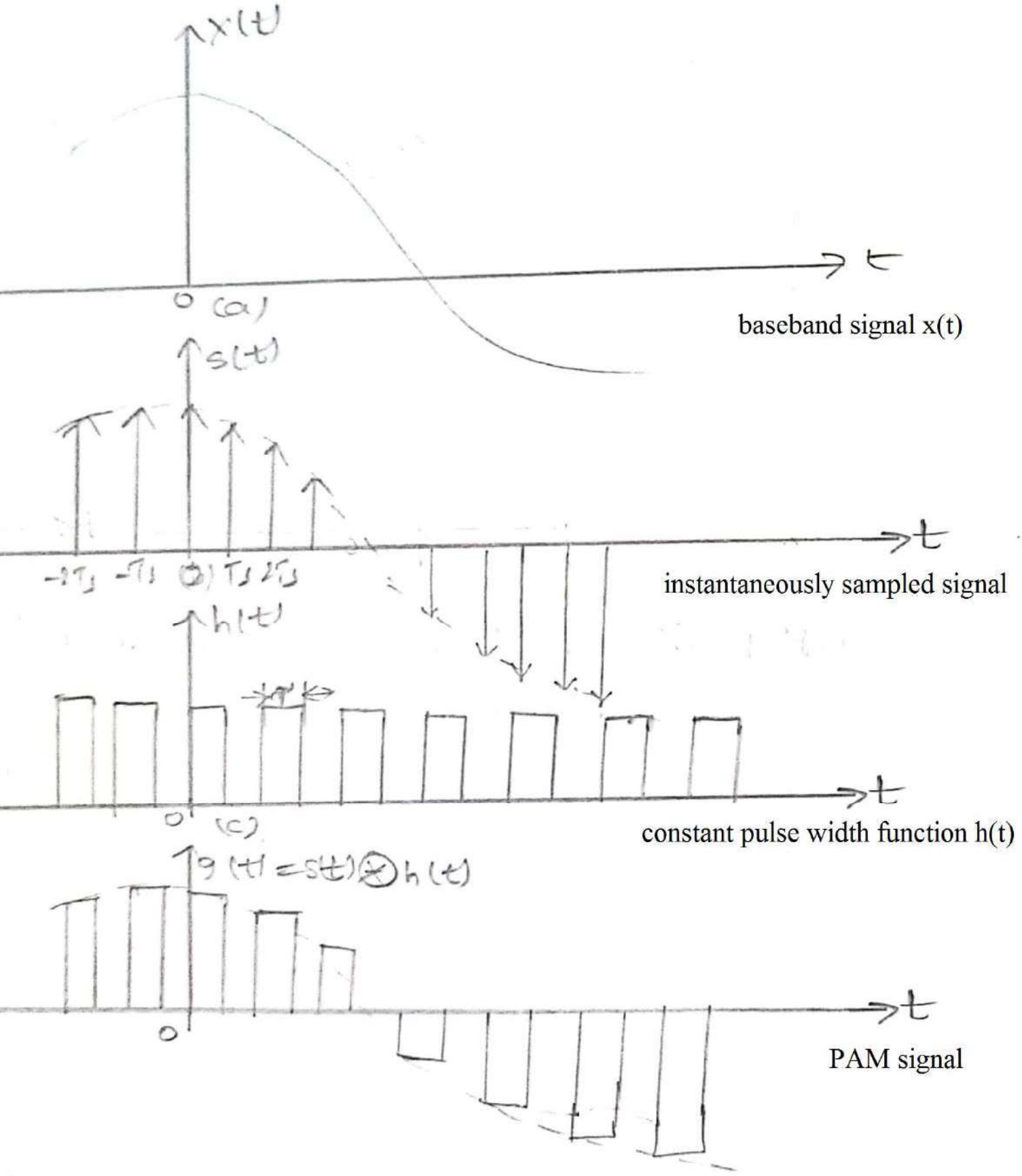
(a) sample & hold ckt to generate flat top samples.



(b). w.f. of flat top sampling



(c) convolⁿ of any fn with delta fn is equal to that function.


x(t)
t
0
(a)
baseband signal x(t)
s(t)
t
instantaneously sampled signal
h(t)
t
0
(c)
constant pulse width function h(t)
g(t) = s(t) ⊛ h(t)
t
0
PAM signal

- During this period the capacitor C is quickly charged upto a voltage equal to the instantaneous sample value of the incoming signal $x(t)$.
- Now the sampling switch is opened and the capacitor C holds the charge.
- The discharge switch is then closed by a pulse applied to gate G_2 of other transistor.
- Due to this the capacitor C is discharged to zero volts.
- The discharge switch is then opened and thus capacitor has no voltage.

Hence the o/p of S/H ckt consists of a sequence of flat top samples as shown in fig b

Mathematical analysis :-

In a flat top PAM, the top of the samples remains constant and is equal to the instantaneous value of the baseband signal $x(t)$ at the start of sampling.

The width of each sample is τ and sampling rate $f_s = \frac{1}{T_s}$

from fig b It may be noted that only starting edge of the pulse represents instantaneous value of baseband signal $x(t)$

Flat top pulse $g(t)$ is equivalent to convolution of instantaneous sample and a pulse $h(t)$.

$\Rightarrow$ Width of the pulse in $g(t)$ is determined by width of $h(t)$ and sampling instant is determined by delta function.

The starting edge of the pulse represents the point where base band signal is sampled and width is determined by $h(t)$.

$\therefore g(t) = s(t) \otimes h(t)$

from the property of delta function, w.k.t $f(t) \otimes \delta(t) = f(t)$

This property is used to obtain flat top samples.

To obtain F-T. sampling we are applying modified form of above eq.

we are taking s(t) in place of delta function δ(t).
δ(t) is a constant amplitude delta fn.
s(t) is a varying amplitude train of impulses
⇒ s(t) is an instantaneously sampled signal.
s(t) is convolved with h(t) to get a pulse whose duration is equal to h(t), amplitude is defined by s(t).

Train of impulses is represented by.

$$\delta_{T_s}(t) = \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$$

$$s(t) = x(t) \cdot \delta_{T_s}(t)$$

$$s(t) = x(t) \sum_{n=-\infty}^{\infty} \delta(t-nT_s)$$

$$= \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t-nT_s).$$

$$g(t) = s(t) \otimes h(t)$$

$$= \int_{-\infty}^{\infty} s(\tau)\, h(t-\tau)\, d\tau.$$

$$= \int_{-\infty}^{\infty} \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t-nT_s)\, h(t-\tau)\, d\tau$$

$$g(t) = \sum_{n=-\infty}^{\infty} x(nT_s) \int_{-\infty}^{\infty} \delta(t-nT_s)\, h(t-\tau)\, d\tau$$

acc. to the shifting property of delta fn.

$$\int_{-\infty}^{\infty} f(t)\, \delta(t-t_0) = f(t_0)$$

$$\therefore g(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, h(t-nT_s)$$

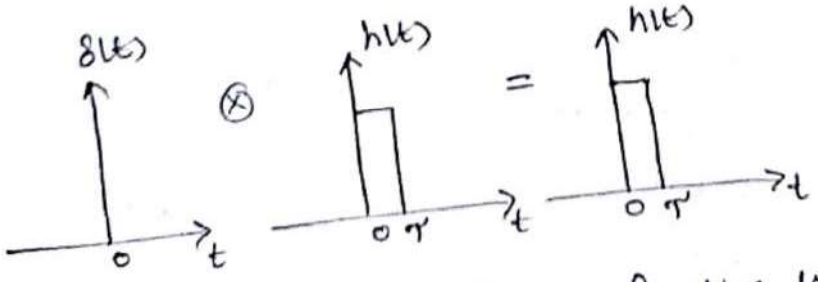


fig.. convolution of any function with delta function is equal to that function.

Demodulation of PAM signal.

For PAM signals demodulation is done using a holding circuit

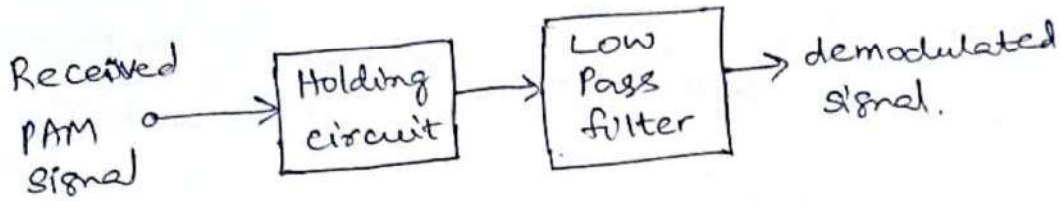
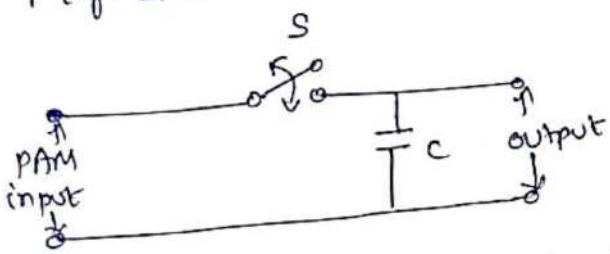
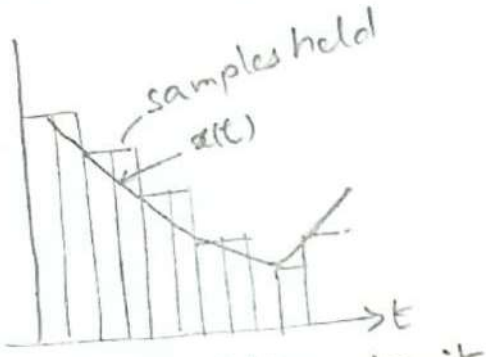


fig PAM demodulator.

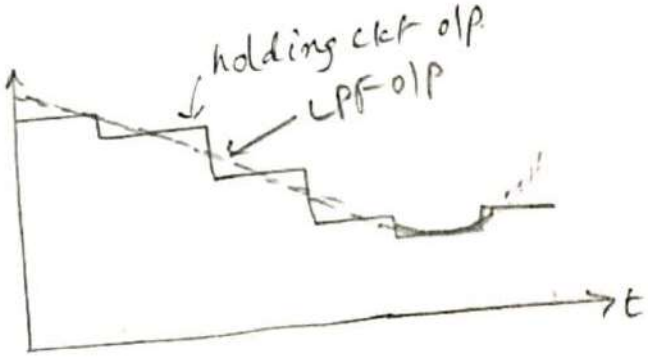
Fig. shows a zero order simple holding circuit.



The switch S is closed after the arrival of the pulse and it is opened at the end of the pulse.
This way the capacitor C is charged to pulse amplitude value and holds this value during the interval between the two pulses
Hence the sampled pulses are held as shown in fig (c).
The holding ckt o/p is smoothened in low pass filter.



o/p of holding circuit



output of LPF

Transmission Bandwidth in PAM.

- In PAM, pulse duration τ is considered very small in comparison to sampling period T_s

$$\tau << T_s.$$

- If the max. freq in the modg signal $x(t)$ is f_m, then sampling freq f_s must be greater than or equal to Nyquest rate. $\therefore f_s \geq 2f_m.$

$$\frac{1}{T_s} \geq 2f_m$$

$$T_s \leq \frac{1}{2f_m}$$

$$\therefore \tau << T_s \leq \frac{1}{2f_m}.$$

- If the ON and off time of PAM pulse is same. Then maximum freq of PAM is

$$f_{max} = \frac{1}{\tau + \tau} = \frac{1}{2\tau}.$$

$\therefore$ B.W. required for the tx^n of PAM signal = f_{max}.

Transmission Bandwidth

$$BW \geq f_{max},$$

$$BW \geq \frac{1}{2\tau}$$

$$\because \tau << \frac{1}{2f_m}$$

$$BW \geq \frac{1}{2\tau} >> f_m$$

$$\underline{BW >> f_m}$$

For a PAM Tx^n of voice signal having $f_m = 3KHz$ cal. Tx^n BW.

Given $f_s = 8KHz$ and $\tau = 0.1 T_s$.

$$T_s = \frac{1}{f_s} = \frac{1}{8K} = 125\mu s.$$

$$\tau = 0.1 T_s = 12.5\mu s.$$

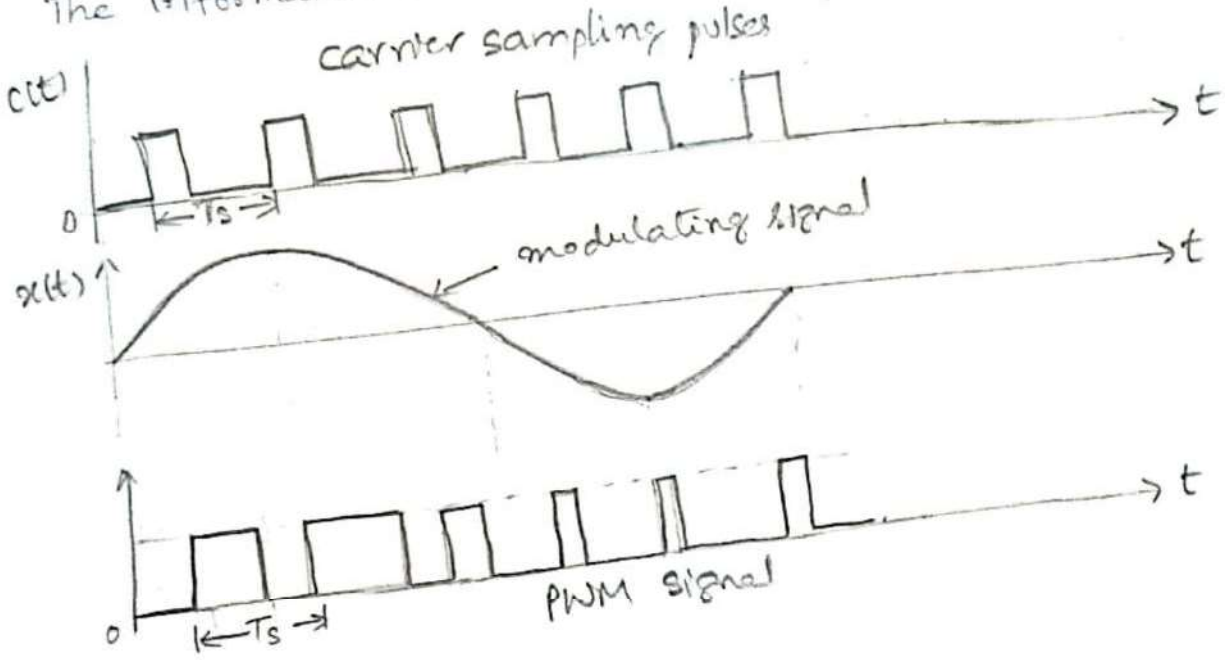
$$BW \geq \frac{1}{2\tau} = \frac{1}{25\mu s} = 40KHz.$$

Drawbacks of PAM :-

1. Tx^n BW >> fm.
2. Interference of noise is maximum.
3. Peak power required by the PAM varies with modg signal.

Pulse Width Modulation or Pulse Duration Modulation.

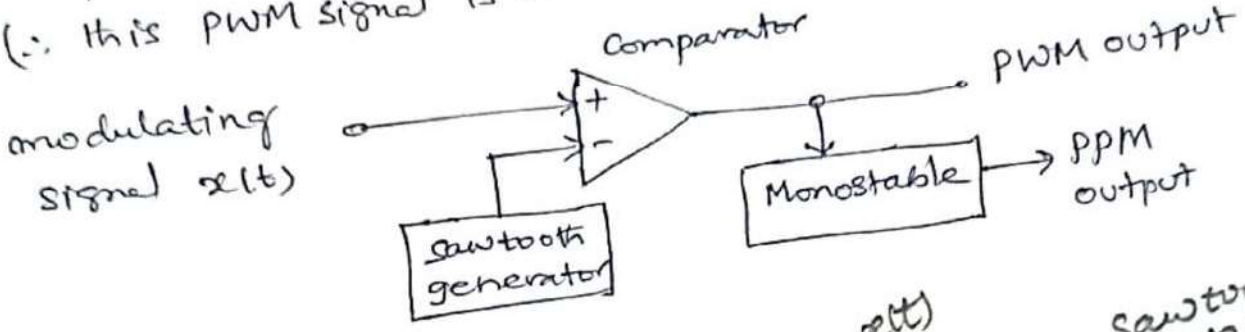
The information is contained in the width variation.

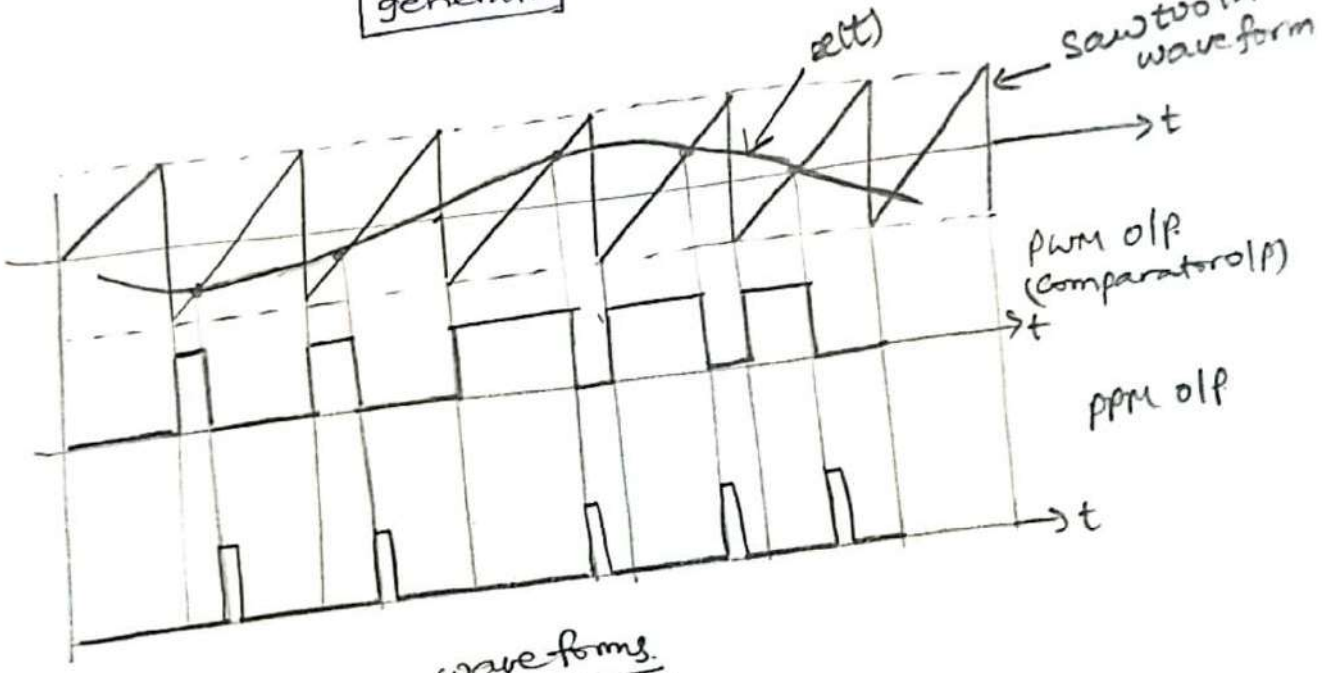


Generation of PWM signal:-

- A sawtooth signal of freq. fs is used as a sampling signal.
- It is applied to the inverting terminal of a comparator.
- The modulating signal x(t) is applied to the non-inverting terminal.
- The comparator o/p is high as long as the instantaneous amplitude of x(t) is higher than that of a ramp signal.

- This gives rise to a PWM signal at the comparator output.
- It may be noted that the leading edges of the PWM w.r.t. coincide with the falling edges of the ramp signal.
- ∴ the leading edges of PWM signal are always generated at fixed time instants.
- The occurence of trailing edges will depend on the instantaneous amplitude of $x(t)$.
(∴ this PWM signal is said to be trail edge modulated PWM)





waveforms.

Detection of PWM signal.

The PWM signal received at the i/p of the detector is contaminated with noise.
This signal is applied to pulse generator circuit which generates PWM signal.
Thus, some of the noise is removed and the pulses are squared up.

The regenerated pulses are applied to a reference pulse generator.
It produces a train of constant amplitude, constant width pulses. These pulses are synchronized to the leading edges of the regenerated PWM pulses but delayed by a fixed interval.
The regenerated pulse are also applied to a ramp generator. At the o/p of it, we get a constant slope ramp for the duration of the pulse.
The height of the ramp is thus proportional to the width of the PWM pulses.
At the end of the pulse, a sample and hold amplifier retains the final ramp voltage until it is reset at the end of the pulse.
The constant amplitude pulses at the o/p of reference pulse generator are then added to ramp signal.
The o/p of the adder is then clipped off at a threshold level to generate a PAM signal at the o/p of the clipper.
A Low pass filter is used to recover the original modulating signal back from the PAM signal.

Advantages of PWM:-

(i) Less effect of noise. i.e., very good noise immunity.
(ii) Synchronization b/n t_x^r & r_x^r is not essential.
(iii) It is possible to seperate signal from noise.

Disadvantages of PWM:-

(i) Due to the variable pulse width, the pulses have variable power contents. ∴ The t_x^n must be powerful enough to handle max. width pulse, though the average power t_x^d can be as low as 50% of max. power.
(ii) To avoid w.f. distortion, BW required for PWM communication is large compared to PAM.

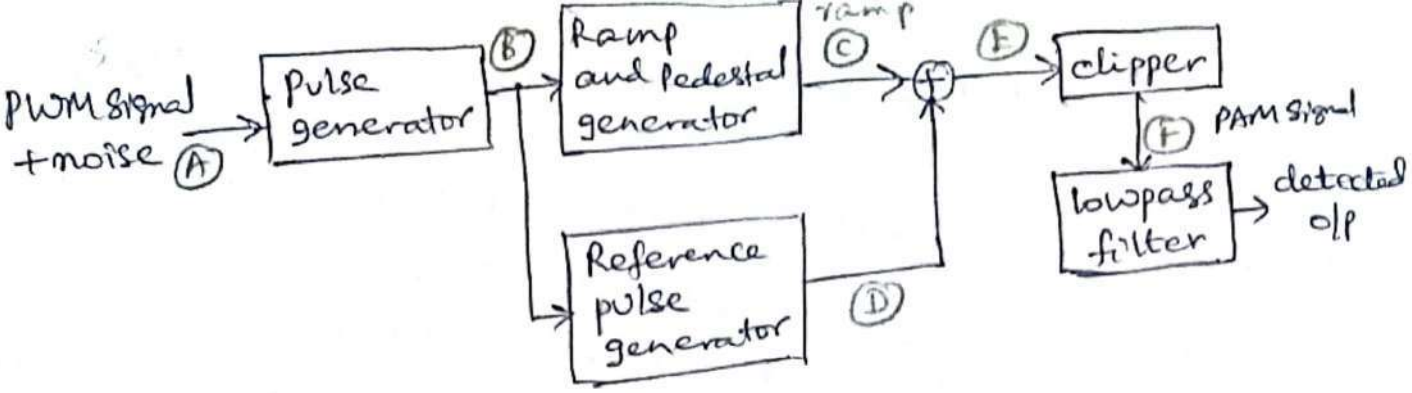
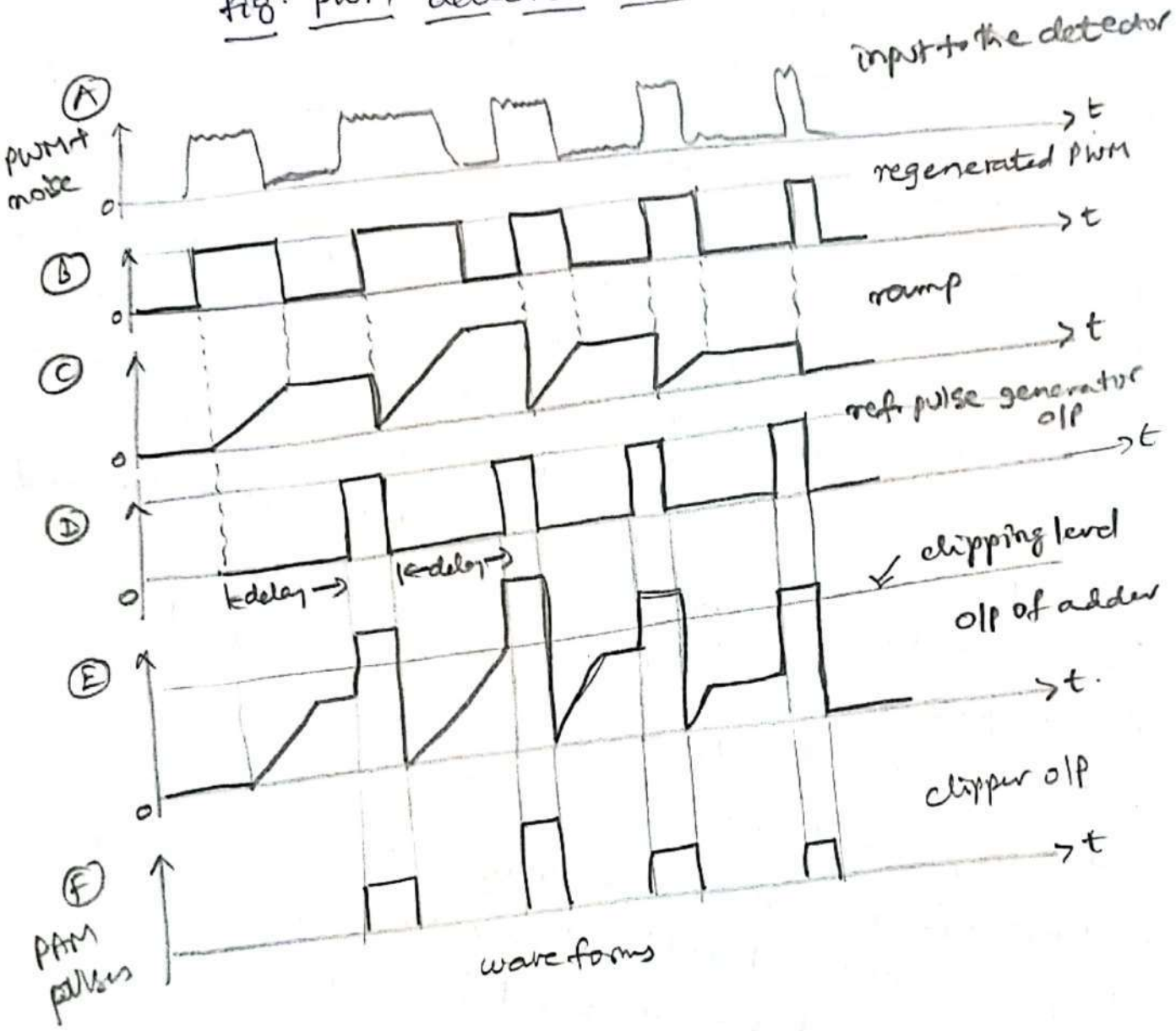


fig. PWM detection circuit

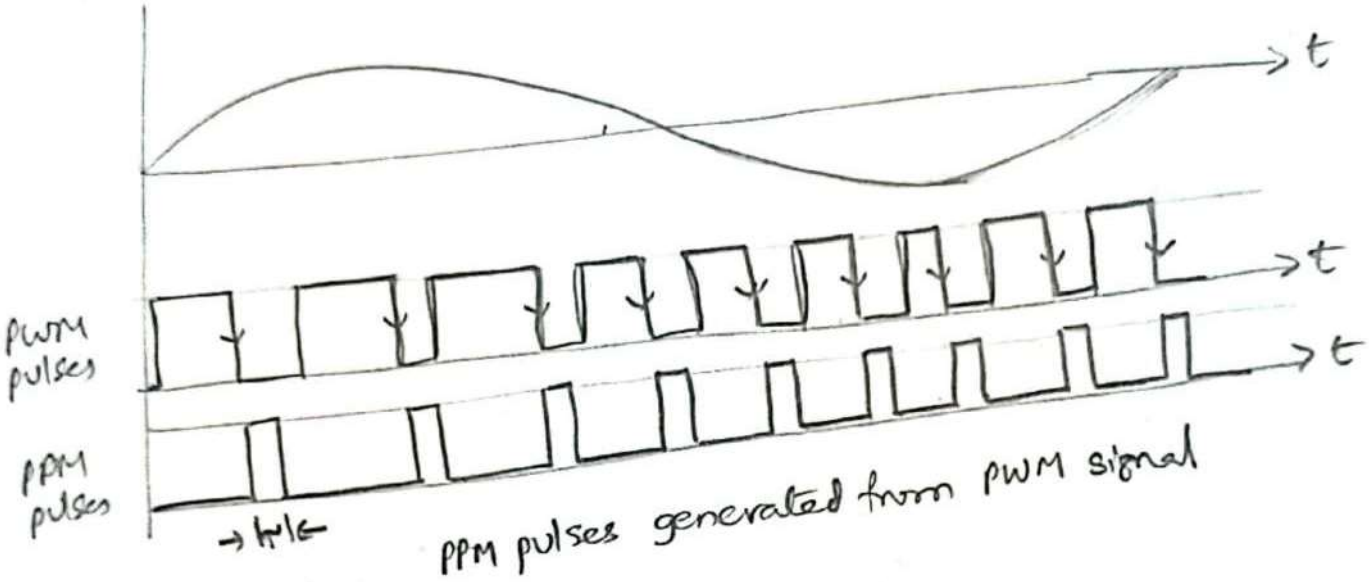


Pulse Position Modulation (PPM)

The position of the pulses is changed w.r.t the position of reference pulses.

PPM pulses can be derived from PWM pulses.

It may be noted that with increase in the modulating voltage the PPM pulses shift further w.r.t. reference.



The vertical dotted lines drawn in fig are treated as reference lines to measure the shift in position of PPM pulses.

PPM pulses marked 1, 2 and 3 go away from their respective reference lines.

This is corresponding to increase in the modulating signal amplitude.

As the modulating voltage decreases, PPM pulses 4, 5, 6, 7. come progressively closer to their respective ref. lines

Generation of PPM signal.

PPM signal can be generated from PWM signal as shown in fig.

PWM pulses obtained at the comparator o/p are applied to a monostable multivibrator.

The MMV is -ve edge triggered.

∴ corresponding to each trailing edge of PWM signal the monostable o/p goes high.

It remains high for a fixed time decided by its own RC components.

Thus as the trailing edges of the PWM signal keep shifting in proportion with the modulating signal x(t), the PPM pulses also keep shifting.

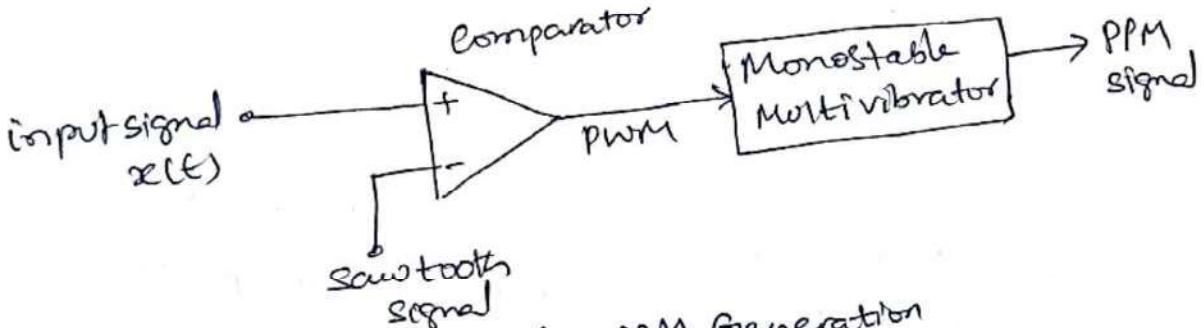


fig PPM Generation

Demodulation of PPM :-

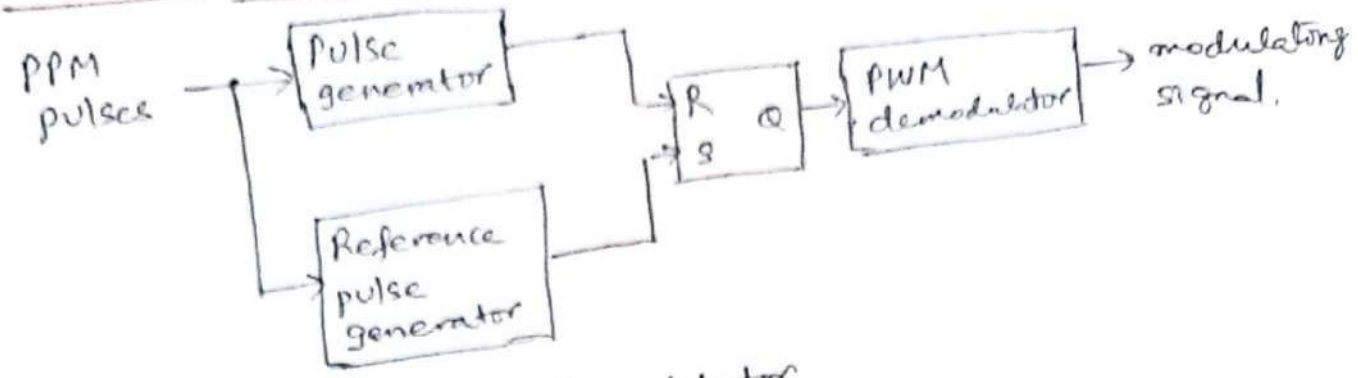


fig. PPM demodulator

The noise corrupted PPM wf received at the input of demodulator is applied to a pulse generator. The pulse generator develops a pulsed w.f. at its o/p of fixed duration and applies these pulses to reset pin of an SR flipflop.

A fixed period ref. pulse is generated from the incoming PPM wf.
This ref pulse is applied to set i/p of SR FF.
Due to the set and reset signals applied to the FF, we get a PWM signal.
This PWM signal is demodulated using PWM demodulator to get original message signal.

Advantages of PPM

(i) Due to constant amplitude of PPM pulses, the information is not contained in the amplitude.
∴ Noise added to PPM signal doesnot distort the information.
It has good noise immunity.

(ii) It is possible to reconstruct PPM signal from the noisy PPM signal.

(iii) As the amplitude of the pulses is constant, the transmitted power always remains constant.

Disadvantages of PPM:-

(i) As the position of PPM pulses is varied w.r.t. a ref. pulse, a transmitter has to send synchronizing pulses to operate the timing ckts in the receiver.

(ii) Large B.W is required to ensure tx^n of undistorted pulses

An analog signal is expressed by $x(t) = 3\cos 50\pi t + 10\sin 300\pi t - \cos 100\pi t$. Cal. the Nyquist rate for this signal.

$x(t) = 3\cos 50\pi t + 10\sin 300\pi t - \cos 100\pi t$

$x(t) = 3\cos \omega_1 t + 10\sin \omega_2 t - \cos \omega_3 t$.

$\omega_1 = 50\pi$

$f_1 = \frac{50\pi}{2\pi} = 25\text{Hz}$ $\quad f_2 = \frac{300\pi}{2\pi} = 150\text{Hz}$ $\quad f_3 = \frac{100\pi}{2\pi} = 50\text{Hz}$

max. freq present in $x(t) = 150\text{Hz}$

Nyquist rate $f_s = 2f_m = 2 \times 150 = 300\text{Hz}$.

Find the Nyquist rate and the Nyquist interval for the signal

$x(t) = \frac{1}{2\pi}\cos(4000\pi t)\cos(1000\pi t)$

$x(t) = \frac{1}{2\pi} \cdot \frac{1}{2}[\cos(4000\pi t + 1000\pi t) + \cos(4000\pi t - 1000\pi t)]$

$= \frac{1}{4\pi}[\cos 5000\pi t + \cos 3000\pi t]$

$\omega_1 = 5000\pi$

$f_1 = \frac{5000\pi}{2\pi} = 2500\text{Hz}$ $\quad f_2 = \frac{3000\pi}{2\pi} = 1500\text{Hz}$

$f_m = 2500\text{Hz}$

Nyquist rate $= 2 \times 2500 = 5\text{kHz}$

Nyquist interval $= \frac{1}{2f_m} = \frac{1}{5K} = 0.2\text{ ms}$.

Determine the Nyquist rate for a continuous time signal.

$x(t) = 6\cos 50\pi t + 20\sin 300\pi t - 10\cos 100\pi t$.

Nyquist rate $= \frac{300\pi}{\pi} = 300\text{Hz}$.

UNIT VI NOISE

Noise in Analog communication system:-

Noise in Amplitude Modulated systems:-

- In comm. sys message signal travels from the t_x^r to the r_x^r via a channel (radio, or line). This is shown in fig.

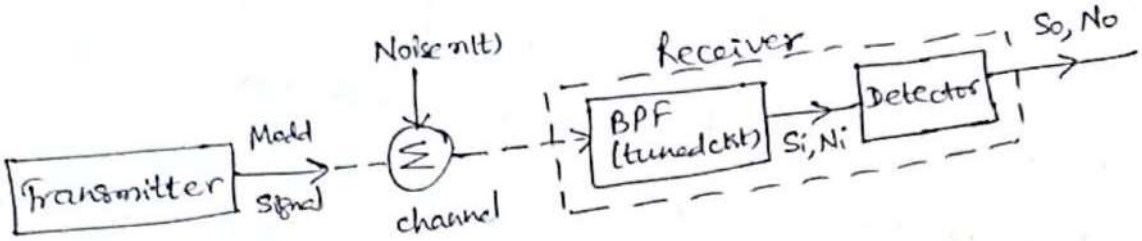


fig. channel noise in comm. sys.

- The channel introduces additive noise in the message.
- Hence the message reaching the receiver becomes corrupted.
- As the receiver detects both noise and message signals, it reproduces a noisy message at the output.
- The noise characteristics of a modulation system is evaluated by a parameter known as figure of merit denoted by γ.
- It is defined as the ratio of o/p SNR to i/p SNR of a receiver.

$$\gamma = \frac{So/No}{Si/Ni} \quad \text{---(1)}$$

where N_i = i/p noise power in a freq range equal to the message BW $2f_m$ including -ve freqs.

- A modulation system with higher γ has a better noise performance and adverse effect of noise is less.

Calculation of figure of merit for various Communication systems:-

- The noise performance of various comm. sys can be compared by evaluating their figure of merit γ.

For convinience, the following assumptions are made;-

(i) Channel noise is additive:- It is assumed that interference effect of noise is obtained by simple addition of signal f(t) and noise n(t)

(ii) Channel noise is white and Gaussian:- channel noise n(t) is white in nature. i.e; it is uniformly distributed over the entire band of freqs under consideration.

∴ The power density spectrum of the noise is uniform over the freq band under consideration.

- Hence the total noise power N is obtained simply by multiplying the noise power density spectrum $\frac{\eta}{2}$ with the bandwidth, i.e.,

$$N = (\eta/2) \times \text{Bandwidth} \quad \text{—(2)}$$

The noise amplitude has a Gaussian distribution.

(iii) Band pass Noise at the input of detector:-

- The first stage of every receiver is a tuned ckt that acts as a bandpass filter.
- This filter allows only the narrowband signal centered about carrier freq $\pm fc$ and rejects the other freqs.
- ∴ The noise signal lying outside this band is also rejected, and the bandwidth of the noise at the demodulator input is same as that of the modulated signal.
- The white noise at the input of the detector has a constant power density spectrum over the pass band as shown in fig.

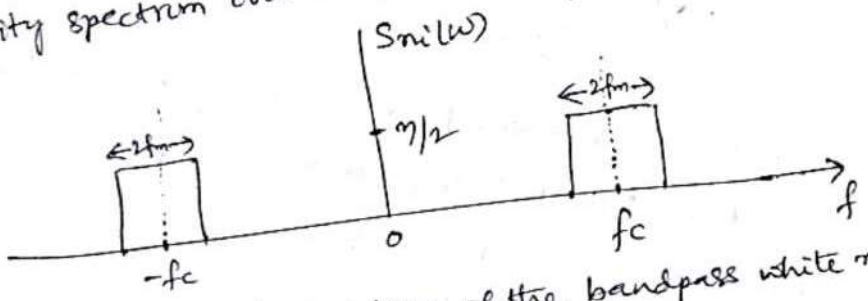


fig. Power density spectrum of the bandpass white noise.

The power density is given by $S_{ni} = \eta/2$ —(3)

(iv) The input noise power Ni:

for evaluating the figure of merit, the input noise power Ni is determined for baseband, i.e., 2fm Hz (including -ve freqs)

from eq(2) $$N_i = (\eta/2)(2f_m) = \eta f_m \quad \text{—(4)}$$

Eq(4) will be taken as reference input noise power for evaluating γ of various AM and FM receivers.

However Ni in eq(4) is the true input noise power only for the case of SSB systems.

Noise Calculation for Various AM Systems:-

- The performance of various AM systems can be studied by evaluating their individual figure of merit.

→ **Noise in DSB system:-**

1. DSB-SC :- The DSBSC system uses synchronous detection at the receiver.

- A schematic diagram of the tx^r and rx^r is shown in fig.

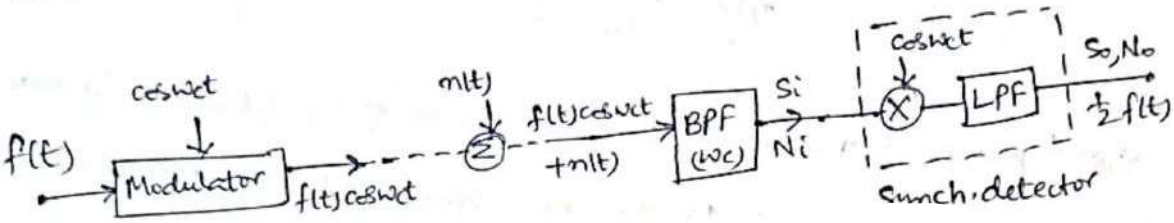


fig. A DSB-SC system.

Noise power:- The i/p noise power is given by eq(4)

- Now we will evaluate the noise power No at the o/p of synch. detector.
- The noise signal at the i/p of this detector is a bandpass noise given by. $n(t) = n_c(t)\cos\omega_c t - n_s(t)\sin\omega_c t$.
- This signal is multiplied by $\cos\omega_c t$ in the synch. detector.

The multiplied signal n_d is given by

$$n_d(t) = n(t)\cos\omega_c t$$
$$= [n_c(t)\cos\omega_c t - n_s(t)\sin\omega_c t]\cos\omega_c t.$$
$$= n_c(t)\cos^2\omega_c t - n_s(t)\sin\omega_c t\cos\omega_c t.$$
$$= \frac{n_c(t)}{2}[1+\cos 2\omega_c t] - \frac{1}{2}n_s(t)\sin 2\omega_c t$$
$$n_d(t) = \frac{1}{2}n_c(t) + \frac{1}{2}n_c(t)\cos 2\omega_c t - \frac{1}{2}n_s(t)\sin 2\omega_c t.$$

- The noise signal $n_d(t)$ is passed through a LPF.
- The 2nd and 3rd terms centered near $\pm 2\omega_c$ are filtered out by the LPF, and hence the noise power $n_o(t)$ appearing at the o/p of the synch. det is only $\frac{1}{2}n_c(t)$ i.e., $n_o(t) = \frac{1}{2}n_c(t)$.

∴ the power density spectrum is proportional to the mean square value of the signal, the power density spectrum of $n_o(t)$ is related to the power density spectrum of $n_c(t)$ as follows.

$$S_{no}(w) = \frac{1}{4}S_{nc}(w)$$

for a DSBSC system $S_{nc}(w) = \eta$, $|f| \le f_m$

$$\Rightarrow S_{no}(w) = \frac{1}{4}\eta. = \eta/4, \quad |f| \le f_m.$$

The BW of the message signal at the o/p of the detector is $2f_m$.

$\therefore$ The o/p noise power N_o is given by

$$N_o = S_{no}(w) \times B.W.$$

$$N_o = \frac{\eta}{4} \times 2f_m = \frac{\eta f_m}{2}.$$

From eq (1) we have $N_i = \eta f_m$.

$$\therefore \frac{N_o}{N_i} = \frac{\eta f_m}{2} \cdot \frac{1}{\eta f_m} = \frac{1}{2}$$

Signal power:- The modd signal at the i/p of the detector is $f(t)\cos\omega_c t$.

$\therefore$ The i/p signal power S_i is the mean square value of $f(t)\cos\omega_c t$.

$$S_i = \overline{[f(t)\cos\omega_c t]^2} = \frac{1}{2}\overline{f^2(t)}$$

The signal voltage at the o/p of the detector is $\frac{1}{2}f(t)$

$\therefore$ The o/p signal power is the m.s. value of $\frac{1}{2}f(t)$

$$S_o = \overline{\left[\frac{1}{2}f(t)\right]^2} = \frac{1}{4}\overline{f^2(t)}$$

$$\frac{S_o}{S_i} = \frac{\overline{f^2(t)}}{4} \cdot \frac{2}{\overline{f^2(t)}} = \frac{1}{2}.$$

figure of merit γ is given by

$$\gamma = \frac{S_o/N_o}{S_i/N_i} = \frac{S_o}{N_o} \cdot \frac{N_i}{S_i} = \frac{S_o}{S_i} \cdot \frac{N_i}{N_o} = \frac{S_o/S_i}{N_o/N_i} = \frac{1/2}{1/2} = 1.$$

Thus the figure of merit for a DSBSC sys. is unity.

$\therefore$ The SNRs at the i/p and o/p of the det are identical.

and There is no improvement in SNR.

2. SSB-SC — Noise in SSB System

SSBSC system also uses synch det. at the receiver.

Noise power :—

$$S_{no}(w) = \frac{1}{4} S_{nc}(w)$$

$S_{nc}(w)$ for SSBSC signal is equal to $\eta/2$

$$S_{no}(w) = \frac{1}{4} \cdot \frac{\eta}{2} = \frac{\eta}{8}.$$

BW of the detected signal at the o/p of the detector is $2f_m$.

Both SSBSC and DSBSC produce identical spectra at the synchronous detector o/p.

In both cases the detected o/p is $\frac{1}{2}f(t)$ with identical spectra.

$$\therefore \boxed{N_o = S_{no}(w) \times B.W = \frac{\eta}{8} \times 2f_m = \frac{\eta f_m}{4}}$$

$$\boxed{N_i = \eta f_m.}$$

Signal power :—

The signal power at the i/p and o/p of the sync det in a SSBSC or x^n can be evaluated by observing the SSBSC spectrum.

[The signal power is proportional to the area under the power density spectrum. (parseval's theorem).

The power density spectrum of the signal is proportional to the square of the fourier transform.

The area under the power density spectrum of an SSBSC signal is same as the area under the power density spectrum $F(w)$]

$\therefore$ the power of the SSBSC signal is same as the power (ms value) of the baseband signal $f(t)$.

→ i.e. $\boxed{S_i = \overline{\phi^2_{SSB}(t)} = \overline{f^2(t)}}$

The det. o/p in the SSBSC system is $\frac{1}{2}f(t)$.

$$\therefore \boxed{S_o = \overline{\left[\frac{1}{2}f(t)\right]^2} = \frac{1}{4}\overline{f^2(t)}}$$

$$\therefore \gamma = \frac{\overline{f^2(t)}}{4\eta f_m} \Big/ \frac{\overline{f^2(t)}}{\eta f_m} = \frac{\overline{f^2(t)}}{\eta f_m} \cdot \frac{\eta f_m}{\overline{f^2(t)}} = 1$$

Noise in AM System :-
Amplitude Modulation system (envelope detector)

- In AM system, a large carrier is accompanied with the two sidebands.
- If $n_i(t)$ is the additive noise signal, the i/p to the detector is given by

$$\phi_{AM}(t) = [A + f(t)]\cos\omega_c t + n_i(t) = A\cos\omega_c t + f(t)\cos\omega_c t + n_i(t)$$

$\Rightarrow$ S_i = (mean square) ms value of carrier + ms value of sidebands

$$S_i = \frac{1}{2}\left[A^2 + \overline{f^2(t)}\right] \qquad N_i = \frac{\eta}{2} \cdot 2f_m = \eta f_m$$

- o/p power :- The o/p of the envelope detector will be the envelope of the AM signal.
- $\therefore$ To find the detected o/p, we need to find the envelope of $\phi_{AM}(t)$.
- Consider $\phi_{AM}(t)$;

$$\phi_{AM}(t) = [A + f(t)]\cos\omega_c t + n_c(t)\cos\omega_c t - n_s(t)\sin\omega_c t.$$

$$= [A + f(t) + n_c(t)]\cos\omega_c t - n_s(t)\sin\omega_c t.$$

$$= A(t)\cos\{\omega_c t + \psi(t)\}$$

- where

$$A(t) = \sqrt{[A + f(t) + n_c(t)]^2 + n_s^2(t)} \quad ; \text{ randomly time varying amplitude}$$

$$\psi(t) = \tan^{-1}\left[\frac{n_s(t)}{A + f(t) + n_c(t)}\right] \quad ; \quad \text{.. .. .. phase angle of } \phi_{AM}(t)$$

- The time varying amplitude $A(t)$ is the envelope of $\phi_{AM}(t)$ and hence the o/p of the envelope detector will be the envelope $A(t)$.
- The envelope $A(t)$ has both signal and noise components.

→ The noise performance depends on the relative magnitudes of the signal and noise.

- The analysis can be carried out for two cases
  (i) Small noise case (ii) Large noise case.

(i) Small noise case :-

In this case, noise is taken to be much smaller than the signal.

$$n_i(t) << [A + f(t)]$$

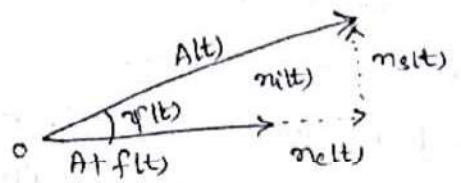


Fig. phasor diagram of the envelope $A(t)$

$n_c(t)$ is shown to be inphase with the signal $[A+f(t)]$

$n_s(t)$ is in phase quadrature.

$\therefore n_i(t) << [A+f(t)]$, $n_s(t)$ is also $<< [A+f(t)]$

It is evident from the phasor diagram that if $n_s(t)$ is much small, then $\psi(t)$ is also much small and may be taken as zero.

When $\psi(t) = 0$, $A(t) \Rightarrow A + f(t) + n_c(t)$.

$\therefore$ The o/p of the envelope detector contains a useful signal $f(t)$ and a noise component $n_c(t)$.

The carrier amplitude 'A' carries no useful information.

Output signal power S_o :- It is the mean square value of useful signal $f(t)$ i.e; $S_o = \overline{f^2(t)}$

Output noise power N_o :- The noise signal $n_o(t)$ at the o/p of the detector is $n_c(t)$ with the power density given by

$$S_{no}(w) = S_{nc}(w)$$

As the AM system contains both side bands, $S_{nc}(w)$ of AM = $S_{nc}(w)$ of DSBSC $= \eta$.

$$N_o = S_{no}(w) \cdot (BW) = \eta \cdot 2f_m = 2\eta f_m.$$

$$\therefore \quad \gamma = \frac{\overline{f^2(t)}}{2\eta f_m} \cdot \frac{\eta f_m}{\frac{1}{2}(A^2 + \overline{f^2(t)})} = \frac{\overline{f^2(t)}}{A^2 + \overline{f^2(t)}}$$

$\Rightarrow$ The noise performance improves with a reduction in carrier amplitude A. and is maximum when A=0, which is eqt to the suppressed carrier AM system.

Thus AM system remains always inferior to the suppressed system from the point of view of noise performance.

The greatest value of γ that can be achieved will depend on the ~~max~~ minimum possible value of carrier amplitude A.

- The min. value of A = max. value of $f(t)$ for avoiding the over modⁿ to the envelope detector.
- $\therefore$ The best noise performance is achieved when carrier amplitude A is equal to the max. value of $f(t)$, i.e; for 100% modulation.

Synchronous detection of AM System:-

Even if the sync. det is used for AM demodulation, the noise performance is identical to the envelope detector. More over the sync det is both complex and costly; hence, it is never used for AM detection.

(ii) Large noise case :- In this case $n_c(t) >> [A+f(t)]$

∴ quadrature components of $n(t)$ are also much larger than $[A+f(t)]$.

Here the noise terms dominate, and the performance of the envelope detector is completely different from the low noise case.

The envelope of the modulated signal at the input of the detector is

$$A(t) = \sqrt{[A+f(t)+n_c(t)]^2 + n_s^2(t)}$$

$$= \sqrt{[A+f(t)]^2 + 2n_c(t)[A+f(t)] + n_c^2(t) + n_s^2(t)}$$

∵ the noise dominates over the signal, $[A+f(t)]^2$ is very small and may be ignored.

$$\Rightarrow A(t) = \sqrt{n_c^2(t) + n_s^2(t) + 2n_c(t)[A+f(t)]}$$

$$= \sqrt{[n_c^2(t) + n_s^2(t)]\left[1 + \frac{2n_c(t)[A+f(t)]}{n_c^2(t)+n_s^2(t)}\right]}$$

Let $R(t) = \sqrt{n_c^2(t) + n_s^2(t)}$; and $\theta(t) = \tan^{-1}\frac{n_s(t)}{n_c(t)}$

we get $A(t) = \sqrt{R^2(t)\left[1 + 2[A+f(t)]\frac{n_c(t)}{R^2(t)}\right]}$

$$= \sqrt{R^2(t)\left[1 + \frac{2[A+f(t)]}{R(t)} \cdot \frac{n_c(t)}{\sqrt{n_c^2(t)+n_s^2(t)}}\right]}$$

$$= \sqrt{R^2(t)\left[1 + \frac{2[A+f(t)]}{R(t)}\right]\cos\theta(t)}$$

$$= R(t)\left[1 + \frac{2[A+f(t)]}{R(t)}\cos\theta(t)\right]^{1/2}$$

∵ the noise component $R(t) >> 2\{A+f(t)\}$

$$\Rightarrow A(t) = R(t)\left[1 + \frac{1}{2}\frac{2\{A+f(t)\}}{R(t)}\cos\theta(t)\right]$$

$$A(t) = R(t) + A\cos\theta(t) + f(t)\cos\theta(t)$$

∴ the envelope $A(t)$ which appears at the o/p of the envelope detector has no exclusive term proportional to the baseband (modulating) signal $f(t)$. The term $f(t)$ is multiplied by a large noise term $\cos\theta(t)$ and hence $f(t)$ cannot be seperated from the noise.

As f(t) is completely mingled with noise, it carries no useful information.

The loss of the message f(t) in an envelope detector due to the presence of large noise is referred to as the threshold effect.

Threshold Effect in an Envelope Detector:-

We have seen that when a noise is large as compared to the signal at the i/p of the envelope detector, the detected o/p has a message signal completely mingled with the noise.

It means that if the (Si/Ni) is below a certain level called threshold level the noise dominates over the message signal.

Threshold is defined as the value of the (Si/Ni) ratio below which the (So/No) ratio deteriorates much more rapidly than the i/p SNR.

Threshold effect starts in an envelope det. whenever carrier power to noise power ratio approaches unity or less.

Threshold effect is the property of an env. det.

Noise in Angle modulation system:-

Noise in Angle Modulated Systems.

Frequency modulation:-

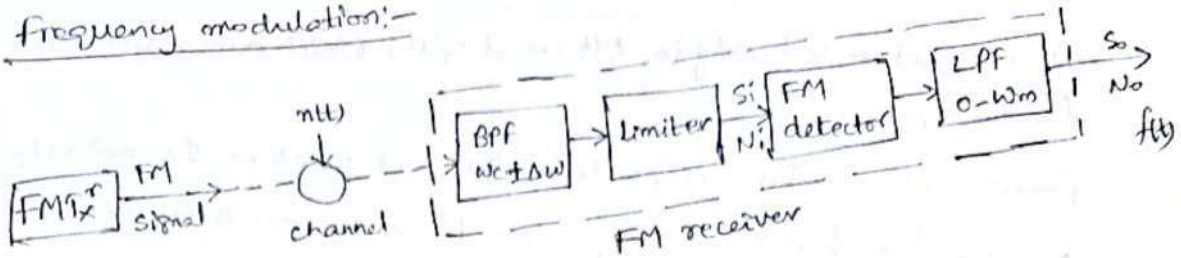


fig: FM sys.

The block diagram of FM system is shown in fig.

The BPF selects the desired carrier along with sidebands

The FM detector comprises a freq discriminator followed by the envelope det.

The ilp signal of the FM det is maintained constant by a limiter.

The olp of the receiver is the base band signal f(t).

Signal power:-

The power of the FM signal is same as that of an unmodd carrier, ∵ the amplitude of the signal remains unchanged after modⁿ.

$\therefore \quad S_i = \overline{(A\cos\omega_c t)^2} = \frac{A^2}{2}$. $\qquad S_i = \frac{A^2}{2}$

The olp power of the signal and noise can be calculated independently. i.e) for evaluating S_o, the noise is assumed to be absent, and vice-versa.

The olp of the detector $S_o(t) \propto w_i$ of the FM signal.

$$S_o(t) = K w_i$$

$$w_i = w_c + k_f f(t)$$

$\therefore S_o(t) = K w_c + K k_f f(t)$. in which the useful message signal is $K k_f f(t)$

The signal power due to this message term is

$$S_o = K^2 k_f^2 \overline{f^2(t)}$$

Noise power:- $N_i = \eta f_m$.

N_o is computed assuming $f(t) = 0$.

$\therefore$ input $g_i(t)$ to the detector will have a carrier term + noise term

$$g_i(t) = A\cos\omega_c t + n_i(t)$$

$$g_i(t) = A\cos\omega_c t + n_c(t)\cos\omega_c t - n_s(t)\sin\omega_c t$$

$$= [A + n_c(t)]\cos\omega_c t - n_s(t)\sin\omega_c t$$

$$g_i(t) = A(t)\cos\{\omega_c t + \psi(t)\} = A(t)\cos\theta(t)$$

where $A(t) = \sqrt{[A+n_c(t)]^2 + n_s^2(t)}$, $\psi(t) = \tan^{-1}\left\{\frac{n_s(t)}{A+n_c(t)}\right\}$, $\theta(t) = \omega_c t + \psi(t)$

∵ the o/p of the FM det depends on the inst freq $\omega_i = \frac{d}{dt}\theta(t)$, $A(t)$ is not of our interest.

We are interested only in $\theta(t)$ and $\psi(t)$ which are responsible for the FM det o/p.

Again the analysis depends on the fact whether the noise signal is small or large in comparison to the carrier amplitude.

(i) small noise case $A >> n_c(t)$ and $n_s(t)$

(ii) large noise case $A << n_c(t)$ and $n_s(t)$

Small noise case:- $A >> n_c(t)$ and $n_s(t)$

$\therefore \psi(t) \cong \tan^{-1}\left(\frac{n_s(t)}{A}\right)$; [$\frac{n_s(t)}{A}$ is small $\tan\psi(t)$ = small $\psi(t)$ is small.]

for small angle ϕ, $\tan\phi = \phi$; $\therefore \psi(t) = \frac{n_s(t)}{A}$

$\therefore \theta(t) = \omega_c t + \frac{n_s(t)}{A}$

$\omega_i = \frac{d}{dt}\theta(t) = \omega_c + \frac{\dot{n}_s(t)}{A}$

where $\dot{n}_s(t) = \frac{d}{dt} n_s(t)$.

The detected o/p $e_d(t) \propto \omega_i$, hence

$e_d(t) = K\omega_i = K\left[\omega_c + \frac{\dot{n}_s(t)}{A}\right] = K\omega_c + \frac{K\dot{n}_s(t)}{A}$

The noise part of this detected o/p is given as

$n_o(t) = \frac{K\dot{n}_s(t)}{A}$

$S_{no}(\omega) = \frac{K^2}{A^2} S_{\dot{n}s}(\omega)$

$S_n(\omega) = F\left\{\lim_{T\to\infty} \frac{1}{2T}\int_{-T}^{T} n^2(t)\,dt\right\}$

$x'(t) \xleftrightarrow{FT} j\omega X(\omega)$

Assuming $n_s(t)$ has the power density spectrum $S_{ns}(\omega)$ then $\dot{n}_s(t)$ will have its p.d.s. $S_{\dot{n}s}(\omega) = \omega^2 S_{ns}(\omega)$

$\therefore S_{no}(\omega) = \frac{K^2}{A^2} \cdot \omega^2 S_{ns}(\omega)$.

The detected o/p is bandlimited to ω_m. ∴ The above eqn holds only for $|\omega| \le \omega_m$ and is zero elsewhere.

The value of $S_{ns}(\omega)$ for a white noise is similar to the DSBSC case ie, $S_{ns}(\omega) = \eta$; $|\omega| \le \omega_m$.

$\therefore S_{no}(\omega) = \frac{K^2\omega^2\eta}{A^2}$, $|\omega| \le \omega_m$.

$= 0$; elsewhere.

- $r(t)$ = resultant of $n_c(t)$ and $n_s(t)$

Threshold Effect in angle modulation:-

- The interfering noise vector $r(t)$ modulates the carrier both in amplitude and in angle.
- The amplitude variation in $A(t)$ doesn't produce any adverse effect in the FM receiver, as it is not responded to by the FM receiver due to the features of limiter, or ratio detector.
- However the angle fluctuation produces a phase deviation of $\pm \psi(t)$ about the carrier phasor.
- The phase deviation is responded by FM receiver.
- As the noise signal $r(t)$ increases in comparison to carrier, the phase deviation $\psi(t)$ also increases
- This increases the distortion produced by noise.
- This type of distortion may be caused by any type of interfering signal accompanying the FM signal.

Capture effect:-

- The phase deviation produced by the noise signal in FM is much smaller than the " " " " modg " , provided that the noise is smaller than the carrier.
- Thus, in the low noise case the distortion produced by the noise at the o/p of the FM detector is negligable in comparison to the desired modg signal.
- In other words, noise is almost suppressed by the signal.
- The noise suppression char. of FM can also be applied to any interfering signal of almost same freq. This phenomena, is known as capture effect

(*) Def:- When FM signals from two transmitters operated on the same or nearly same carrier freq reach the receiver simultaneously, the signal of a weak magnitude is suppressed by a strong signal, and the FM receiver reproduces only the strong signal.

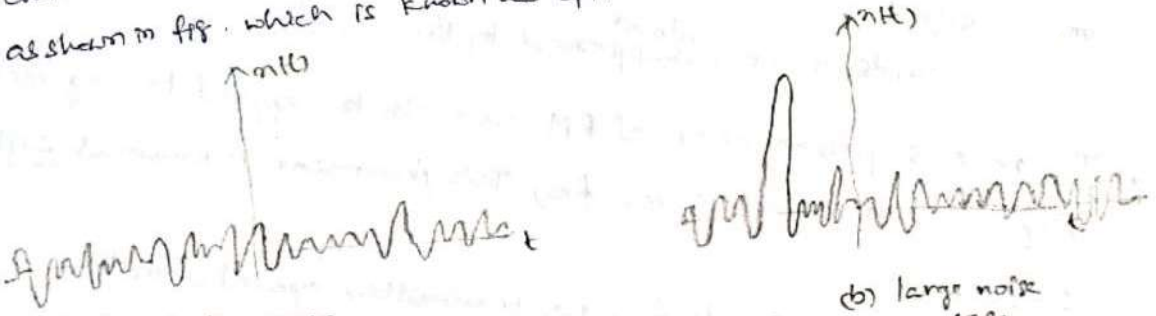
→ This effect is similar to noise suppressing action, where the weaker signal plays the role of the noise.

→ The weak signals (due to common channel & adjacent channel interferences) are suppressed.

- This is a very useful feature of FM system.
- This feature is not present in AM system.

Large noise case:-

- we know that $\gamma_{FM} = \frac{S_o/N_o}{S_i/N_i} = \frac{3}{2} m_f^2$
- under a low noise case, the noise performance ($\propto m_f \propto BW$) in FM goes on improving as the modⁿ index and BW of FM is increased.
- This improvement doesnot continue for an infinitely large increase in BW.
- This is because the i/p noise power introduced by the ckt components and devices also increases with bandwidth (Δf), and it reaches a point when the noise signal $r(t)$ becomes higher than the carrier amplitude A.
- Then $\frac{S_i}{N_i}$ becomes less than 1.
- It is observed experimentally that when a CNR (Carrier to Noise ratio) becomes even slightly less than unity, an impulse of noise is generated. CNR<1, N↑
- This noise impulse appears at the o/p of the FM discriminator in the form of a "click" sound.
- The discriminator o/p noise is shown below.
- For low noise case, the instantaneous ckt noise $n(t)$ has the gaussian probability density.
- This type of noise is known as smooth noise and is always suppressed by the message signal.
- But when noise is large and $\frac{C}{N}$ ratio becomes slightly less than unity, even momentarily, a noise impulse is superimposed on the smooth noise as shown in fig. which is known as spike noise or impulse noise.



(a) low noise case (b) large noise case

ckt noise at discriminator o/p

Threshold Effect in FM (angle modulation):-

- When the $\frac{C}{N}$ ratio is slightly less than unity, the freq of spike generation is small, and each spike produces individual clicking sound in the receiver.

- But when the $\frac{C}{N}$ ratio is further decreased so that the ratio is moderately less than unity, the spikes are generated rapidly and the clicks merge into a sputtering sound.
- This phenomena is known as "threshold effect in FM.

* The minimum $\frac{C}{N}$ ratio for which the FM noise improvement is not deteriorated significantly as compared to the small noise case is defined as "threshold".

- If the $\frac{C}{N}$ ratio is less than the threshold value, the sputter effect causes distortion in the FM receiver and the desired message signal cannot be resolved from the noise.

→ The threshold effect is more severe in FM than AM.

⇒ Threshold improvement in discriminators:-

- Threshold effect is more serious in FM because S/N at the i/p of a det, at which threshold effect starts, is higher.
- we know that lower threshold level (S/N) better is the system, because threshold can be avoided at a comparatively lower (S/N) ratio, and a small signal is needed to avoid threshold for an equivalent noise power.
- Hence, it is desirable to lower the threshold level in the FM receivers.

* The process of lowering the threshold level is known as threshold improvement or threshold reduction.

⇒ Methods used for the improvement of the threshold:

(i) pre emphasis and deemphasis ckts
(ii) FMFB (FM with feedback)
(iii) FM demodulator using PLL

Pre emphasis and De emphasis:-

- In this method simple RC n/ws are used to improve the threshold.
- We know that the noise power density spectrum increases as a parabolic fn with freq.
- Also a message signal has a decaying fourier spectrum with freq.
- As a result the So/No ratio at the o/p of the detector becomes very low towards the higher edge of the message band; and may cause threshold effect (inspite of the fact that So/No is large enough at lower edge of the message band.)
- Threshold effect may be avoided by improving So/No at the higher edge of the message band
- This is done by a simple RC n/w known as <u>preemphasis ckt</u> which boosts the signal amplitude of higher freqs in the message band before they modulate the carrier.
- The boosted signal at the t_x^r increases the So/No ratio at the detector o/p.
- Thus So/No ratio becomes large enough to improve the threshold level over the entire message band.
- However, the high freq components of the message signal reprod uced by the detector at a raised amplitude level, and ∴ amplitude distribution of the base band is **disturbed.**
- An inverse action is needed at the discriminator o/p to bring back the original level of high freq components and restore the amplitude distribution of message band.
- This is done by another RC n/w known as <u>decmphasis ckt</u>

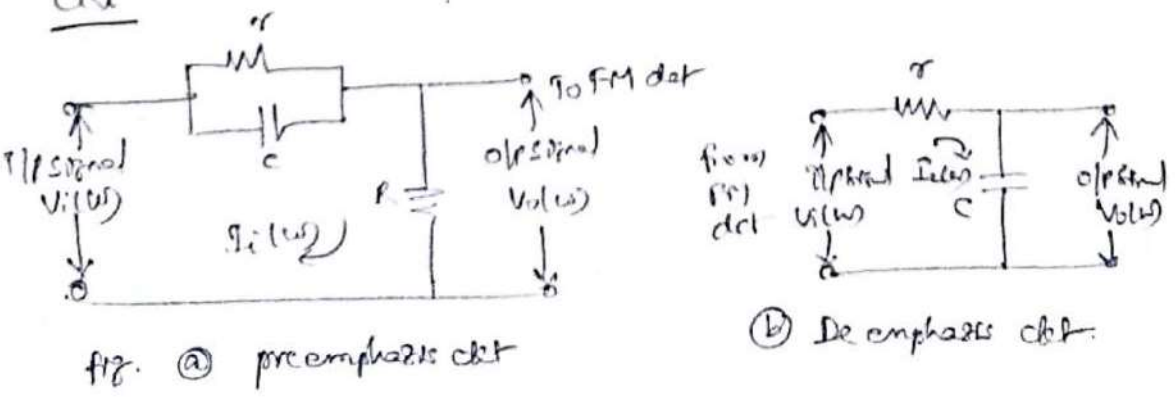


fig. (a) preemphasis ckt

(b) De emphasis ckt

- The preemphasis ckt at the tx is a high pass n/w which behaves like a differentiator.
- The deemphasis ckt at the receiver is a low pass n/w which behaves like a integrator

Improvement factor:-

- The preemphasis ckt doesn't increase the transmitted signal power because it is a passive n/w.
- The improvement in the SNR at the detector o/p is because of the reduction in the noise power due to the deempha- sis ckt. (LPF) (HF)
- The deemphasis ckt reduces the HF components of the noise, and cuts down the overall noise because of its Low pass filtering characteristic.
- The improvement in (S_o/N_o), in effect, improves the threshold level.
- The extent of improvement in SNR depends upon the noise reduction by the deemphasis ckt. It is defined as

(*) Improvement factor $= \frac{\text{mean No without deemphasis } (N_o)}{\text{mean No with deemphasis } (N_o)_d}$.

Ex6.8] Single tone modulation

S&S.

A single tone modg signal $E_m \cos\omega_m t$, freq modulates a carrier $A\cos\omega_c t$.

ST. (a) the detector (S_o/N_o) is proportional to the square of BW of the FM signal

(b) The figure of merit γ is given by $\gamma_{FM} = (3/2)\, m_f^2$

Sol^n: (a) $f(t) = E_m \cos\omega_m t$

$K_f f(t) = K_f E_m \cos\omega_m t$

The mean square value of both the sides yields

$$K_f^2 \overline{f^2(t)} = K_f^2 \frac{E_m^2}{2}$$

we know that $K_f E_m = \Delta\omega$

$$K_f^2 \overline{f^2(t)} = \frac{(\Delta\omega)^2}{2}$$

$$\frac{S_o}{N_o} = \frac{3\pi K_f^2 \overline{f^2(t)} A^2}{\eta \omega_m^3} = \frac{3\pi}{\eta}\frac{(\Delta\omega)^2}{2}\frac{A^2}{\omega_m^3} = 3\pi\left(\frac{\Delta\omega}{\omega_m}\right)^2 \frac{(A^2/2)}{2\pi f_m \eta} = \frac{3}{2}\left(\frac{\Delta\omega}{\omega_m}\right)^2 \frac{A^2/2}{\eta f_m}$$

This shows that (S_o/N_o) is proportional to $(\Delta\omega)^2$

Now the bandwidth of FM is twice the deviation $\Delta\omega$, and hence the (S_o/N_o) ratio is proportional to the square of bandwidth of the FM signal.

(b) By definition $\frac{\Delta\omega}{\omega_m} = m_f$

$$\frac{S_o}{N_o} = \frac{3}{2} m_f^2 \frac{A^2/2}{\eta f_m} = \frac{3}{2} m_f^2 (S_i/N_i)$$

$$\therefore \gamma_{FM} = \frac{S_o/N_o}{S_i/N_i} = \frac{3}{2} m_f^2.$$

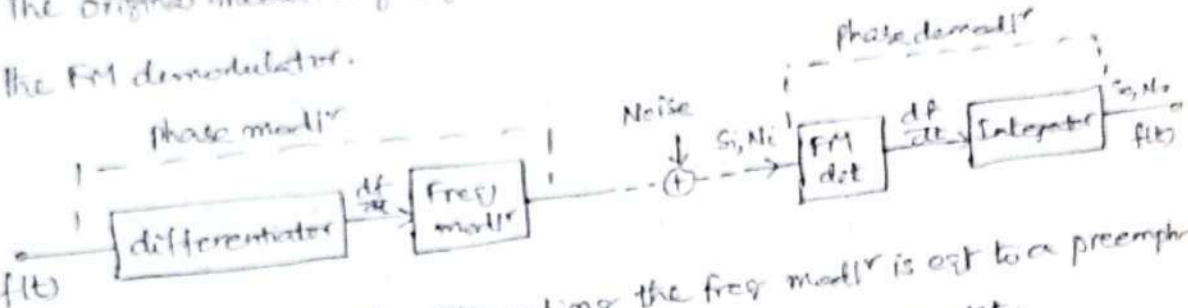
(*) $$\frac{S_o}{N_o} = \frac{3\pi K_f^2 \overline{f^2(t)} A^2}{\eta \omega_m^3} = \frac{3\pi A^2 K_f^2 (E_m^2/2)}{\eta 2\pi f_m \omega_m^2} = \frac{3\pi A^2}{2\eta 2\pi f_m}\left(\frac{K_f E_m}{\omega_m}\right)^2$$

$$\frac{S_o}{N_o} = \frac{3}{2}\left(\frac{\Delta\omega}{\omega_m}\right)^2 \frac{A^2}{2\eta f_m} = \frac{3}{2} m_f^2 (S_i/N_i)$$

$$\frac{S_o/N_o}{S_i/N_i} = \frac{3}{2} m_f^2 = \gamma \qquad \therefore \gamma_{FM} = \frac{3}{2} m_f^2$$

Noise in phase Modulation

- The phase modulation is equivalent to FM, except that $f(t)$ is first differentiated, and then it freq modulates the carrier. $f'(t) \rightarrow$ [frq mod^r] $\rightarrow$ PM (Actual)
- If we use an FM detector to demodulate the phase modulated signal, the demodulated signal will be a derivative of $f(t)$, i.e., $\frac{df}{dt}$.
- The original modulating signal can be obtained by integrating the o/p of the FM demodulator.



- The differentiator ckts preceding the freq modl^r is eqt to a preemphasis ckt, where as the integrator is eqt to a deemphasis ckt.
- These ckts are used to improve the noise performance of in FM receiver.
- Thus a FM sys equipped with the facility of preemphasis and deemphasis, is equivalent to a phase modulation system.
- obviously, FM has better noise performance as compared to phase modl^r.

(*) Figure of merit (γ_{PM}) :—

- $S_i = \frac{A^2}{2}$; $N_i = \eta f_m$.
- The signal power at the FM detector o/p is proportional to $\frac{df}{dt}$ and, after integrator, o/p becomes $f(t)$.

$$S_o = K^2 K_p^2 \overline{f^2(t)}$$

- $\|^{ly}$ $n_o(t) = \frac{K n_i(t)}{A}$; $n_i(t)$ = noise signal at the i/p of FM det
- After integration (assumed ideal), the noise power at the o/p of the PM detector is $n_o(t) = \frac{K n_i(t)}{A}$

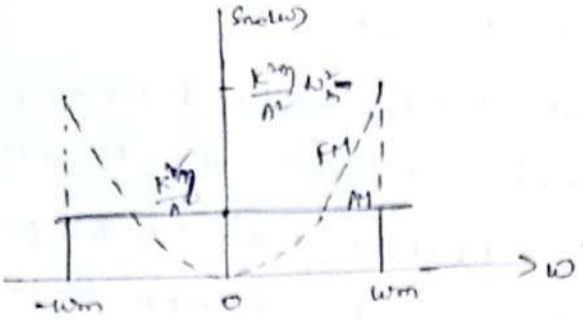
$$S_{no}(\omega) = \frac{K^2}{A^2} S_{ni}(\omega)$$

- $S_{ni}(\omega)$ for a white noise is given by $S_{ni}(\omega) = \eta$, $|\omega| \le \omega_m$

$$\therefore S_{no}(\omega) = \frac{K^2}{A^2}\eta \quad , |\omega| \le \omega_m$$
$$= 0 \quad , \text{elsewhere}$$

- In FM the power density has a quadratic relation with modg freq, where as in PM it is const. of freq as shown in fig.

[No dc term in $S_{no}(\omega)$]



$$N_o = \frac{1}{\pi}\int_0^{w_m} S_{no}(w)\,dw$$

$$N_o = \frac{1}{\pi}\int_0^{w_m} \frac{K^2\eta}{A^2}\,\eta\,dw = \frac{1}{\pi}\frac{K^2\eta}{A^2}\,w_m = \frac{2K^2\eta f_m}{A^2}$$

$$Y = \frac{K^2K_p^2\overline{f^2(t)}\,A^2}{2K^2\eta f_m} \Big/ \frac{A^2}{2\eta f_m} = \frac{K^2K_p^2\overline{f^2(t)}\,A^2}{2K^2\eta f_m}\cdot\frac{2\eta f_m}{A^2}$$

$$\boxed{Y = K_p^2\overline{f^2(t)}}$$

Ex 6.13 SWS: A single tone modg signal $f(t) = E_m\cos w_m t$ phase modulates a carrier $A\cos w_c t$. ST $Y_{pm} = \frac{1}{2}m_f^2$

Soln:- $f(t) = E_m\cos w_m t$

$K_p f(t) = K_p E_m \cos w_m t$

The value of both sides are given as $K_p^2\overline{f^2(t)} = K_p^2\frac{E_m^2}{2}$

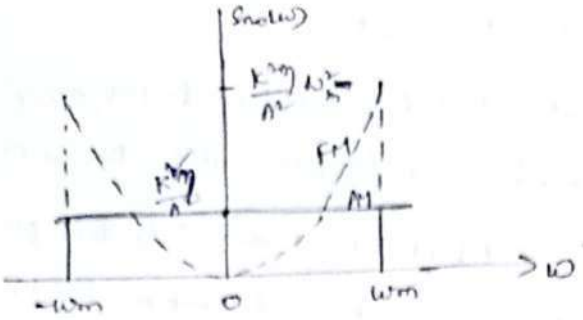
But the phase deviation $K_p E_m \left(= \frac{\Delta w}{w_m}\right) = m_f$

FM Eq(4), β = phase dev $= \frac{\Delta f}{f_m}$

$$\therefore K_p^2\overline{f^2(t)} = \frac{m_f^2}{2}$$

$$\Rightarrow \boxed{Y = \frac{m_f^2}{2}}.$$

Thus, the nature of noise figure dependance on m_f (hence BW) is identical to FM system.



$$N_o = \frac{1}{\pi}\int_0^{w_m} S_{no}(w)\,dw$$

$$N_o = \frac{1}{\pi}\int_0^{w_m} \frac{K^2\eta}{A^2}\,\eta\,dw = \frac{1}{\pi}\frac{K^2\eta}{A^2}\,w_m = \frac{2K^2\eta f_m}{A^2}$$

$$\gamma = \frac{K^2K_p^2\overline{f^2(t)}\,A^2}{2K^2\eta f_m} \Big/ \frac{A^2}{2\eta f_m} = \frac{K^2K_p^2\overline{f^2(t)}\,A^2}{2K^2\eta f_m}\cdot\frac{2\eta f_m}{A^2}$$

$$\boxed{\gamma = K_p^2\overline{f^2(t)}}$$

Ex 6.13 SWS: A single tone modg signal $f(t) = E_m\cos w_m t$ phase modulates a carrier $A\cos w_c t$. ST $\gamma_{pm} = \frac{1}{2}m_f^2$

Soln:- $f(t) = E_m\cos w_m t$

$K_p f(t) = K_p E_m \cos w_m t$

The value of both sides are given as $K_p^2\overline{f^2(t)} = K_p^2\frac{E_m^2}{2}$

But the phase deviation $K_p E_m \left(= \frac{\Delta w}{w_m}\right) = m_f$ (FM Eq(4), β = phase dev $= \frac{\Delta f}{f_m}$)

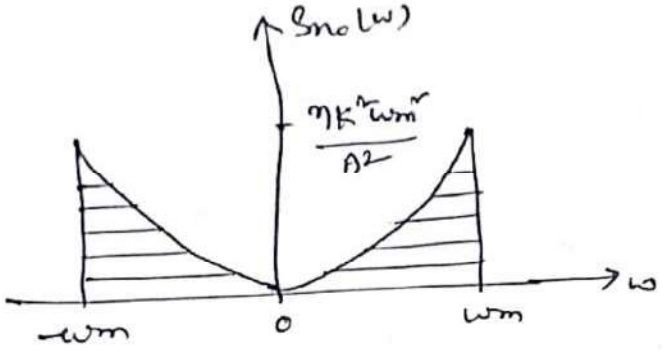
$$\therefore K_p^2\overline{f^2(t)} = \frac{m_f^2}{2}$$

$$\Rightarrow \boxed{\gamma = \frac{m_f^2}{2}}.$$

Thus, the nature of noise figure dependance on m_f (hence BW) is identical to FM system.

The spectrum $S_{no}(w)$ is shown in fig.

The spectrum is a parabolic fn of w.



$$N_o = \frac{1}{\pi}\int_0^{w_m} S_{no}(w)\,dw = \frac{K^2\eta}{\pi A^2}\int_0^{w_m} w^2\,dw \qquad \left(P_{av} = \frac{1}{\pi}\int_0^{\infty} S(w)\,dw\right)$$

$$= \frac{\eta K^2}{\pi A^2}\left|\frac{w^3}{3}\right|_0^{w_m} = \frac{\eta K^2 w_m^3}{3\pi A^2}.$$

Noise quieting effect: From the above eq. N_o of the FM detector varies inversely with carrier power $\frac{A^2}{2}$.

This decrease in noise power with an increase in carrier power is known as noise quieting effect of the carrier.

$$\frac{S_o}{N_o} = \frac{K_f^2\,\overline{f^2(t)}\cdot 3\pi A^2}{3\pi A\,\eta K^2 w_m^3} = \frac{3\pi K_f^2\,\overline{f^2(t)}\,A^2}{\eta w_m^3}$$

$$\frac{S_o}{N_o} = \frac{3\pi K_f^2\,\overline{f^2(t)}\,A^2}{\eta w_m^3}.$$

Figure of merit $\gamma = \dfrac{S_o/N_o}{S_i/N_i} = \dfrac{3\pi K_f^2\,\overline{f^2(t)}\,A^2}{\eta w_m^3}\cdot\dfrac{2\eta f_m}{A^2}$.

$$\boxed{\gamma_{FM} = \frac{3K_f^2\,\overline{f^2(t)}}{w_m^2}}$$

$$= \frac{3\pi K_f^2\,\overline{f^2(t)}\,A^2\cdot 2\eta f_m}{\eta\cdot\frac{8}{4}\pi^3 f_m^3\cdot A^2}$$

Noise Performance of FM as compared to AM :—

$$\gamma_{FM} = \frac{3}{2} m_f^2$$

for AM with a large carrier, $\gamma_{AM} = \frac{m_a^2}{2+m_a^2}$

the best noise performance is obtained for $m_a = 1$; $\gamma_{AM} = \frac{1}{2+1} = \frac{1}{3}$.

$$\therefore \frac{\gamma_{FM}}{\gamma_{AM}} = \frac{3 m_f^2}{2} \frac{3}{1} = \frac{9 m_f^2}{2}$$

The performance of FM and AM will be the same if RHS = 1

$$\frac{9 m_f^2}{2} = 1 \quad ; \quad m_f^2 = \frac{2}{9} \quad ; \quad m_f = \frac{\sqrt{2}}{3} \cong 0.5$$

This shows that FM has no improvement over AM when $m_f \leq 0.5$.

Thus $m_f = 0.5$ may be considered as a transition point between WBF & NBFM.

It should be recalled that NBFM and AM are identical in BW.

⇒ that the noise improvement depends on the BW.

∴ NBFM and AM have same BW, there is no improvement in noise performance of NBFM over AM.

The improvement in noise performance is at the expense of BW is as follows.

BW of FM signal $W_{FM} = 2(\Delta W) = 2 m_f W_m$ and

BW of AM signal $W_{AM} = 2 W_m$

Hence $\frac{W_{FM}}{W_{AM}} = m_f$

$$\therefore \frac{\gamma_{FM}}{\gamma_{AM}} = \frac{9}{2} m_f^2 = \frac{9}{2} \left(\frac{W_{FM}}{W_{AM}}\right)^2$$

This reveals that if a BW ratio is increased by a factor 2, the $(\gamma_{FM}/\gamma_{AM})$ increases by a factor 4 i.e; 6 dB which verifies the fact that noise performance is improved by increasing the BW.

ANALOG COMMUNICATION

By

D. RAJESH SETTY
Asst. Professor
DEPT. of E.C.E
RGMCET, NANDYAL

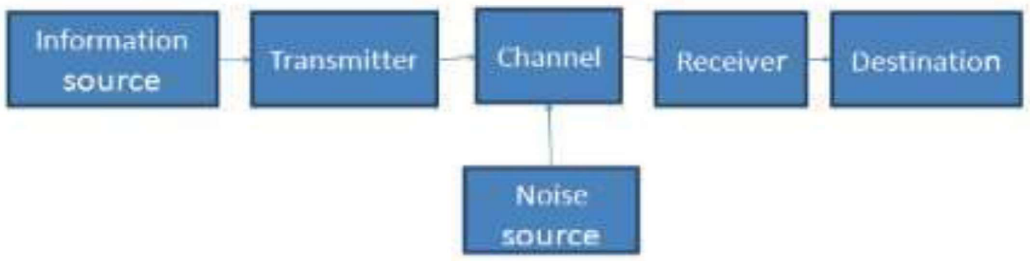
UNIT I

Introduction

Communication: It is the process of conveying or transferring information from one point to another. (Or)

It is the process of establishing connection or link between two points for information exchange.

Elements of Communication System:



Elements of Communication System:

Information source:

The message or information to be communicated originates in information source.

Message can be words, group of words, code, data, symbols, signals etc.

Transmitter :

The objective of the transmitter block is to collect the incoming message signal and modify it in a suitable fashion (if needed), such that, it can be transmitted via the chosen channel to the receiving point.

Elements of Communication System:

Channel :

Channel is the physical medium which connects the transmitter with that of the receiver.

The physical medium includes copper wire, coaxial cable, fibre optic cable, wave guide and free space or atmosphere.

Receiver:

The receiver block receives the incoming modified version of the message signal from the channel and processes it to recreate the original (non-electrical) form of the message signal.

Types of communication systems

Coaxial (Wired)
Microwave (Wireless)
Satellite (Wireless)
Cable (Wired)
Cellular (Wireless)

Frequency ranges of communication:

Table 1.1 *EM spectrum classified in terms different frequency ranges and corresponding wavelength ranges, nomenclature and typical application. The abbreviations in the table have the following values: 1 kHz = 1×10^3 Hz, 1 MHz = 1×10^6 Hz, 1 GHz = 1×10^9 Hz, 1THz = 1×10^{12} Hz, 1 μm = 1×10^{-3} m and 1 μm = 1×10^{-6}m.*

Frequency (f) range	Wavelength (λ) range	EM Spectrum Nomenclature	Typical Application
30 – 300 Hz	$10^7 - 10^6$ m	Extremely low frequency (ELF)	Power line communication
0.3 – 3 kHz	$10^6 - 10^5$ m	Voice frequency (VF)	Face to face speech communication Intercom
3 – 30 kHz	$10^5 - 10^4$ m	Very low frequency (VLF)	Submarine communication
30 – 300 kHz	$10^4 - 10^3$ m	Low frequency (LF)	Marine communication
0.3 – 3 MHz	$10^3 - 10^2$ m	Medium frequency (MF)	AM Broadcasting
3 – 30 MHz	$10^2 - 10^1$ m	High frequency (HF)	Landline Telephony
30 – 300 MHz	$10^1 - 10^0$ m	Very high frequency (VHF)	FM Broadcasting, TV
0.3 – 3 GHz	$10^0 - 10^{-1}$ m	Ultra high frequency (UHF)	TV, Cellular telephony
3 – 30 GHz	$10^{-1} - 10^{-2}$ m	Super high frequency (SHF)	Microwave oven, radar
30 – 300 GHz	$10^{-2} - 10^{-3}$ m	Extremely high frequency (EHF)	Satellite communication, radar
0.3 – 3 THz	0.1 – 1 mm	Experimental	For all new explorations
43 – 430 THz	7 – 0.7 μm	Infrared	LED, Laser, TV Remote
430 – 750 THz	0.7 – 0.4 μm	Visible light	Optical communication
750 – 3000 THz	0.4 – 0.1 μm	Ultravoilet	Medical application
> 3000 THz	< 0.1 μm	X-rays, gamma rays, cosmic rays	Medical application

Modulation

It is the process of varying the characteristics of high frequency carrier in accordance with instantaneous values of modulating or message or baseband signal.

(Or)

It is a frequency translation technique which converts baseband or low frequency signal to bandpass or high frequency signal.

Modulation is used in the transmitter.

Benefits or Need of Modulation

- To reduce the length or height of antenna
- For multiplexing
- For narrow banding or to use antenna with single or same length
- To reduce noise effect
- To avoid equipment limitation or to reduce the size of the equipment.

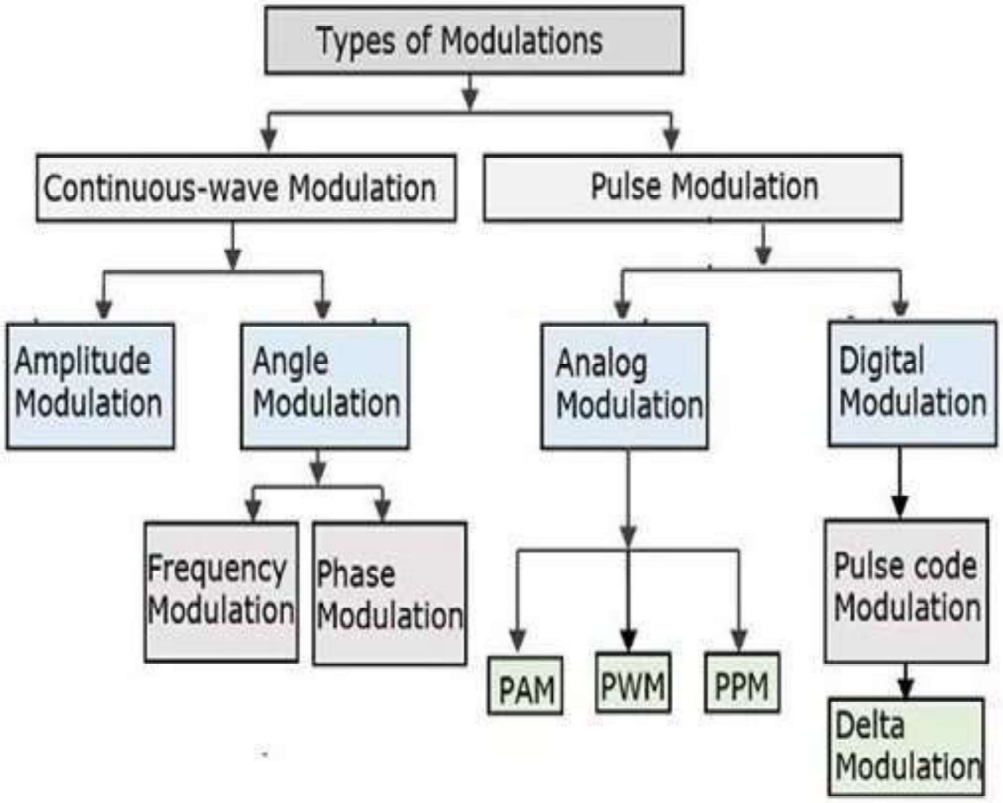
Types of Modulation

- **Amplitude Modulation**: Amplitude of the carrier is varied in accordance with the instantaneous values of modulating signal.

- **Frequency Modulation**: Frequency of the carrier is varied in accordance with the instantaneous values of modulating signal.

- **Phase Modulation**: Phase of the carrier is varied in accordance with the instantaneous values of modulating signal.

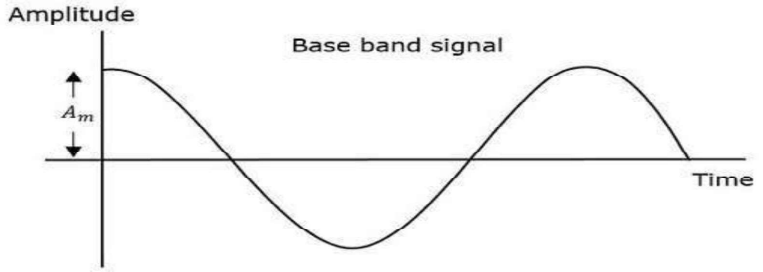
Types of Modulation

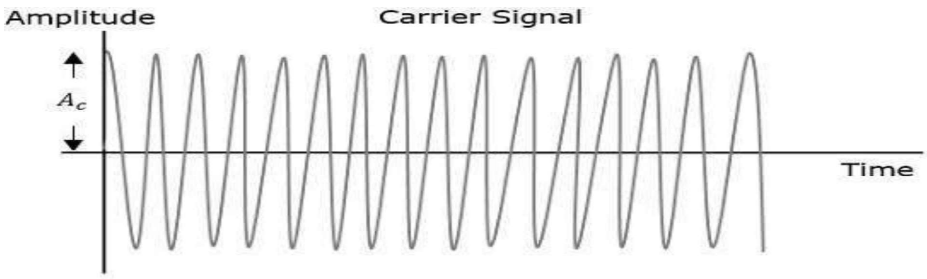


UNIT II

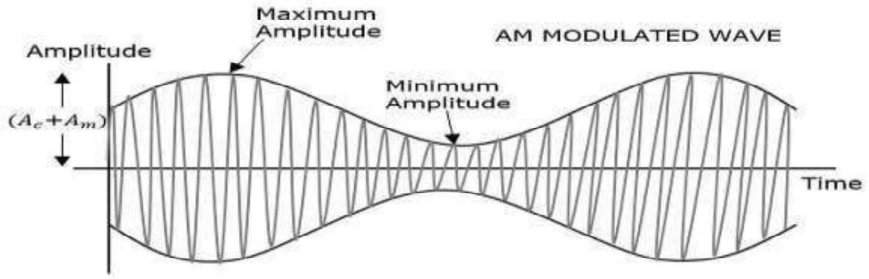
Amplitude Modulation

The amplitude of the carrier signal varies in accordance with the instantaneous amplitude of the modulating signal.





Amplitude Modulation



The carrier signal is given by,

$$C(t) = Ac\ Cosw_c t$$

Where, Ac= Maximum amplitude of the carrier signal.

W= 2πfc= Frequency of the carrier signal.

Modulating or baseband signal is given by,

$$X(t) = Am\ Cosw_m t$$

Where, Am = Amplitude of the baseband signal.

Amplitude Modulation

The standard equation for amplitude modulated signal is expressed as,

$$S(t)= Ac\ Cos2\pi f_c t[1+m_a(Cos2\pi f_m t)]$$

Where, $m_a = A_m/A_c$ = Modulation Index

Time Domain representation of AM:

$S(t)=AcCos2\pi f_c t+\mu Ac/2Cos[2\pi f_c+2\pi f_m]t+\mu Ac/2Cos[2\pi f_c-2\pi f_m]t$

I term: Carrier signal with amplitude Ac and frequency fc.

II. term: Amplitude= $\mu Ac/2$, frequency= f_c+f_m, Upper sideband frequency

III. term: Amplitude= $\mu Ac/2$, frequency= f_c-f_m, Lower sideband frequency

Amplitude Modulation

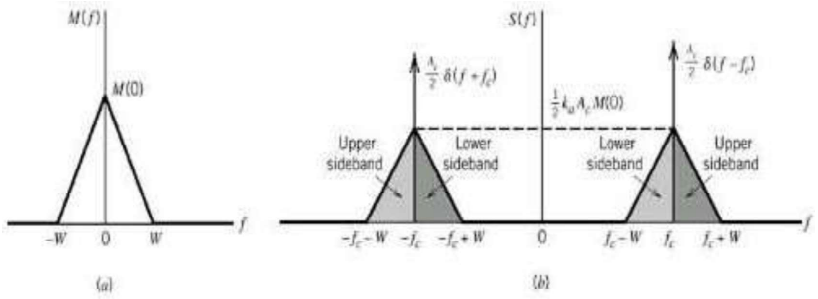
Frequency Domain representation of AM:

The time domain representation of AM wave is given by,

$$S(t)= Ac\ Cos2\pi f_c t[1+m_a(Cos2\pi f_m t)]$$

Taking Fourier transform on both sides,

$$S(f) = A_c/2[\delta(f-f_c)+ \delta(f+f_c)] + A_c m_a/2[M(f-f_c)+ M(f+f_c)]$$



Modulation Index

Modulation index or depth of modulation is given by,

$$m_a = [Amax-Amin/ Amax+Amin]= A_m/A_c$$

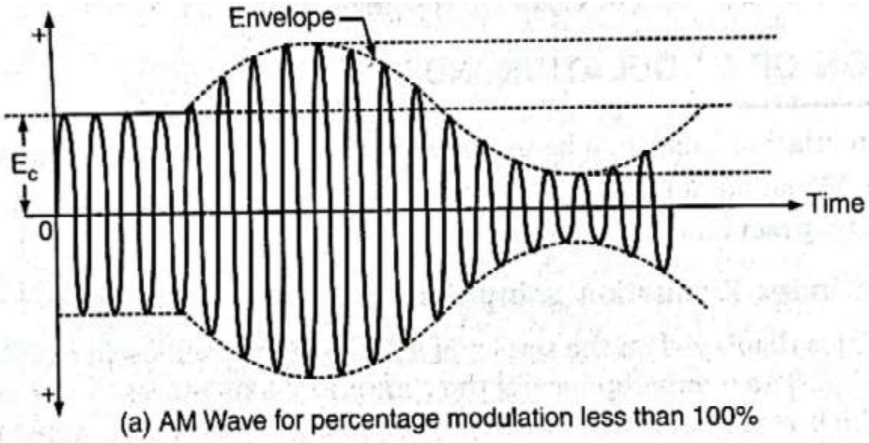
Percentage of modulation index is,

$$\%m_a = [Amax-Amin/ Amax+Amin]X100= [A_m/A_c]X100$$

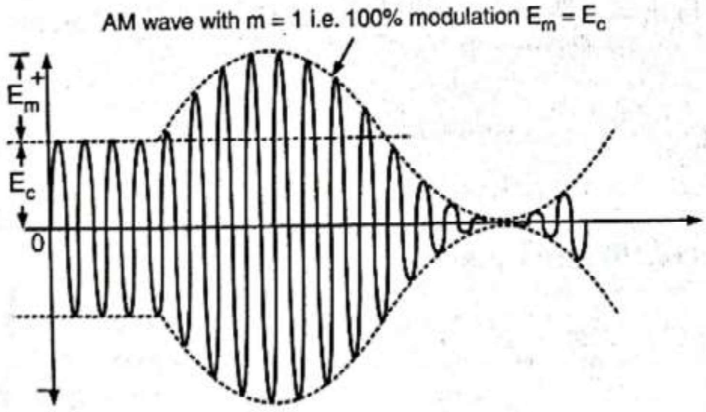
Types of AM with respect to modulation index:

- Under Modulation (m_a <1)
- Critical Modulation (m_a =1)
- Over Modulation (m_a >1)

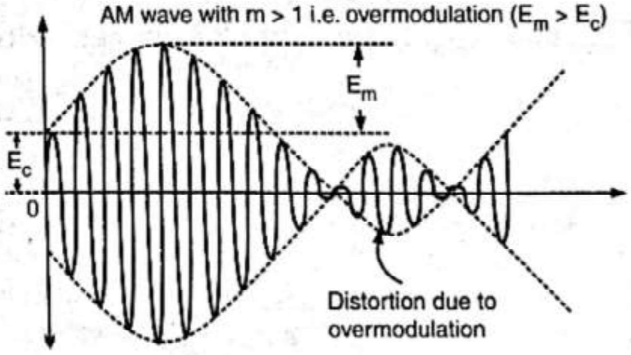
Types of AM



(a) AM Wave for percentage modulation less than 100%



(b) AM wave with 100% modulation



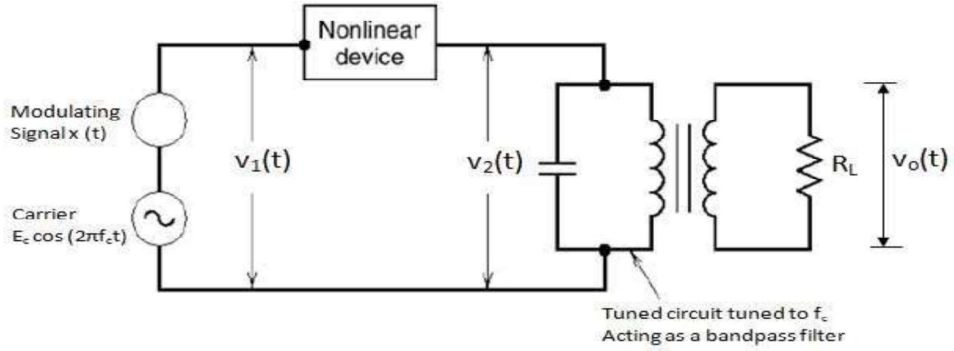
(c) AM wave with over modulation

Generation of AM Wave

Square Law modulator:

This circuit consists of,

- A non-linear device
- Band pass filter
- Carrier source and modulating signal



Generation of AM Wave

The modulating signal and carrier are connected in series with each other and their sum V1(t) is applied at the input of non-linear device such as diode or transistor.

$$V1(t) = x(t) + Ac \cos Wct \text{ --- (1)}$$

The input-output relation of non-linear device is,

$$V2(t) = aV1(t) + b\,V1^2(t) \text{ --- (2)}$$

Using (1) in (2),

$$V2(t) = a\,x(t) + a\,Ac \cos(2\pi fct) + bx^2(t) + 2bx(t)\,Ac \cos(2\pi fct) + b\,Ac^2 \cos^2(2\pi fct) \text{---(3)}$$

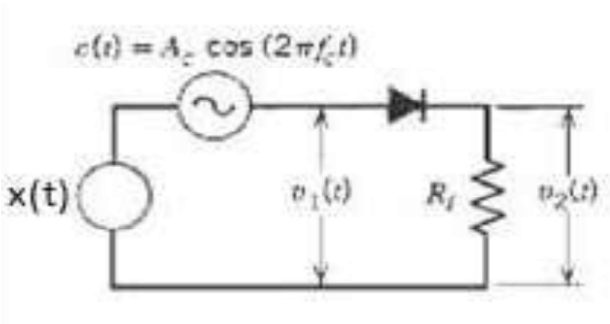
Out of these 5 terms, 1,3,5 terms are unuseful terms are eliminated by BPF.

Generation of AM Wave

Output of BPF is given by,

V0(t) = a Ac Cos (2πfct)+ 2bx(t) Ac Cos (2πfct)---(4)

Switching Modulator:



Generation of AM Wave

The carrier signal c(t) is connected in series with modulating signal x(t).

Sum of these two signals is passed through a diode.

Output of the diode is passed through a band pass filter and the result is an AM wave.

$$V1(t) = x(t) + c(t) \text{ ---(1)}$$

Amplitude of c(t) is much greater than x(t), so ON & OFF of diode is determined by c(t)

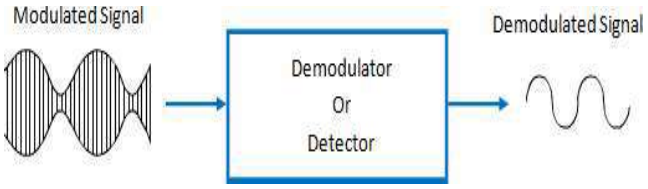
When c(t) is positive, V2(t) = V1(t) ---(2)

When c(t) is negative, V2(t) = 0 ---(3), Finally,

$$V2(t) = \frac{m(t)}{2} + \frac{Ac}{2}\text{Cos Wct} + \frac{2}{\pi}x(t)\text{Coswct} + \frac{2}{\pi}\text{Ac Cos2 wct}$$

Detection of AM Wave

Demodulation or detection is the process of recovering the original message signal from the received modulated signal.

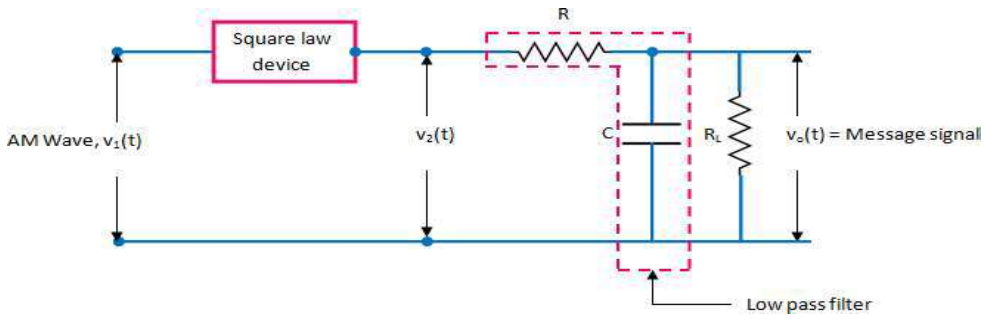


Types of AM Detectors:

1. Square Law detector
2. Envelope detector

Detection of AM Wave

Square Law detector:



The amplitude modulated wave is given as input to the square law device.

$$V2(t)= aV1(t) + b\ V1^2(t)\text{---}(1)$$

When this is passed through square law device,

$$V2(t) = aAcCoswct + aAcmx(t)Coswct+ bAc^2Cos^2wct+ 2bAc^2mx(t)Cos^2wct+ bAc^2m^2x^2(t)Cos^2wct\text{---}(2)$$

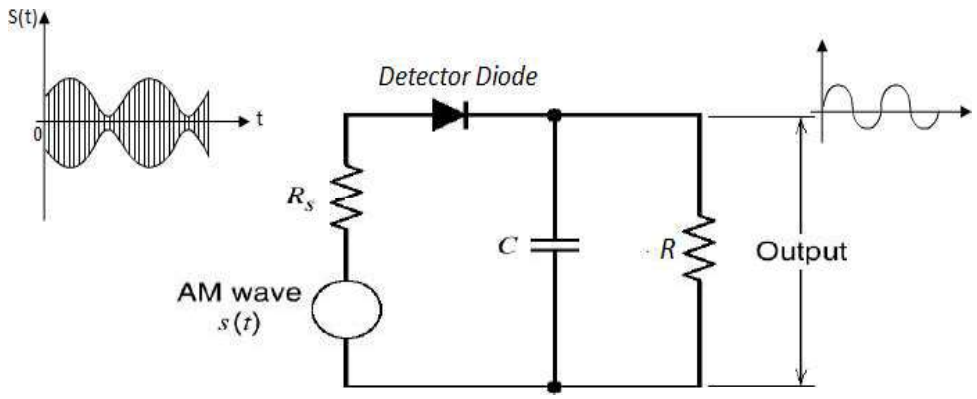
Detection of AM Wave

In order to extract the original message signal, V2(t) is passed through a low pass filter .

The output of LPF is,

$$V0(t) = mbAc^2x(t) \text{ ---(3)}$$

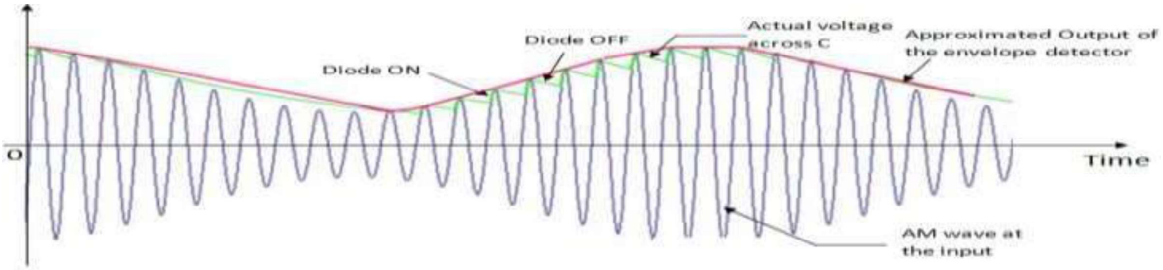
Envelope Detector:



Detection of AM Wave

- The standard AM wave is applied at the input of detector .
- In every positive half cycle of input, diode is forward biased which charges capacitor ‘C’.
- When capacitor charges to peak value of input voltage, diode stops conducting.
- The capacitor discharges through ‘R’ between positive peaks.
- This process continuous and capacitor charges and discharges repeatedly.

Detection of AM Wave



Double Sideband-Suppressed Carrier(DSB-SC)

The equation of AM wave in simple form is given by,

$$S(t) = Ac \cos wct + \frac{Ac}{2} Cos(wc + wm)t + \frac{Ac}{2} Cos(wc - wm)t$$

Here, carrier component remains constant and does not convey any information.

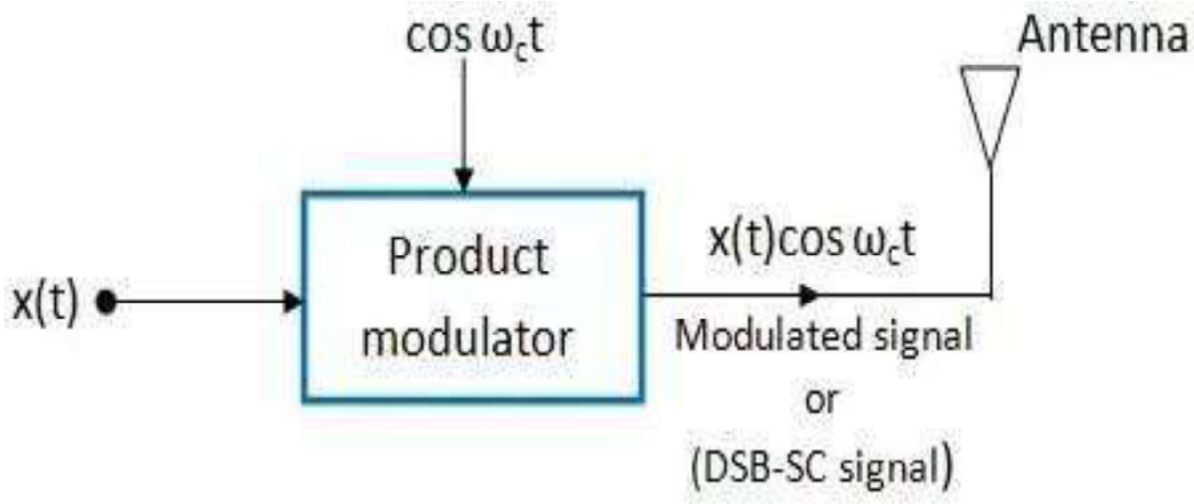
Therefore, if the carrier is suppressed, only sidebands remain in the spectrum requiring less power.

- DSB-SC Contains two side bands i.e USB & LSB
- Power efficiency is 100%
- % Power saving in DSB-SC w.r.t AM is 66.67%.

DSB-SC Modulation

A DSB-SC signal is obtained by multiplying the modulating signal x(t) with carrier signal c(t).

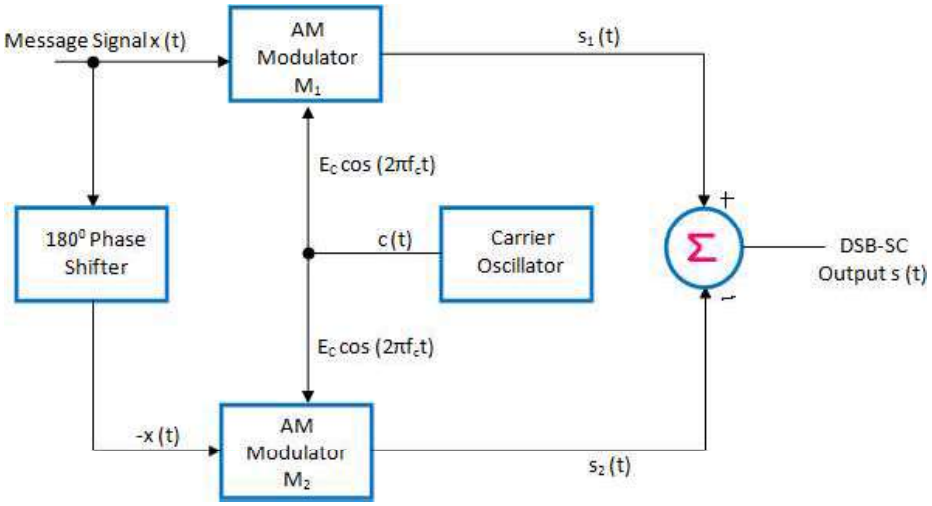
So, we need a product modulator for the generation of DSB-SC wave.



DSB-SC Modulation

1. Balanced Modulator:

It consists of two amplitude modulators arranged in balanced configuration to suppress the carrier completely.



DSB-SC Modulation

Operation:

- Carrier c(t) is applied to both the modulators.
- Message signal x(t) is applied directly to modulator 1 and with a phase shift of 180º to modulator 2.

Output of modulator 1 is,

$S1(t) = Ac[1+ mx(t)] \cos 2\pi fct$ ---(1)

Output of modulator 2 is,

$S2(t) = Ac[1- mx(t)] \cos 2\pi fct$ ---(2)

These two outputs are applied to subtractor, whose output is, $2mAcx(t) \cos 2\pi fct$---(3)

DSB-SC Modulation

2. **Ring Modulator**:

It operates in two modes

- Mode1: Without modulating signal x(t)
- Mode 2: With modulating signal x(t)

Mode1: **c(t) is positive**

- Diodes D1, D2 forward biased, D3,D4 Reverse biased
- Output of ring modulator will be zero.

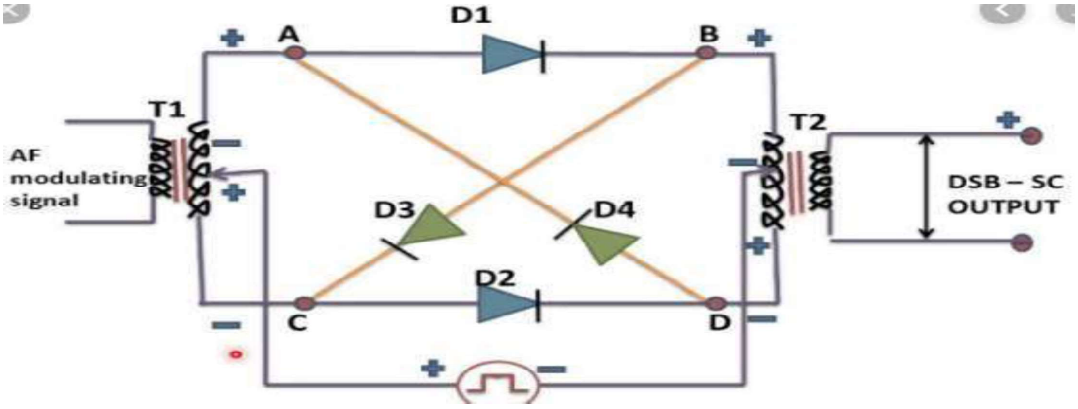
C(t) is negative

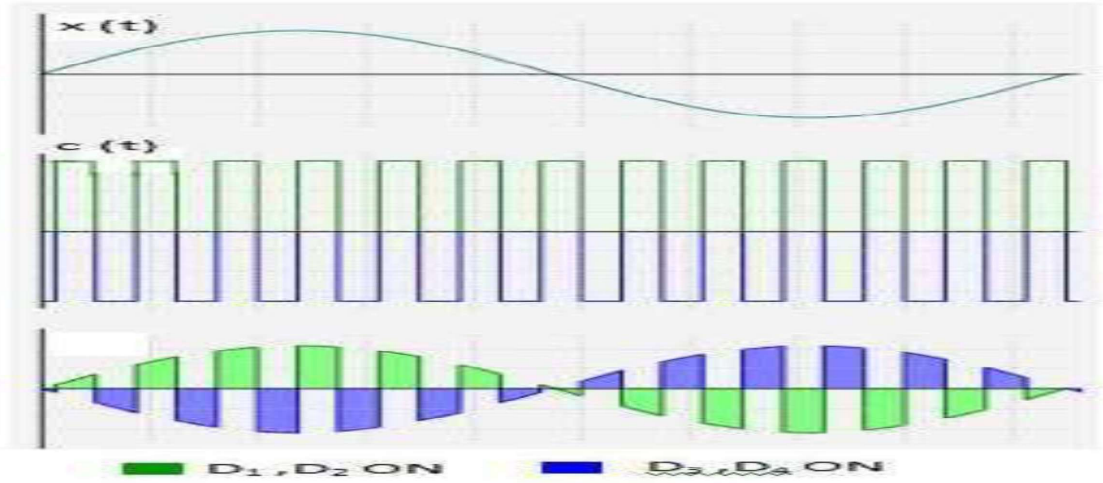
- Diodes D1, D2 reverse biased, D3,D4 forward biased
- Output of ring modulator will be zero.

Mode2:

- When modulating signal is present, during positive half cycle D1, D2 will be ON and secondary of T1 is directly applied to primary of T2.
- Output will be positive
- During negative half cycle of modulating signal D3, D4 will be ON producing positive voltage.

DSB-SC Modulation





Time Domain representation of DSB-SC

Message signal is given by,

$$x(t) = Am \cos(2\pi fmt) \text{ ---(1)}$$

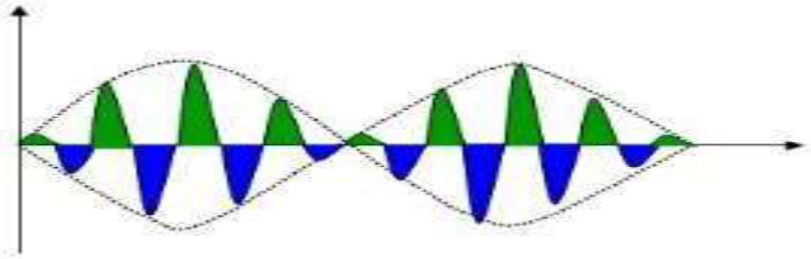
Carrier signal is given by,

$$C(t) = Ac \cos(2\pi fct) \text{ ---(2)}$$

DSB-SC modulated signal is given by,

$$S(t) = x(t)\, c(t) \text{ ---(3)}$$

$$S(t) = 1/2AmAc[\cos2\pi(fc+fm)t + \cos2\pi(fc-fm)t] \text{--(4)}$$



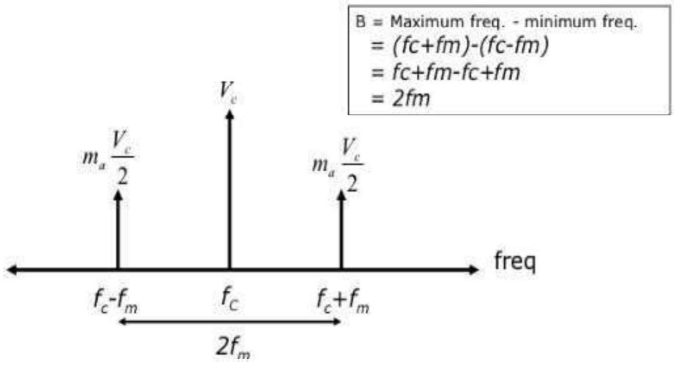
Frequency Domain representation of DSB-SC

The frequency spectrum of DSB-SC is obtained by taking Fourier transform of s(t)

S(f) = F{[1/2AmAc[cos2π(fc+fm)t + cos2π(fc-fm)t]}

S(f) = $\frac{1}{4}AmAc[\delta(f+fc+fm)+\delta(f-fc-fm)+\delta(f+fc-fm)+\delta(f-fc+fm)$

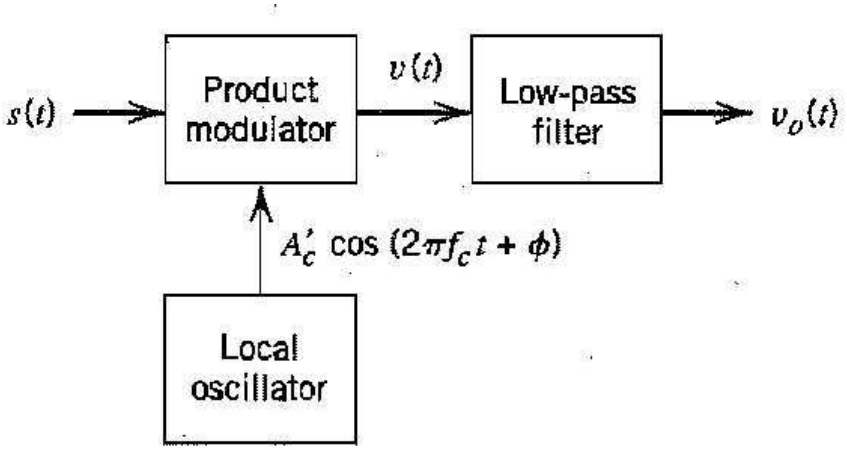
This is the spectrum of DSB-SC wave.



Demodulation of DSB-SC

Coherent Detection:

The modulating signal x(t) is recovered from DSB- SC wave s(t) by multiplying it with a locally generated carrier and then passing through a LPF.



Coherent detector for demodulating DSB-SC modulated wave.

Demodulation of DSB-SC

V(t) = s(t) c(t) ---(1)

Where,

S(t) = 1/2AmAc[cos2π(fc+fm)t + cos2π(fc-fm)t]—(2)

C(t) = cos2πfct ---(3)

Substituting (2) & (3) in (1)

$$m(t) = \frac{1}{4} VmVc[cos2\pi(2fc + fm)t + cos2\pi fmtcos2\pi(2fc - fm)t + cos2\pi fmt$$

When this is passed through a LPF,

V0(t) = $\frac{1}{4} VmVcos2\pi fmt$

Single Sideband-Suppressed Carrier(SSB-SC)

The modulation process in which only one side band is transmitted and with carrier suppression is called Single sideband suppressed carrier (SSB-SC).

Modulating Signal $m(t) = A_m \cos(2\pi f_m t)$ and Carrier Signal $c(t) = A_c \cos(2\pi f_c t)$

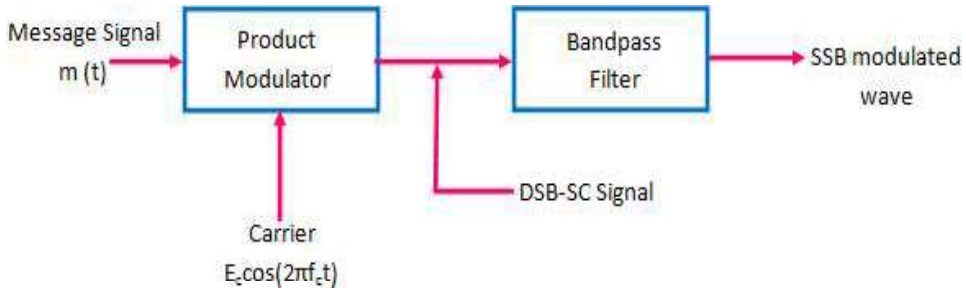
SSB-SC signal can be generated by passing DSB-SC signal through BPF. And DSB-SC signal is generated by multiplying m(t) & c(t).

$A_{SSB\text{-}SC}(t) = \frac{A_m A_c}{2} \cos 2\pi(f_c + f_m)t$ (or)

$A_{SSB\text{-}SC}(t) = \frac{A_m A_c}{2} \cos 2\pi(f_c - f_m)t$

Generation of SSB-SC

1. Filter or Frequency Discrimination Method:



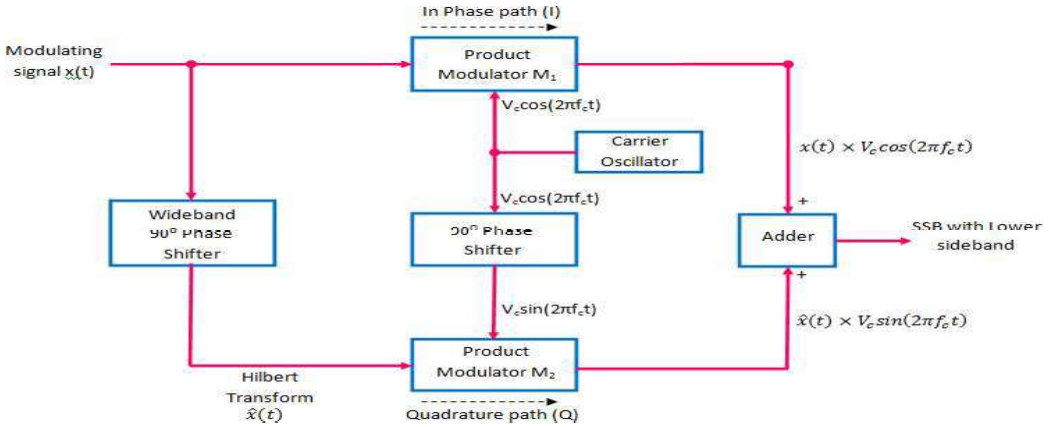
Filter method of generating DSB-SC Signal requires product modulator and BPF as shown in figure.

Here Product Modulator generates DSB-SC Signal which contains two side bands i.e USB & LSB.

By passing DSB-SC Signal through BPF either of sidebands are removed for generating SSB-SC Signal.

Generation of SSB-SC

2.Phase Shift or Phase Discrimination Method:



The figure shows the block diagram for the phase shift method of SSB generation and this system is used for the suppression of lower sideband.

This system uses two Product modulators M_1 and M_2 and two 90^o phase shifting networks.

Vestigial Sideband Transmission

- VSB-SC is used to transmit Video Signal which is large BW signal containing very low and very high frequency components.
- Very low Frequencies raise sidebands near to carrier frequency.
- It is not possible to suppress one complete sideband.
- Very low frequencies contain most of useful information, any effect to complete suppress the one sideband would result phase distortion at these frequencies.
- Therefore compromise has been made to suppress the part of sideband. Hence VSB-SC Signal contain one full sideband & part of other side band.

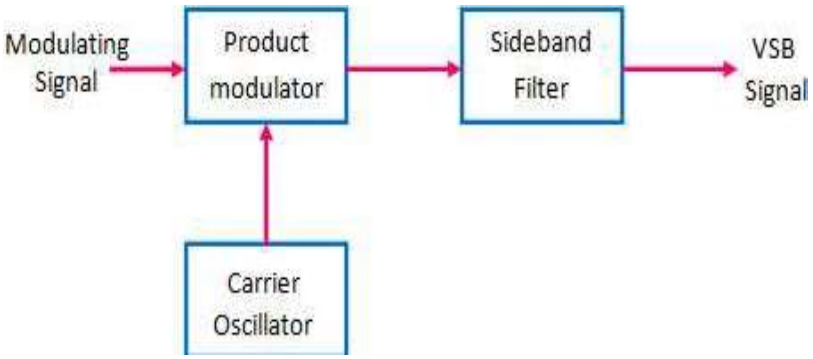
VSB Modulation & Demodulation

Modulation:

Modulating signal x(t) and carrier signal c(t) are applied as inputs to the product modulator.

$$S(t) = x(t)c(t)$$

This is the DSB-SC wave. It is applied to a side band filter which passes the wanted sideband completely and vestige of unwanted sideband.



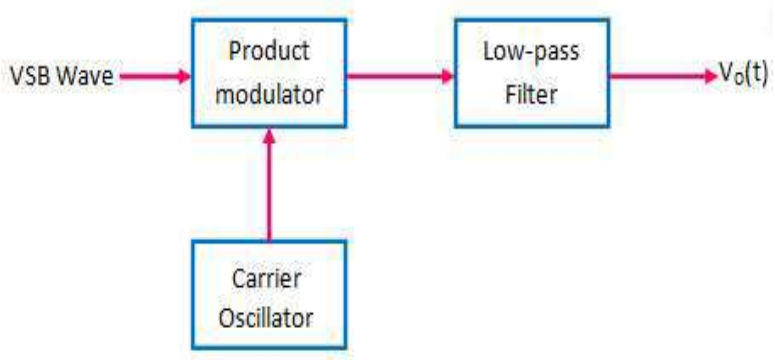
VSB Modulation & Demodulation

Demodulation:

The demodulation of VSB signal can be achieved using a coherent detector by multiplying s(t) with a locally generated carrier.

$$V(t) = s(t)AcCos2\pi fct$$

This signal is then passed through a LPF which passes low frequency message signal and rejects carrier.



UNIT III

ANGLE MODULATION

Angle modulation is a process of varying angle of the carrier in accordance with the instantaneous values of modulating signal.

Angle can be varied by varying frequency or phase.

Angle modulation is of 2 types.

- **Frequency Modulation**
- **Phase Modulation**

Frequency Modulation

The process of varying frequency of the carrier in accordance with the instantaneous values of the modulating signal.

Relation between angle and frequency :

Consider carrier signal c(t)= Ac Cos (wct+φ)

= Ac Cos (2πfct +φ)

Where, Wc= Carrier frequency

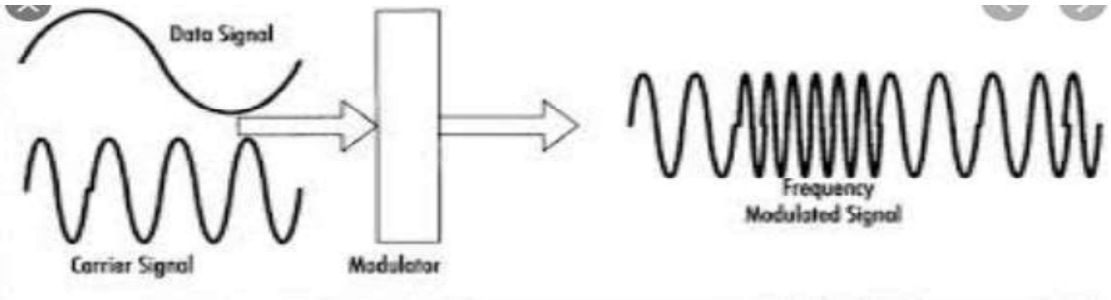
φ = Phase

C(t) = Ac Cos[ψ(t)], where, ψ(t)= wct+φ

$$\frac{d}{dt}\psi(t) = wc$$

i.e Frequency can be obtained by derivating angle and angle can be obtained by integrating frequency.

Frequency Modulation



Frequency modulator converts input voltage into frequency i.e the amplitude of modulating signal m(t) changes to frequency at the output.

Consider carrier signal c(t) =Ac Coswct

The frequency variation at the output is called instantaneous frequency and is expressed as,

$$w_i = w_c + k_f\, m(t)$$

Where, k_f = frequency sensitivity factor in Hz/volt

Frequency Modulation

The angle of the carrier after modulation can be written as,

$$\psi i(t) = \int wi\,dt = \int [wc + km(t)]\,dt$$

$$\psi i(t) = wct + kf\int m(t)\,dt$$

Frequency modulated signal can be written as,

$$A_{FM}(t) = Ac\ Cos\ [\psi_{i(}t)] = Ac\ Cos\ [w_c t + k_f \int m(t)dt]$$

Frequency Deviation in FM:

The instantaneous frequency, $wi = w_c + k_f\, m(t)$

$$= w_c + \Delta w$$

Where, $\Delta w = k_f\, m(t)$ **is called frequency deviation which may be positive or negative depending on the sign of m(t).**

Phase Modulation

The process of varying the phase of carrier in accordance with instantaneous values of the modulating signal.

Consider modulating signal x(t) and carrier signal c(t) = Ac Coswct

Phase modulating signal,

$$A_{PM}(t) = Ac\ Cos[\ \psi_i(t)]$$

Where, $\psi_i(t) = wct + k_p m(t)$

Where, k_p **= Phase sensitivity factor in rad/volt**

$$A_{PM}(t) = Ac\ Cos[wct + k_p m(t)]$$

Phase Modulation

Frequency deviation in PM:

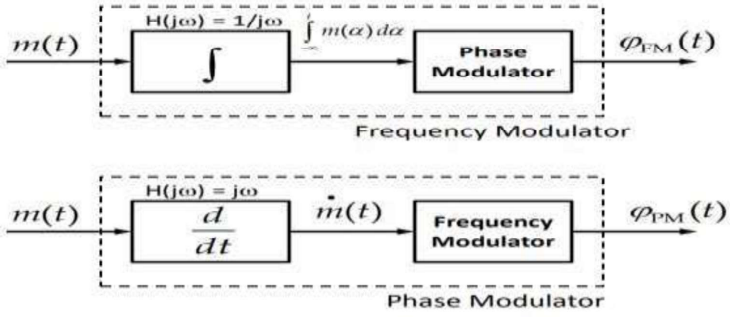
$$W_i = \frac{d}{dt}\psi i(t) = \frac{d}{dt}[wct + kpm(t)]$$

$$= wc + kp\frac{d}{dt}m(t)$$

$$Wc + \Delta w$$

Where, Δw = Frequency deviation= $kp\frac{d}{dt}m(t)$

Conversion between Frequency and Phase Modulation:



Modulation Index

Definition:

Modulation Index is defined as the ratio of frequency deviation (δ) to the modulating frequency (f_m).

$$\text{M.I.} = \frac{\text{Frequency Deviation}}{\text{Modulating Frequency}}$$

$$mf = \frac{\delta f}{fm}$$

In FM M.I.>1

Modulation Index of FM decides –
(i)Bandwidth of the FM wave.
(ii)Number of sidebands in FM wave.

Deviation Ratio

The modulation index corresponding to maximum deviation and maximum modulating frequency is called deviation ratio.

$$\text{Deviation Ratio} = \frac{\text{Maximum Deviation}}{\text{Maximum modulating Frequency}}$$

$$= \frac{\delta max}{fmax}$$

In FM broadcasting the maximum value of deviation is limited to **75 kHz.** The maximum modulating frequency is also limited to **15 kHz.**

Percentage M.I. of FM

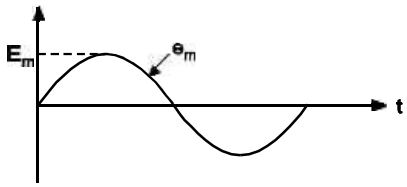
The percentage modulation is defined as the ratio of the actual frequency deviation produced by the modulating signal to the maximum allowable frequency deviation.

$$\% \text{ M.I} = \frac{\text{Actual deviation}}{\text{Maximum allowable deviation}}$$

$\therefore$

Mathematical Representation of FM

(i) Modulating Signal:



It may be represented as,

$$e_m = E_m \cos \omega_m t \quad \ldots(1)$$

Here cos term taken for simplicity

where,

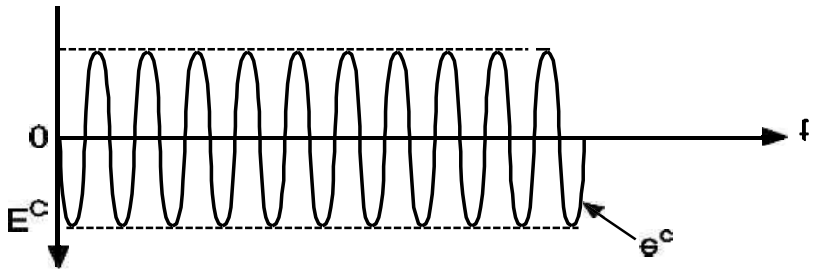
e_m = Instantaneous amplitude

ω_m = Angular velocity

= $2\pi f_m$

f_m = Modulating frequency

(ii) Carrier Signal:



Carrier may be represented as,

$$e_c = E_c \sin(\omega_{ct} + \phi) \quad \text{-----(2)}$$

where,

e_c = Instantaneous amplitude

ω_c = Angular velocity

= $2\pi f_c$

f_c = Carrier frequency

ϕ = Phase angle

(iii) FM Wave:

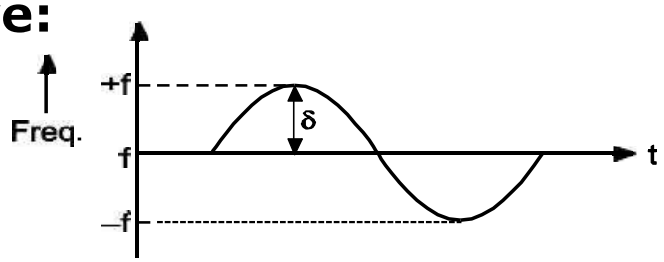


Fig. Frequency Vs. Time in FM

FM is nothing but a deviation of frequency.

From Fig. 2.25, it is seen that instantaneous frequency 'f' of the FM wave is given by,

$$\mathbf{f = f_c\,(1 + K\,E_m \cos \omega_m t)} \ldots (3)$$

where,

f_c =Unmodulated carrier frequency

K = Proportionality constant

$E_m \cos \omega_m t$ =Instantaneous modulating signal

(Cosine term preferred for simplicity otherwise we can use sine term also)

- The maximum deviation for this particular signal will occur, when $\cos \omega_m t = \pm 1$ i.e. maximum.

$\therefore$ Equation (2.26) becomes,

$$\mathbf{f = f_c\,(1 \pm K\,E_m)} \quad \ldots \mathbf{(4)}$$

$\therefore$ $$\mathbf{f = f_c \pm K\,E_m f_c} \quad \ldots \mathbf{(5)}$$

So that maximum deviation δ will be given by,

$$\delta = \mathbf{K E_m f_c} \quad \ldots (6)$$

The instantaneous amplitude of FM signal is given by,

$$e_{FM} = A \sin [f(\omega_c, \omega_m)]$$
$$= A \sin \theta \quad \ldots (7)$$

where,

$f(\omega_c, \omega_m)$= Some function of carrier and modulating frequencies

Let us write equation (2.26) in terms of ω as,

$$\omega = \mathbf{\omega_c (1 + K E_m \cos \omega_m t)}$$

To find θ, ω must be integrated with respect to time.

Thus,

$$\theta = \omega \, dt$$
$$= \omega_c (1 + K E_m \cos \omega_m t) \, dt$$
$$\theta = \omega_c (1 + K E_m \cos \omega_m t) \, dt$$
$$= \omega_c (t + KEm \frac{\sin \omega mt}{\omega m}$$
$$= \omega_c t + KEm\omega_c \frac{\sin \omega mt}{\omega m}$$
$$= \omega_c t + KEmf_c \frac{\sin \omega mt}{\omega m}$$

$= \omega_c t + \frac{\delta \sin\omega mt}{fm}$ $\quad [\because \quad \delta = K E_m f_c]$

$\therefore$

Substitute value of θ in equation (7)

Thus,

$e_{FM} = A \sin (\omega_c t + \frac{\delta}{fm} \sin\omega mt \ )$---(8)

$e_{FM} = A \sin (\omega_c t + mf \sin\omega mt \ )$---(9)

This is the equation of FM.

Frequency Spectrum of FM

Frequency spectrum is a graph of amplitude versus frequency. The frequency spectrum of FM wave tells us about number of sideband present in the FM wave and their amplitudes.

The expression for FM wave is not simple. It is complex because it is sine of sine function.

Only solution is to use **'Bessels Function'.**

Equation (2.32) may be expanded as,

$$\begin{aligned} e_{FM} = {} & \{A J_0 (m_f) \sin \omega_c t \\ & + J_1 (m_f) [\sin (\omega_c + \omega_m) t - \sin (\omega_c - \omega_m) t] \\ & + J_1 (m_f) [\sin (\omega_c + 2\omega_m) t + \sin (\omega_c - 2\omega_m) t] \\ & + J_3 (m_f) [\sin (\omega_c + 3\omega_m) t - \sin (\omega_c - 3\omega_m) t] \\ & + J_4 (m_f) [\sin (\omega_c + 4\omega_m) t + \sin (\omega_c - 4\omega_m) t] \\ & + \ldots\ldots\} \qquad \ldots (2.33) \end{aligned}$$

From this equation it is seen that the FM wave consists of:

(i)Carrier (First term in equation).

(ii)Infinite number of sidebands (All terms except first term are sidebands).

The amplitudes of carrier and sidebands depend on 'J' coefficient.

$\omega_c = 2\pi f_c$, $\omega_m = 2\pi f_m$

So in place of ω_c and ω_m, we can use f_c and f_m.

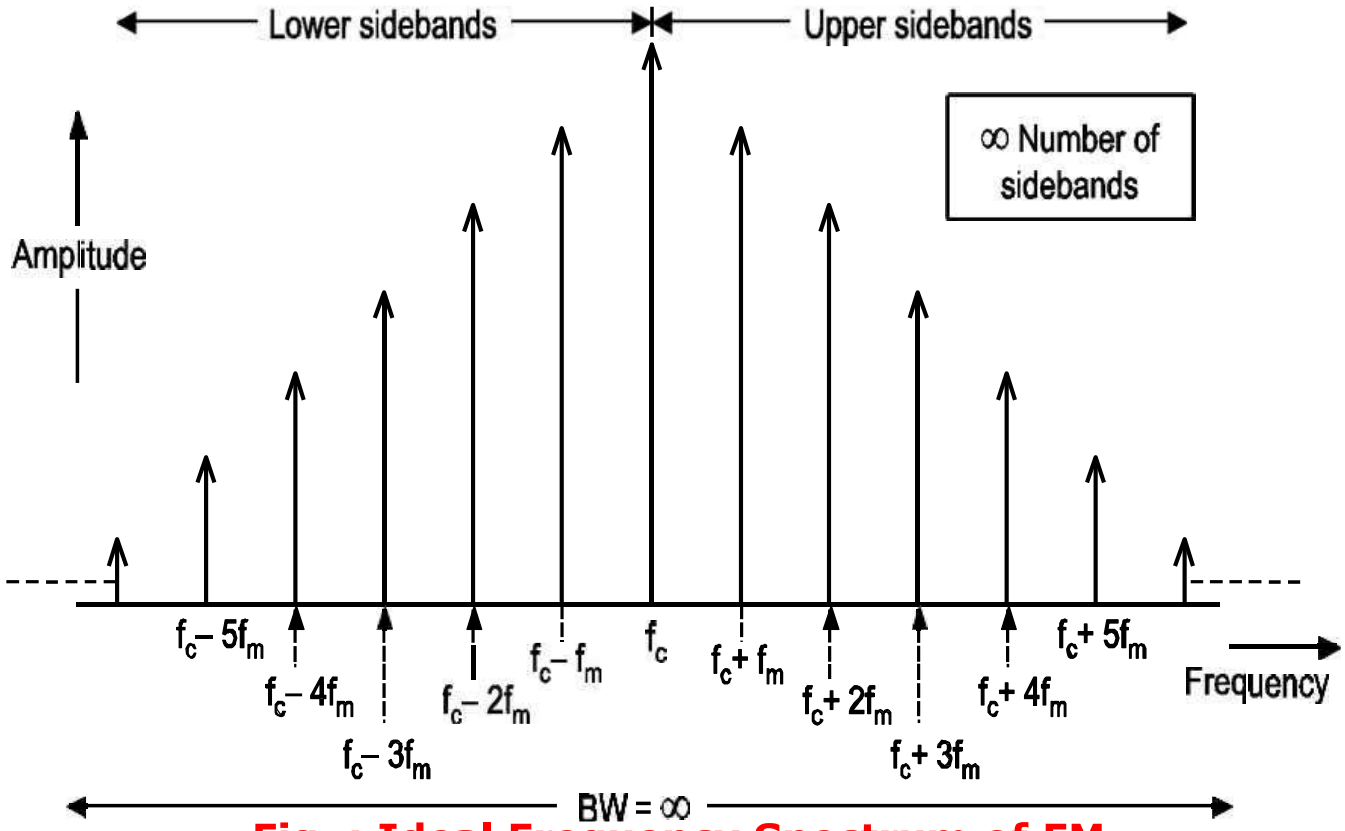


Fig. : Ideal Frequency Spectrum of FM

Bandwidth of FM

From frequency spectrum of FM wave shown in Fig. 2.26, we can say that the bandwidth of FM wave is infinite.

But practically, it is calculated based on how many sidebands have significant amplitudes.

(i)The Simple Method to calculate the bandwidth is –

$$BW = 2f_m \times \text{Number of significant sidebands} \quad \text{--(1)}$$

With increase in modulation index, the number of significant sidebands increases. So that bandwidth also increases.

(ii)The second method to calculate bandwidth is by **Carson's rule.**

Carson's rule states that, the bandwidth of FM wave is twice the sum of deviation and highest modulating frequency.

$$BW = 2(\delta + f_{mmax}) \quad \ldots(2)$$

Highest order side band = To be found from table 2.1 after the calculation of modulation Index m where, $m = \delta / f_m$

e.g. If m= 20KHZ/5KHZ

From table, for modulation index 4, highest order side band is 7^{th}.

Therefore, the bandwidth is

$$\begin{aligned} B.W. &= 2 f_m \times \text{Highest order side band} \\ &= 2 \times 5 \text{ kHz} \times 7 \\ &= 70 \text{ kHz} \end{aligned}$$

Types of Frequency Modulation

1. Narrowband FM (NBFM) [When modulation index is small]
2. Wideband FM (WBFM) [When modulation index is large]

Comparison between Narrowband and Wideband FM

Sr. No.	Parameter	NBFM	WBFM
1.	Modulation index	Less than or slightly greater than 1	Greater than 1
2.	Maximum deviation	5 kHz	75 kHz
3.	Range of modulating frequency	20 Hz to 3 kHz	20 Hz to 15 kHz
4.	Maximum modulation index	Slightly greater than 1	5 to 2500
5.	Bandwidth	Small approximately same as that of AM BW = $2f_m$	Large about 15 times greater than that of NBFM. BW = $2(\delta+fmmax)$
6.	Applications	FM mobile communication like police wireless, ambulance, short range ship to shore communication etc.	Entertainment broadcasting (can be used for high quality music transmission)

Representation of FM

FM can be represented by two ways:

1. Time domain.
2. Frequency domain.

1.FM in Time Domain

Time domain representation means continuous variation of voltage with respect to time as shown in Fig.

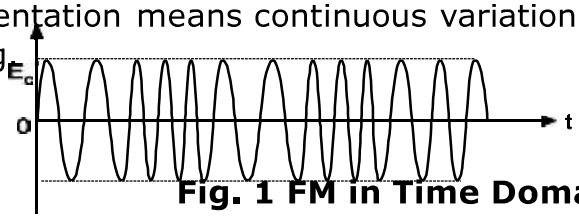


Fig. 1 FM in Time Domain

2.FM in Frequency Domain

- Frequency domain is also known as **frequency spectrum.**
- FM in frequency domain means graph or plot of amplitude versus frequency as shown in Fig. 2.29.

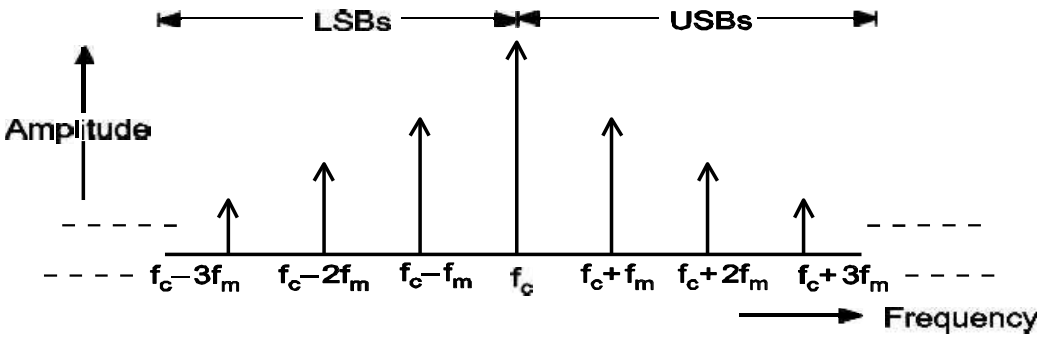
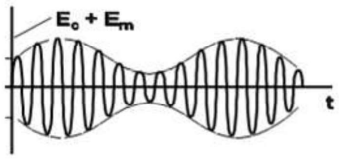
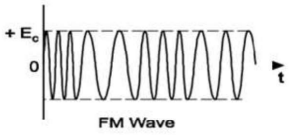


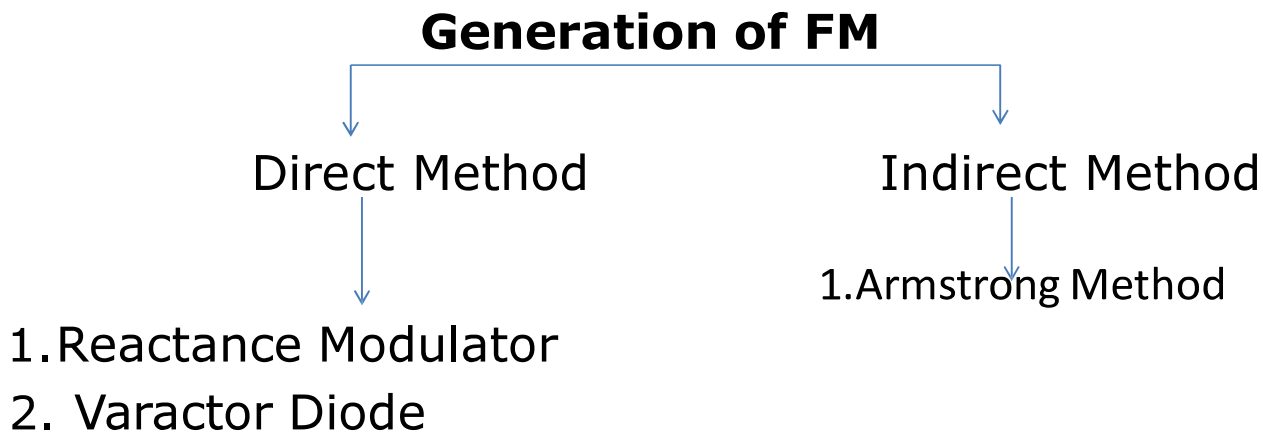
Fig. 2: FM in Frequency Domain

Comparison between AM and FM

Parameter	AM	FM
1. Definition	Amplitude of carrier is varied in accordance with amplitude of modulating signal keeping frequency and phase constant.	Frequency of carrier is varied in accordance with the amplitude of modulating signal keeping amplitude and phase constant.
2. Constant Parameters	Frequency and phase.	Amplitude and phase.
3. Modulated signal		
4. Modulation Index	m=Em/Ec	m = Δf/ fm
5. Number of Sidebands	Only two	Infinite and depends on m_f.
6. Bandwidth	BW = $2f_m$	BW = 2 (Δf + $f_{m\,(max)}$)
7. Application	MW, SW band broadcasting, video transmission in TV.	Broadcasting FM, audio transmission in TV.

FM GENERATION

There are two methods for generation of FM wave.



Varactor Diode Modulator

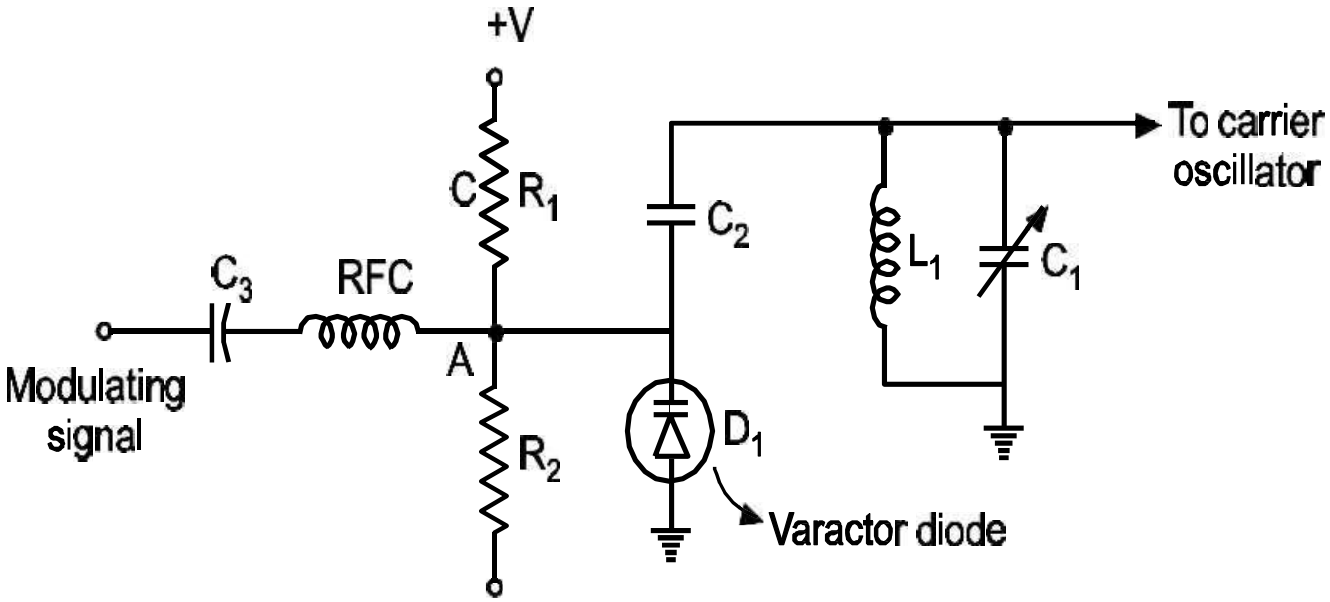


Fig. : Varactor Diode Frequency Modulator

Limitations of Direct Method of FM Generation

1. In this method, it is very difficult to get high order stability in carrier frequency because in this method the basic oscillator is not a stable oscillator, as it is controlled by the modulating signal.

2.Generally in this method we get distorted FM, due to non-linearity of the varactor diode.

Advantages/ Disadvantages/Applications of FM

Advantages of FM

1. Transmitted power remains constant.
2. FM receivers are immune to noise.
3. Good capture effect.
4. No mixing of signals.

Disadvantages of FM

The greatest disadvantages of FM are:

1. It uses too much spectrum space.
2. The bandwidth is wider.
3. The modulation index can be kept low to minimize the bandwidth used.
4. But reduction in M.I. reduces the noise immunity.
5. Used only at very high frequencies.

Applications of FM

1. FM radio broadcasting.
2. Sound transmission in TV.
3. Police wireless.

Indirect Method of FM Generation

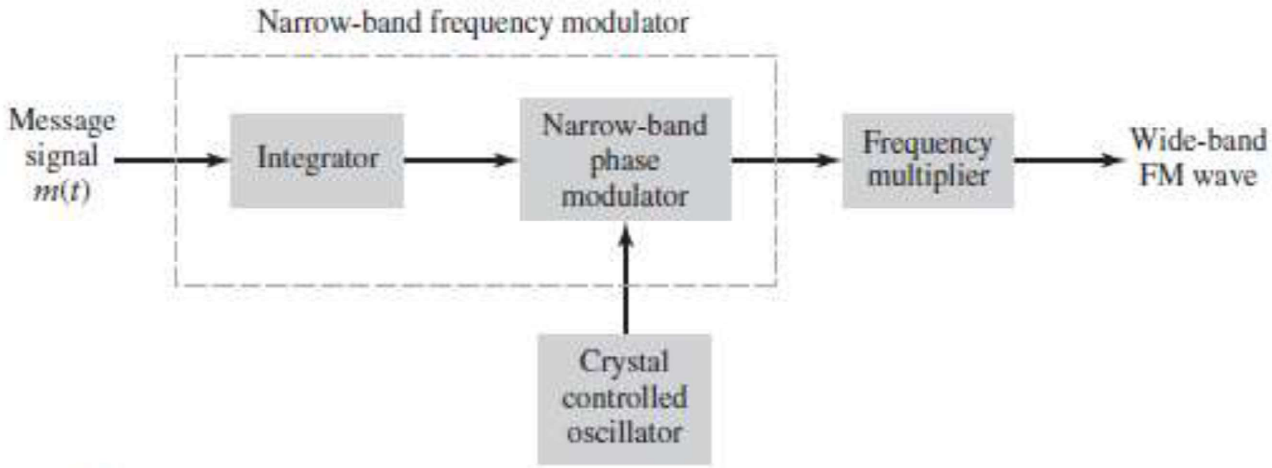


FIGURE 4.10 Block diagram of the indirect method of generating a wide-band FM wave.

Pre-emphasis and De-emphasis

- Pre and de-emphasis circuits are used only in frequency modulation.
  - Pre-emphasis is used **at transmitter** and de-emphasis **at receiver**.

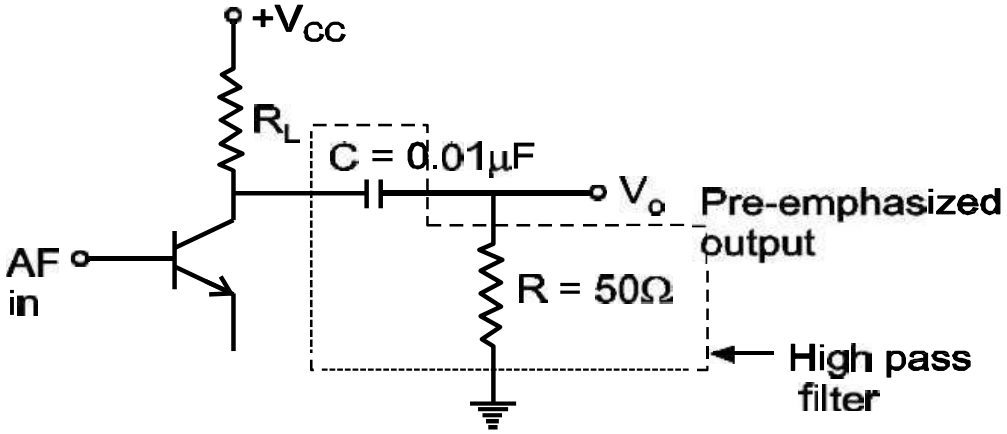
1. Pre-emphasis

- In FM, the noise has a greater effect on the higher modulating frequencies.
- This effect can be reduced by increasing the value of modulation index (m_f), for higher modulating frequencies.
- This can be done by increasing the deviation 'δ' and 'δ' can be increased by increasing the amplitude of modulating signal at higher frequencies.

Definition:

The artificial boosting of higher audio modulating frequencies in accordance with prearranged response curve is called pre-emphasis.

- Pre-emphasis circuit is a high pass filter as shown in Fig.



As shown in Fig. 1, AF is passed through a high-pass filter, before applying to FM modulator.

- As modulating frequency (f_m) increases, capacitive reactance decreases and modulating voltage goes on increasing.

$f_m \propto$ Voltage of modulating signal applied to FM modulat

Boosting is done according to pre-arranged curve as shown in Fig. 2.

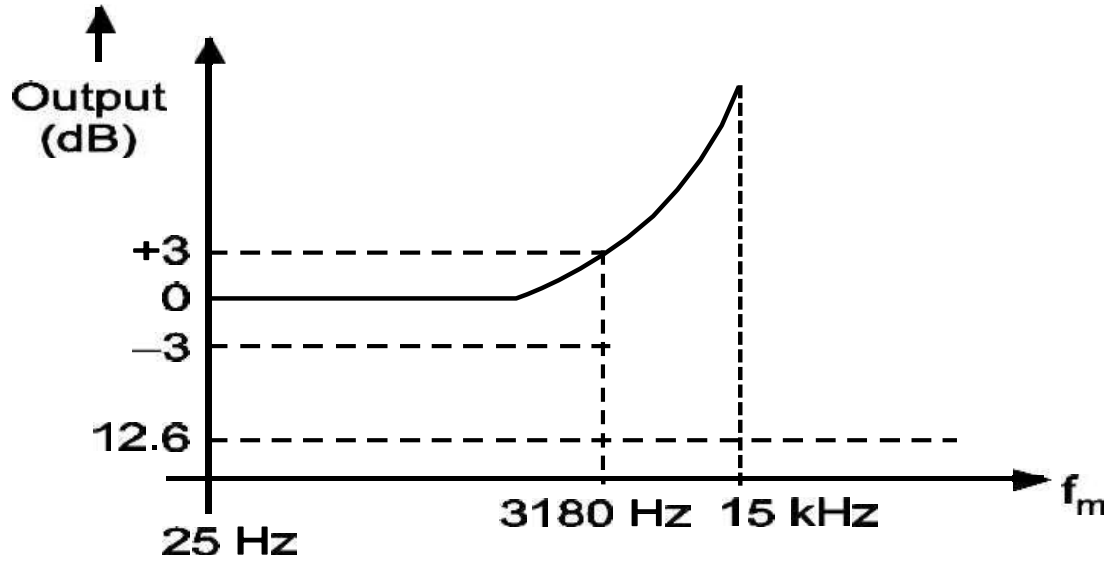


Fig. 2: P re-emphasis Curve

.
The time constant of pre-emphasis is at 50 μs in all CCIR standards.

- In systems employing American FM and TV standards, networks having time constant of 75 μsec are used.

- **The pre-emphasis is used at FM transmitter** as shown in Fig.

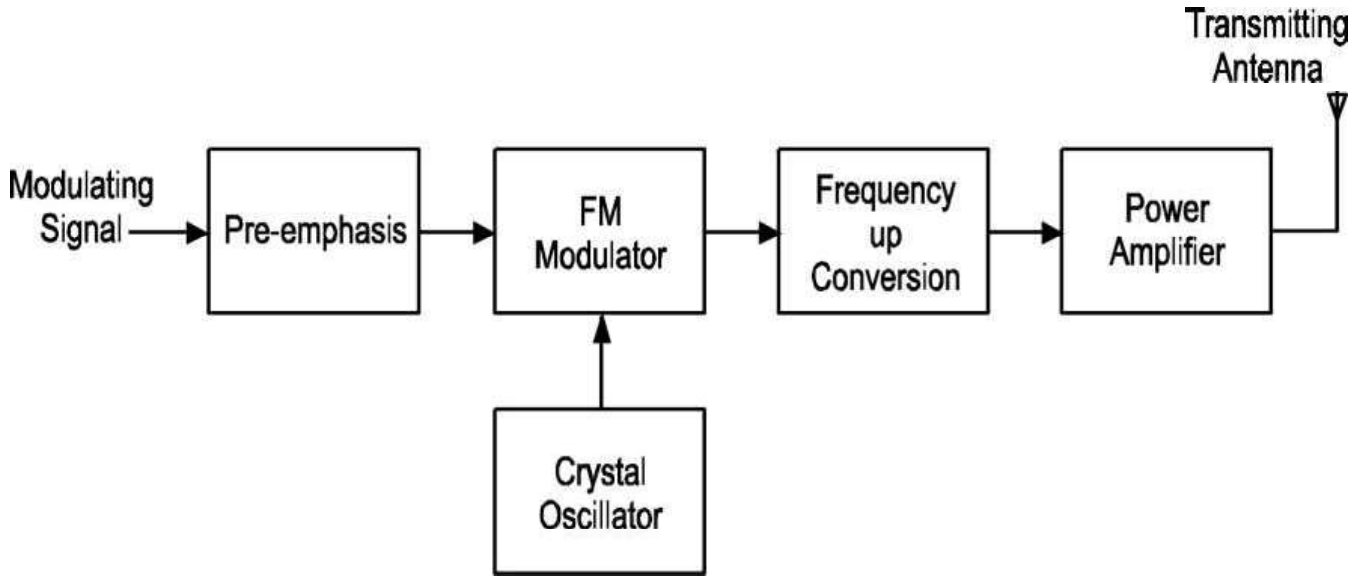


Fig. FM Transmitter with Pre-emphasis

De-emphasis

- De-emphasis circuit is **used at FM receiver**.

Definition:

The artificial boosting of higher modulating frequencies in the process of pre-emphasis is nullified at receiver by process called de-emphasis.

- De-emphasis circuit is a low pass filter shown in Fig.

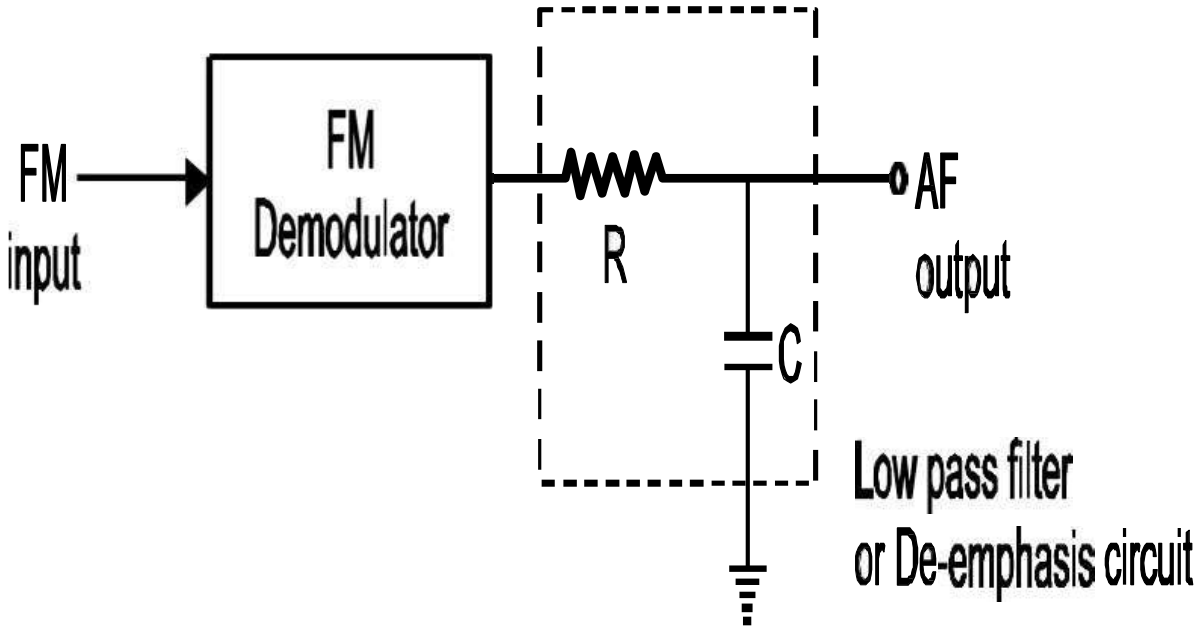
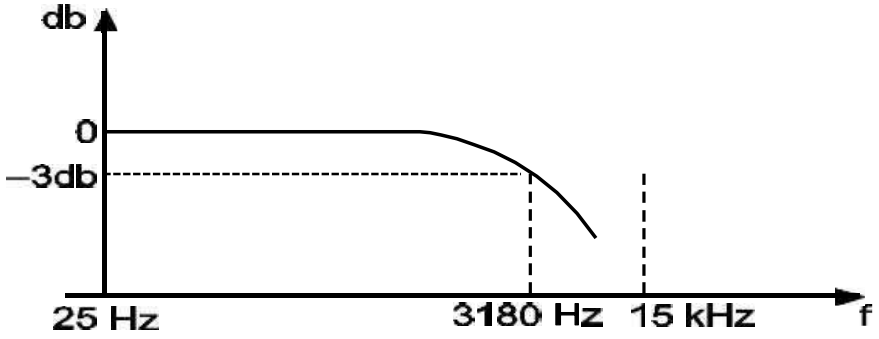


Fig. De-emphasis Circuit

Fig. De-emphasis Curve



As shown in Fig.5, de-modulated FM is applied to the de-emphasis circuit (low pass filter) where with increase in f_m, capacitive reactance X_c decreases. So that output of de-emphasis circuit also reduces •

Fig. 5 shows the de-emphasis curve corresponding to a time constant

50 μs. A 50 μs de-emphasis corresponds to a frequency response curve that is 3 dB down at frequency given by,

$$f = 1/ 2\pi RC$$
$$= 1/ 2\pi \times 50 \times 1000$$
$$= 3180 \text{ Hz}$$

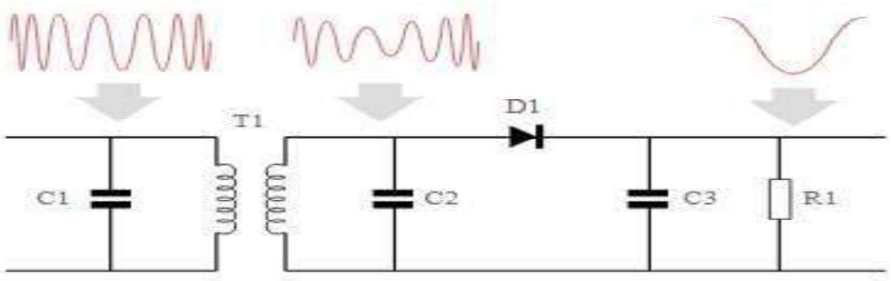
Demodulation of FM Signal

Two steps involved in FM demodulation

•Conversion of FM signal into AM signal, Tank or parallel resonance circuit converts FM into AM signal.

•An envelope detector is used to extract modulating signal from modulated signal.

Slope Demodulator:



FM slope detector circuit showing the signal waveforms

Demodulation of FM Signal

•The input signal is a frequency modulated signal. It is applied to the tuned transformer (T1, C1, C2 combination) which converts the incoming FM signal into AM.

•This AM signal is applied to a simple diode detector circuit, D1. Here the diode provides the rectification, while C3 removes any unwanted high frequency components, and R1 provides a load.

Advantages:

Simple and low cost

Enables FM to be detected without any additional circuitry.

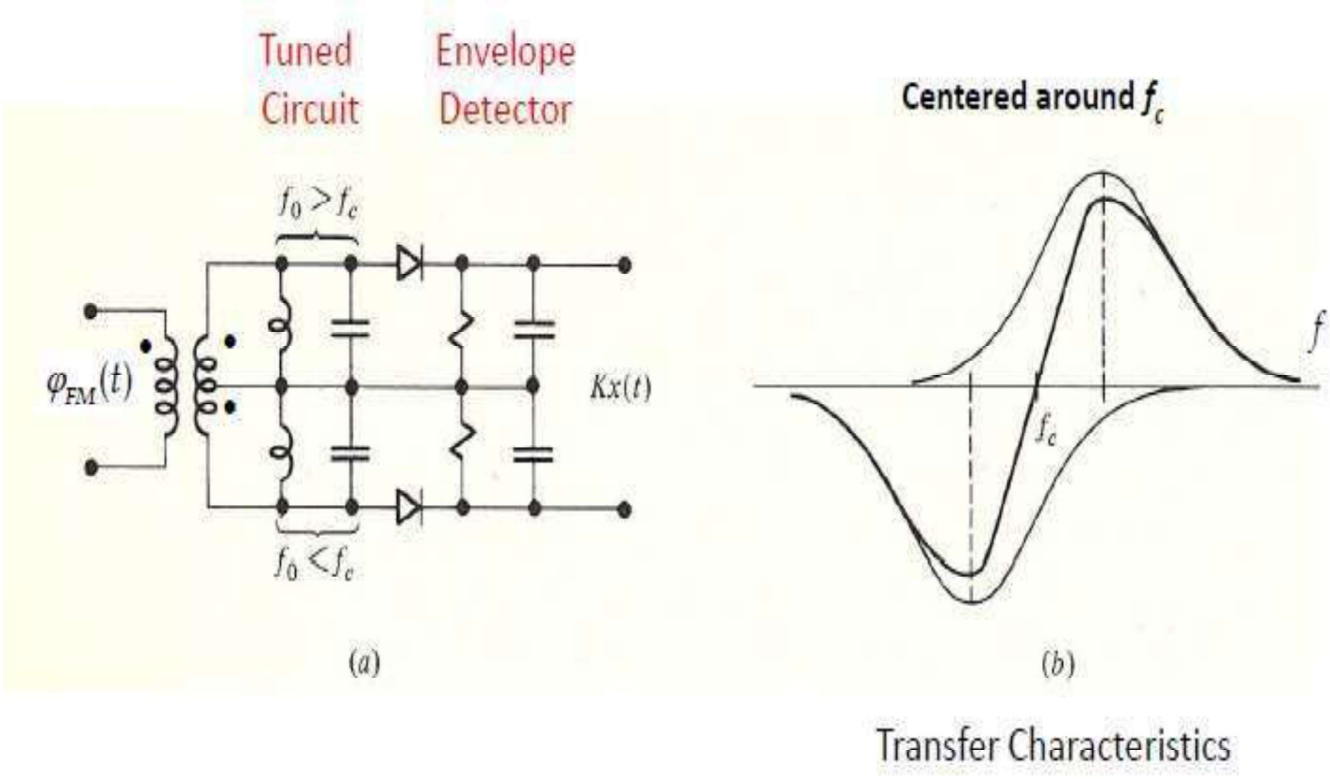
Disadvantages:

Nonlinear operation

Both frequency and amplitude variations are demodulated and this means that much higher levels of noise and interference are experienced.

Demodulation of FM Signal

Foster Seeley Demodulator or detector:



Demodulation of FM Signal

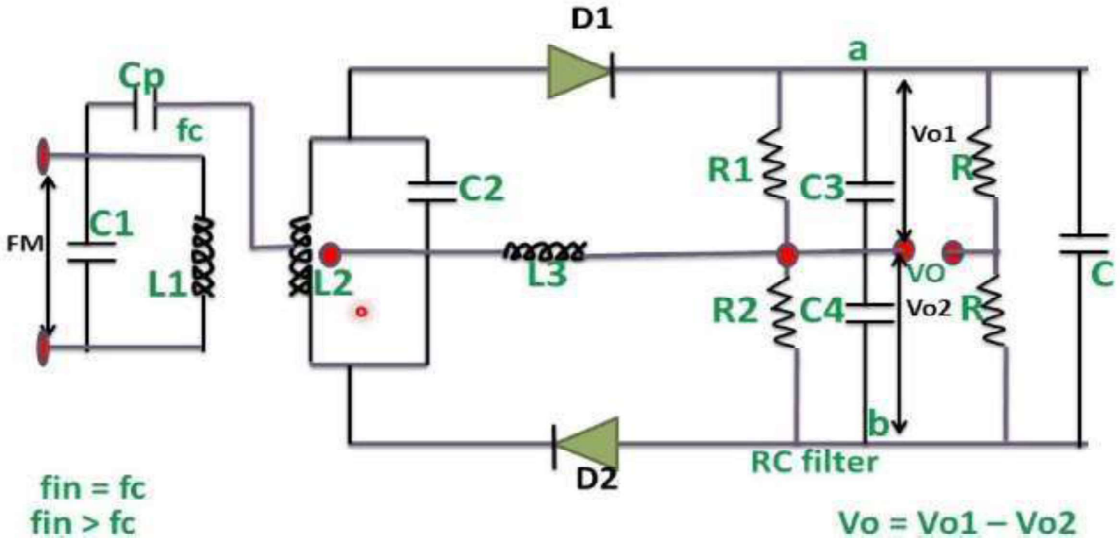
•Foster seeley demodulator contains two tuning circuits and two envelope detectors.

•One section of tuning circuit and envelope detector works for incoming frequency is greater than carrier frequency and vice versa for incoming frequency less than carrier frequency.

•Tuning circuit converts FM signal to AM signal and Envelope detector extracts message signal from AM signal.

Demodulation of FM Signal

Ratio Demodulator:



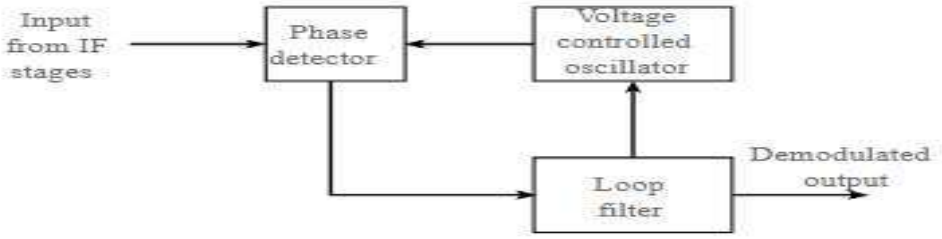
•Ratio detector is similar to Foster seeley demodulator except of Diode of D2 is reversed potential divider circuit.

•Potential divider circuit suppress the noise and this advantage of ratio detector.

Demodulation of FM Signal

PLL Demodulator or detector:

•Phase Locked Loop is closed loop system which contains Phase detector, VCO and loop filter or LPF as shown in figure.



PLL Phase locked Loop FM demodulator

•It continuously finds the phase difference between incoming FM signal and locally generated carrier.

•And based on Phase difference it generates Modulating signal.

UNIT IV

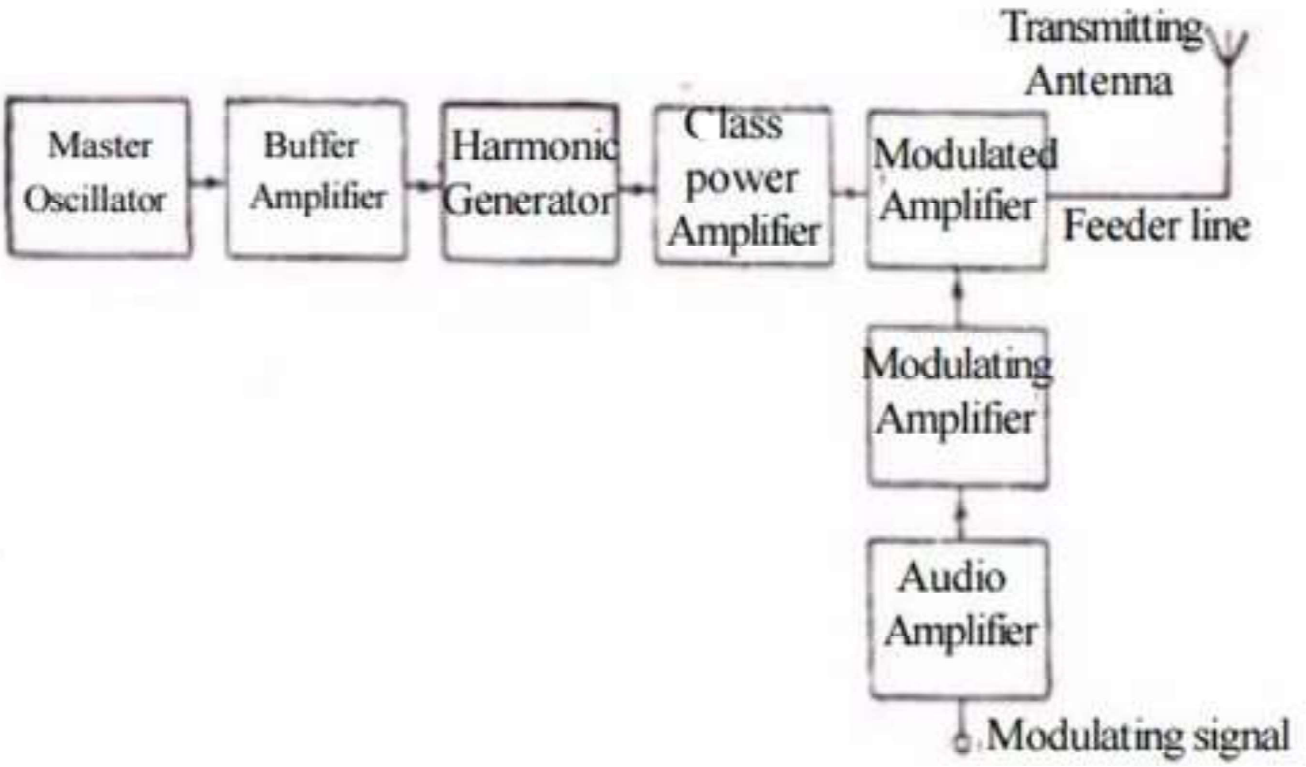
TRANSMITTERS:

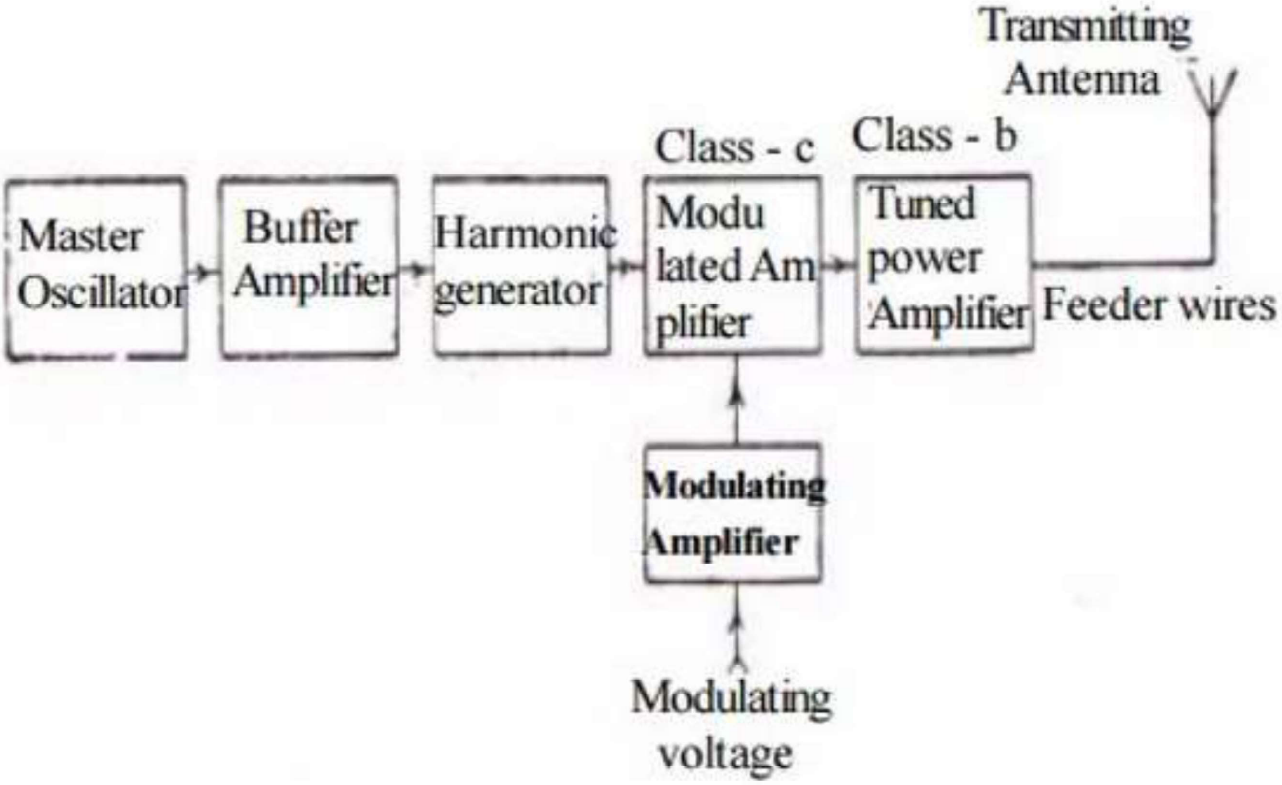
High Level and low level modulation

In high power level modulation system, the carrier voltage is modulated at the highest power level. The required power level is obtained by using class c amplifiers.

In low power level modulation system, the carrier is modulated at low power level and the carrier power is subsequently raised to the desired level in class B Tuned power amplifiers. Transmitters using high power level are widely used at present.

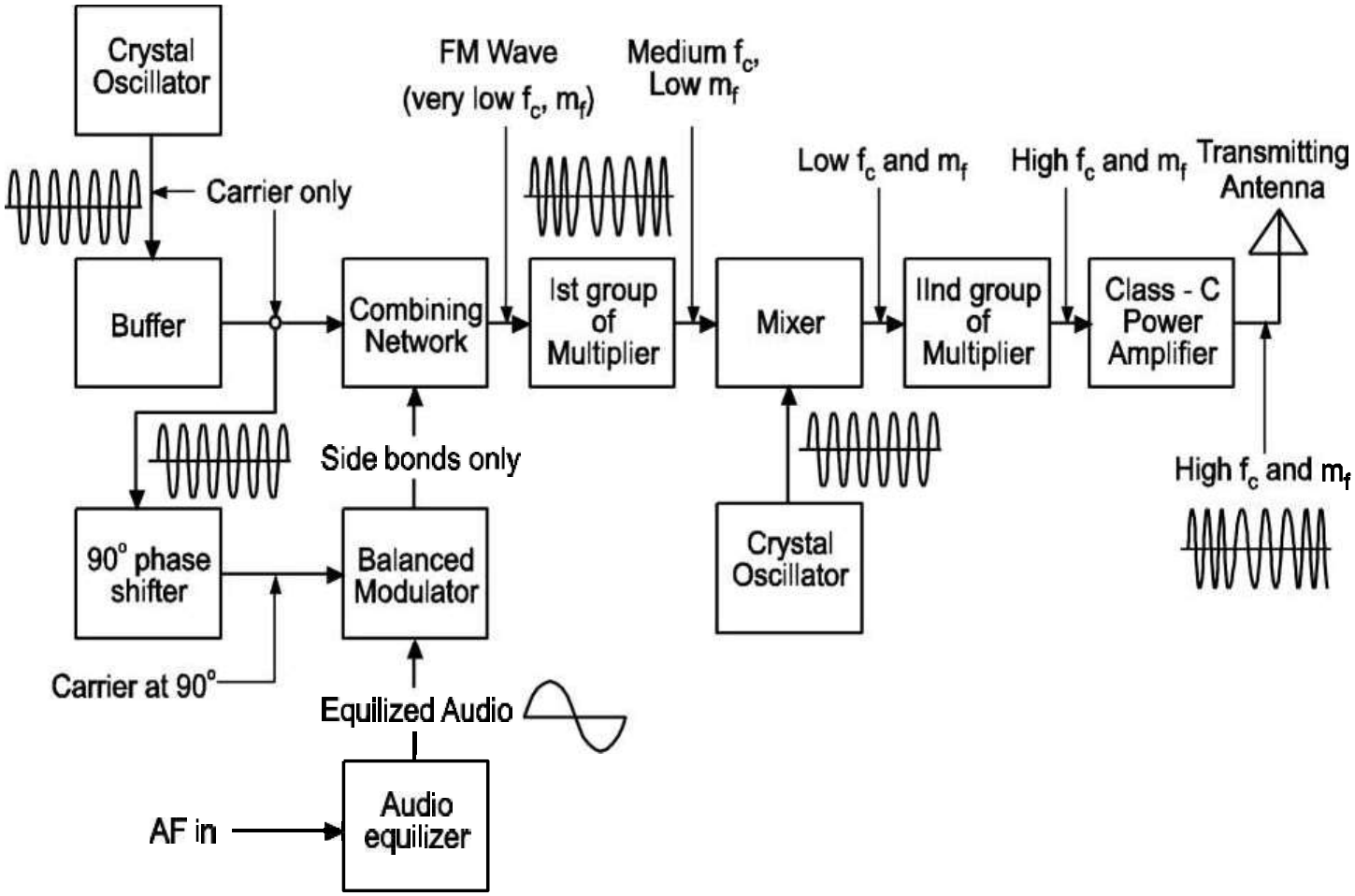
Block diagram of high level AM transmitter





Block diagram of AM low level Transmitter

FM Transmitter (Armstrong Method)



Tuned Radio Frequency Receiver:

The Tuned Radio Frequency Receiver is a simple “logical” receiver. Two or perhaps three RF amplifiers, all tuning together, were employed to select and amplify the incoming frequency and simultaneously to reject all others.

After the signal was amplified to a suitable level, it was demodulated (detected) and fed to the loudspeaker after being passed through the appropriate audio amplifying stages.

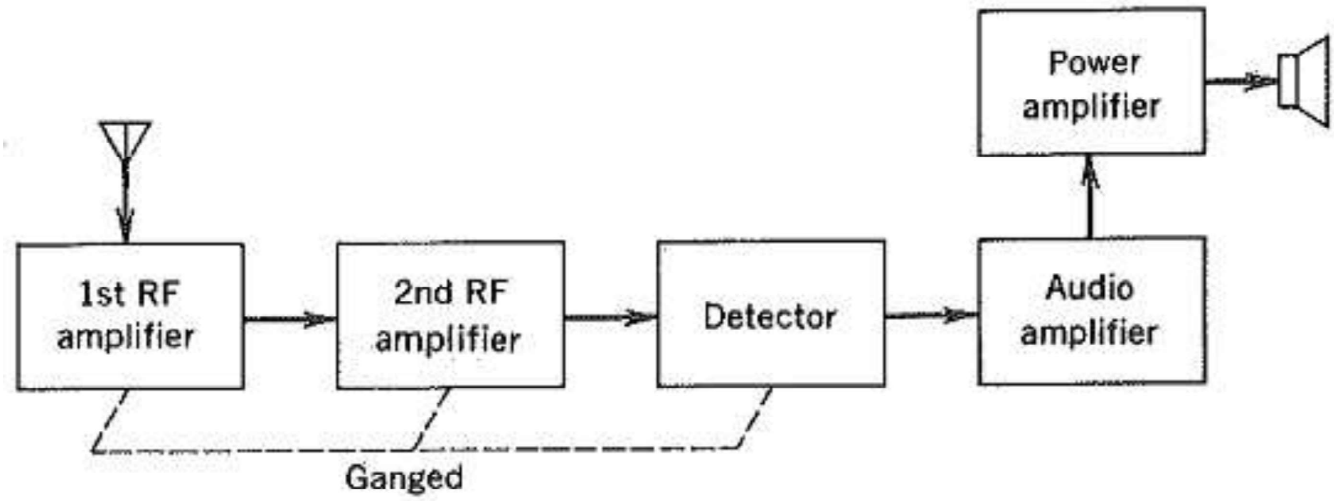


FIGURE 6-1 The TRF receiver.

Advantages:

simple to design and align at broadcast frequencies (535 to 1640 kHz)

Disavantages:

- They present difficulties at higher frequencies.
- This was mainly because of the instability associated with high gain.
- Variation in bandwidth over the tuning range.
- Unable to achieve sufficient selectivity at high frequencies

Consider a tuned circuit required to have a bandwidth of 10 kHz at a frequency of 535 kHz. The Q of this circuit must be $Q = f/\Delta f = 535/10 = 53.5$.

At the other end of the broadcast band, i.e., at 1640 kHz, Q of the coil is increased to 164.

Losses dependent on frequency will prevent so large an increase. Thus the Q at 1640 kHz is unlikely to be in excess of 120, giving a bandwidth of $\Delta f = 1640/120 = 13.7$ kHz and ensuring that the receiver will pick up adjacent stations as well as the one to which it is tuned.

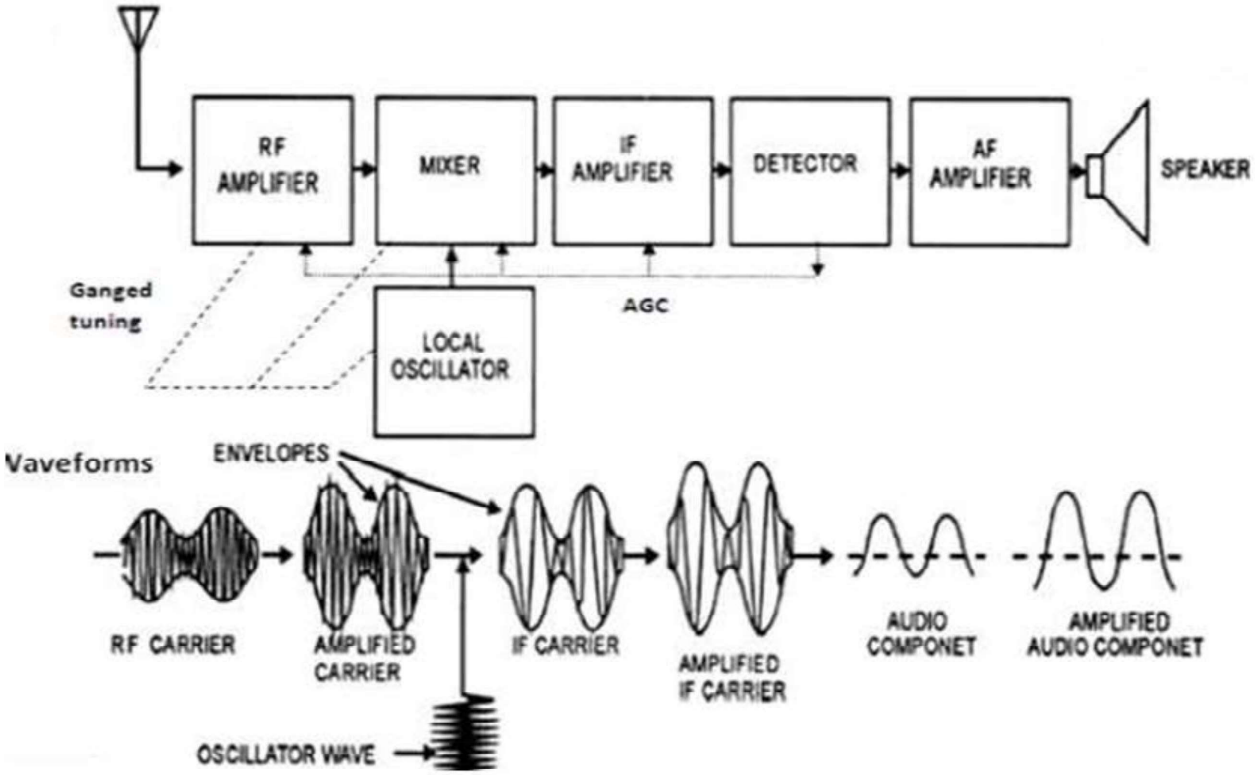
Super Heterodyne AM Receiver

Heterodyne means mixing two frequencies and generating single or constant frequency and the output of mixer will be fixed frequency.

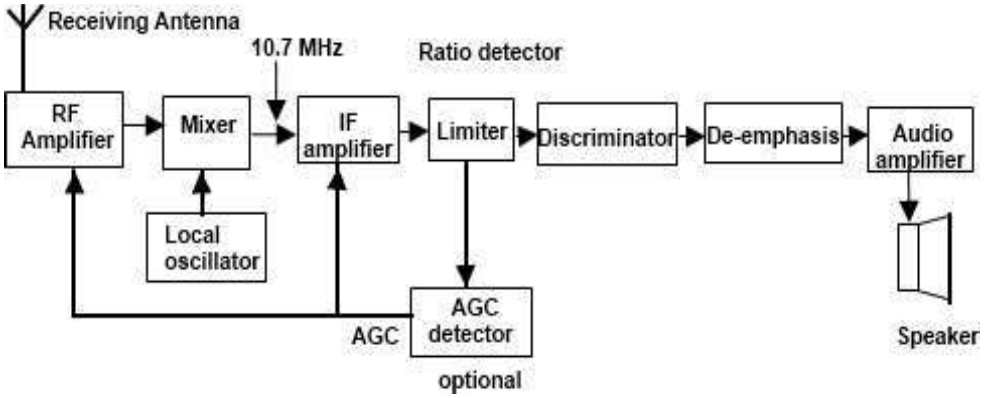
Specification of AM Receiver:

- The frequency range of AM-MW(Medium wave) : (540-1640) KHz
- Band width of receiver:1640 KHz – 540 KHz = 1100 KHz
- Band width of each AM station : 10 KHz
- No. of stations available: 110
- Intermediate frequency ($\mathbf{f_{IF}}$): 455 KHz

Super Heterodyne AM Receiver



Super Heterodyne FM Receiver



Antenna: It is passive device which converts electromagnetic signal into electrical signal.

RF Tuned Amplifier:

It is broad band amplifier which contain tuning circuit and amplifier.

Tuning circuit designed to select 100 stations and amplifier provides amplification for 20MHz or20 000 KHzband width.

RF tuned amplifier is responsible for sensitivity, selectivity, Image signal rejection and noise reduction.

Super Heterodyne FM Receiver

Mixer: It is combination of frequency mixer and Band Pass Filter (BPF).

Frequency generates sum and difference frequency of incoming signal and locally generated signal.

BPF selects difference frequency at the output whose center frequency is equal to = 10.7MHz.

Local Oscillator:

It is either Colpits or Hartley oscillator.

It generates carrier frequency 10.7MHz.greater than the incoming carrier frequency to produce constant or fixed frequency.

IF Amplifier:

It is narrow band, high gain and fixed frequency amplifier which provides amplification for 20 MHz band width at center frequency of 10.7 MHz.

Super Heterodyne FM Receiver

Limiter:

It is combination of hard limiter and BPF.

Hard limiter is two sided independent clipper removes the noise spikes.

Detector or Demodulator or Discriminator:

It is frequency translator circuit which extracts modulating signal from FM signal.

De-emphasis:

It is LPF which attenuates frequencies of Audio signal from 2 KHz to 20 KHz to get the original modulating signal.

Audio Amplifier:

It is low frequency amplifier which provides amplification at (20-20K) Hz.

Loud Speaker:

It converts Electrical signal into sound or audio signal.

Radio receiver measurements

The important characteristics of superheterodyne radio receiver are,

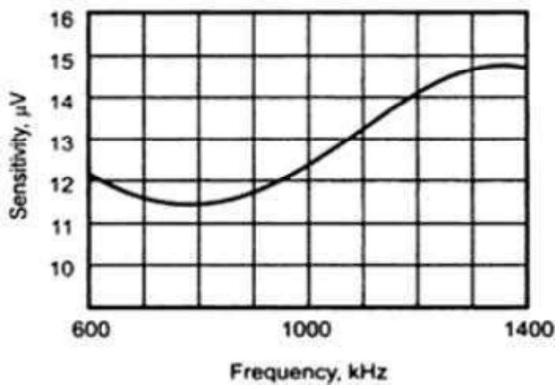
- Sensitivity
- Selectivity
- Fidelity

Sensitivity:

- It is defined as the ability of receiver to amplify weak signals
- It is defined in terms of voltage which must be applied at the receiver input terminals to provide a standard output power at the receiver output.

Radio receiver measurements

- Sensitivity is expressed in milli volts
- For practical receivers sensitivity is expressed in terms of signal power required to produce minimum acceptable output with minimum acceptable noise.

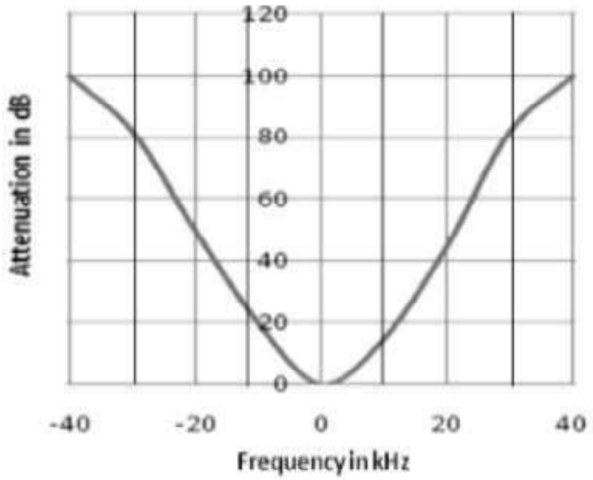


- Sensitivity of superheterodyne radio receiver depends on
- Gain of RF amplifier
- Gain of IF amplifier
- Noise figure of RX

Radio receiver measurements

Selectivity:

It is defined as the ability of receiver to reject unwanted signals.



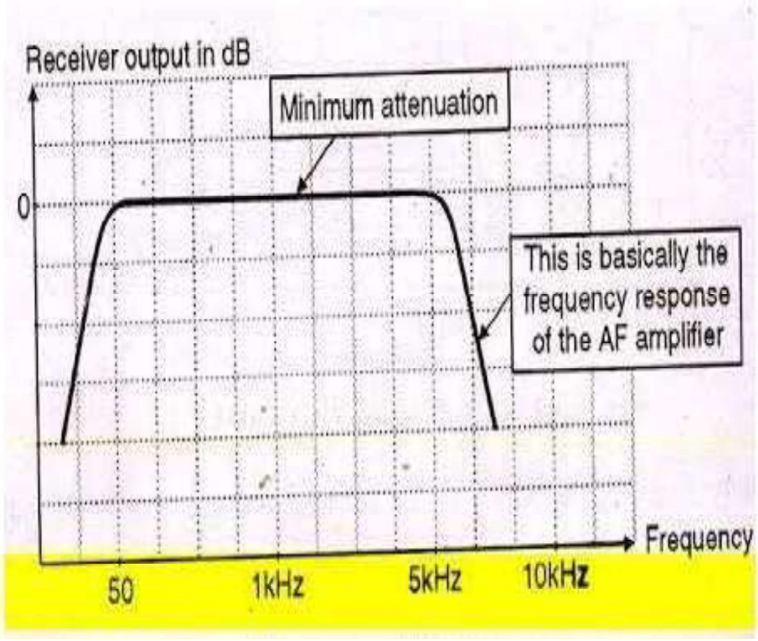
Selectivity depends on

- Receiving frequency
- Response of IF section

Radio receiver measurements

Fidelity:

It is the ability of a receiver to reproduce all the modulating frequencies equally.



UNIT V

SAMPLING

It is the process of converting a continuous time signal into a discrete time signal

During sampling, sufficient number of samples of the signal must be taken so that original signal is correctly represented in its samples and possible for reconstruction.

Number of samples to be taken depends on maximum signal frequency present in the signal.

- Different types of samples are,
- Ideal
- Natural
- Flat top

SAMPLING

Sampling theorem:

A continuous time signal may be completely represented in its samples and recovered back if the sampling frequency fs>2fm

Nyqyist rate and Nyquist interval:

When sampling rate becomes exactly equal to 2fm samples per second, it is called Nyquist rate

$$fs=2fm \text{ Hz}$$

Maximum sampling interval is called Nyquist interval.

$$Ts = 1/fs=1/2fm \text{ sec}$$

Ideal sampling

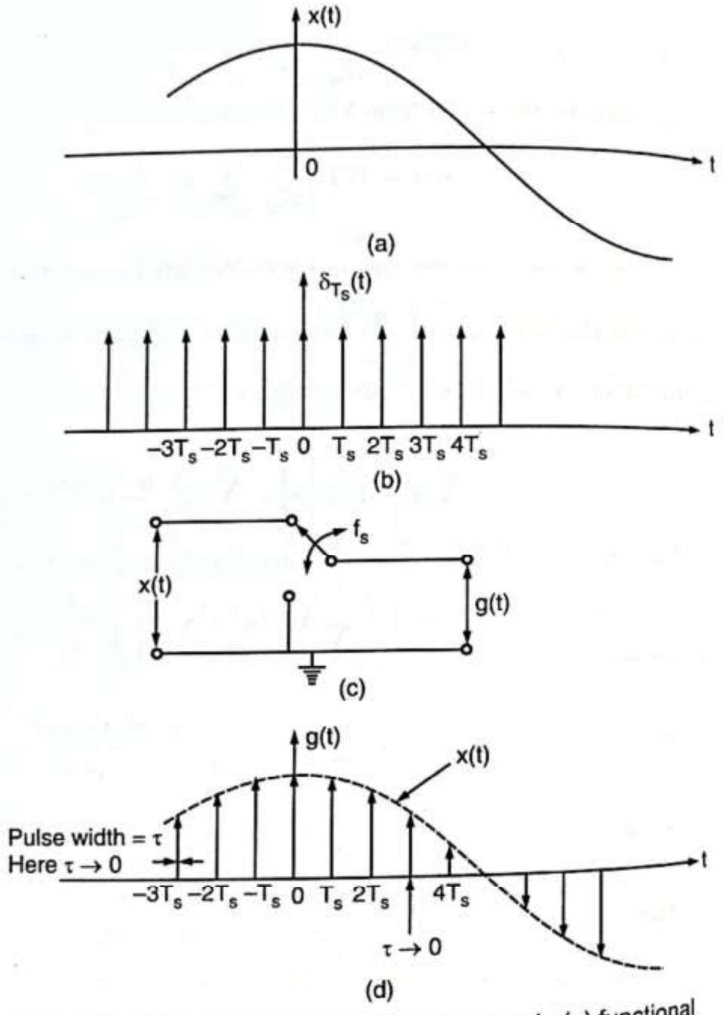


Fig. 5.11. (a) Baseband signal (b) impulse train (c) functional diagram of a switching sampler (d) sampled signal

NATURAL SAMPLING

In natural sampling, pulse has a finite width equal to τ.

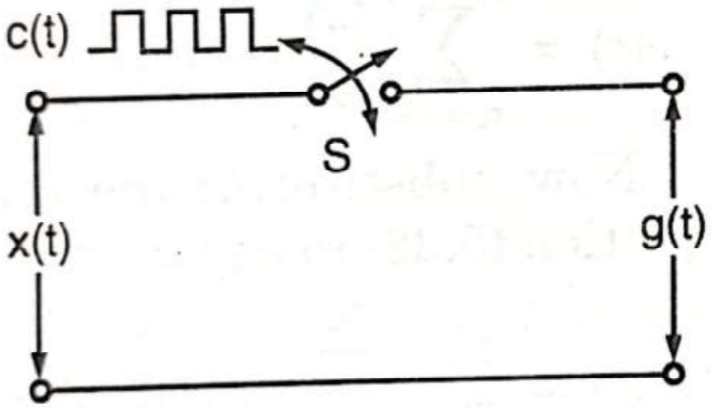


Fig. 5.12. A functional diagram of a Natural Sampler

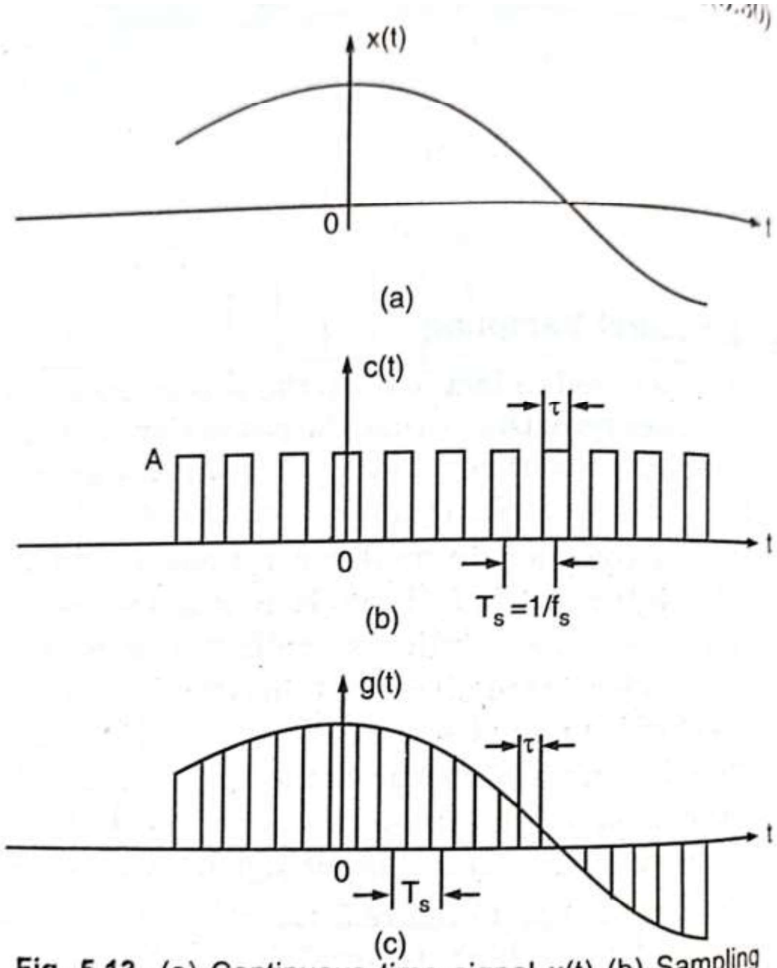


Fig. 5.13. (a) Continuous time signal x(t) (b) Sampling function waveform i.e. periodic pulse train (c) Naturally sampled signal waveform g(t)

NATURAL SAMPLING

Let an analog continuous time signal x(t) sampled at a rate fs Hz and sampling function c(t) which is a train of periodic pulse of width τ and frequency fs Hz

Case i: When c(t) is high

Switch S is closed and output g(t) is exactly equal to input

$$g(t) = x(t)$$

NATURAL SAMPLING

Case ii: When c(t) is low

Switch s is open

$$g(t) = 0$$

The time domain representation of naturally sampled signal is given by,

$$g(t) = x(t)\sum_{n=-\alpha}^{\alpha} \frac{\tau A}{Ts} Sinc(fn.\tau)e^{j2\pi fsnt}]$$

The spectrum of naturally sampled signal is given by,

$$G(f) = \frac{\tau A}{Ts}\sum_{n=-\alpha}^{\alpha} Sinc(nfs\tau)X(f - nfs)$$

Flat Top Sampling

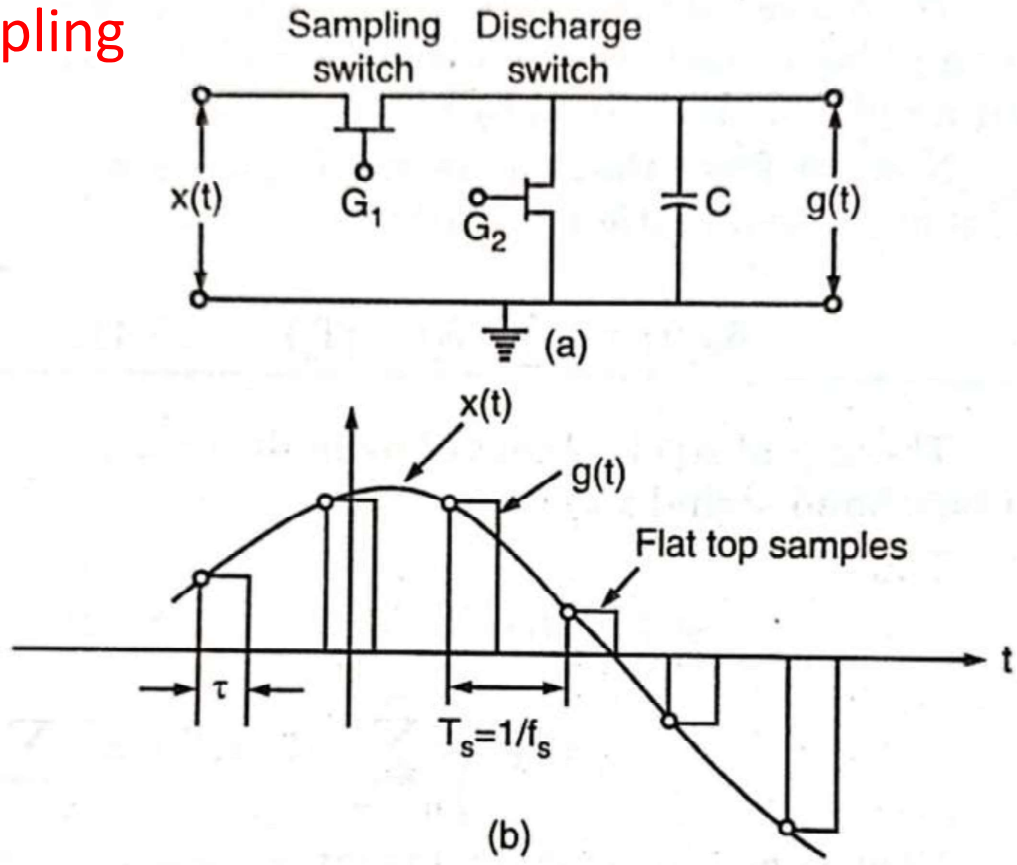


Fig. 5.15. (a) A sample and hold circuit to generate flat top samples (b) A general waveform of flat top sampling

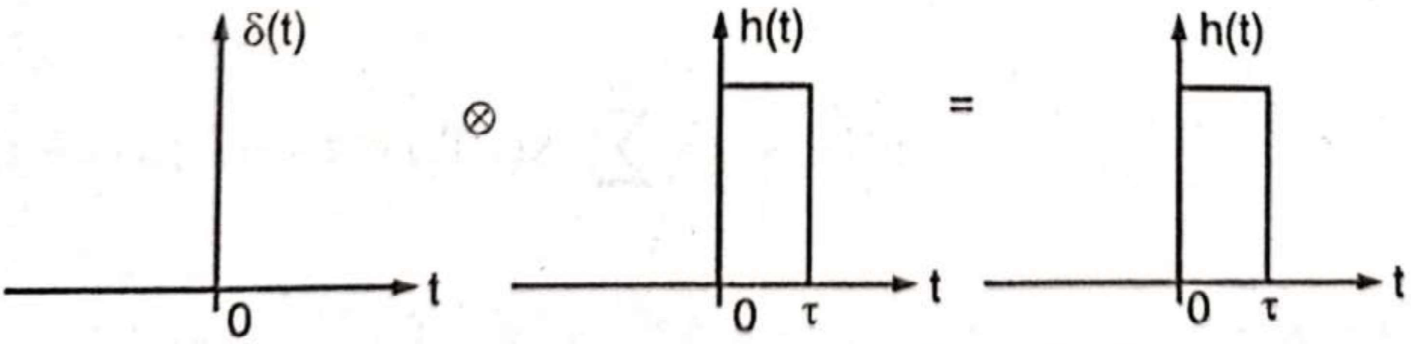


Fig. 5.16. Convolution of any function with delta function is equal to that function

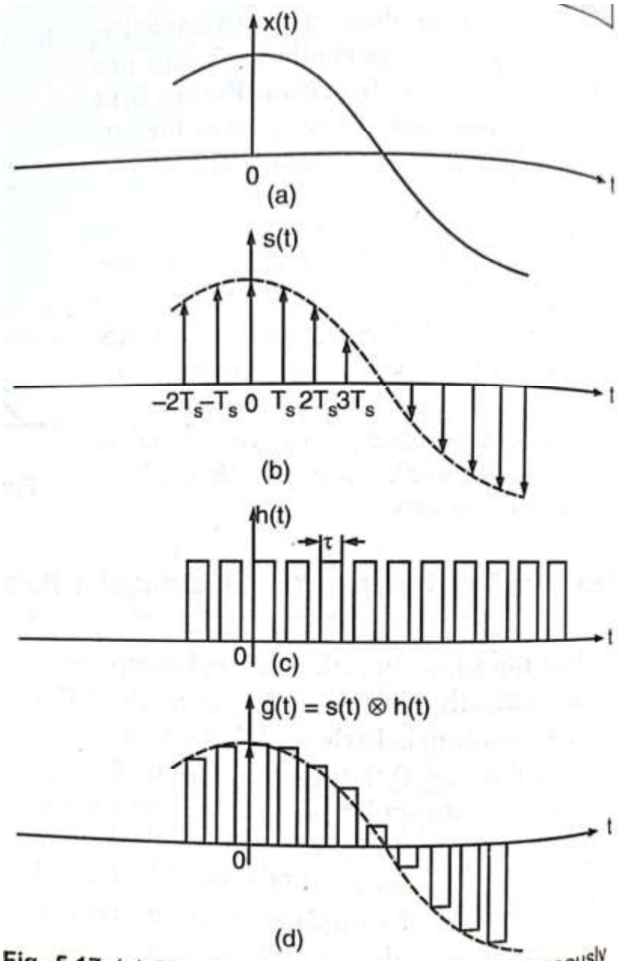


Fig. 5.17 (a) Baseband signal x(t) (b) Instantaneously sample signal s(t) (c) Constant pulse width function h(t) (d) Flat top sampled signal g(t) obtained through convolution of h(t) and s(t)

ANALOG PULSE MODULATION SCHEMES

Pulse Modulation:

The process of transmitting the signals in the form of pulses by using some special techniques.

There are two types of pulse modulation systems,

1. Pulse Amplitude Modulation
2. Pulse Time Modulation

Pulse time modulation is further divided into,

- Pulse Width Modulation
- Pulse Position Modulation

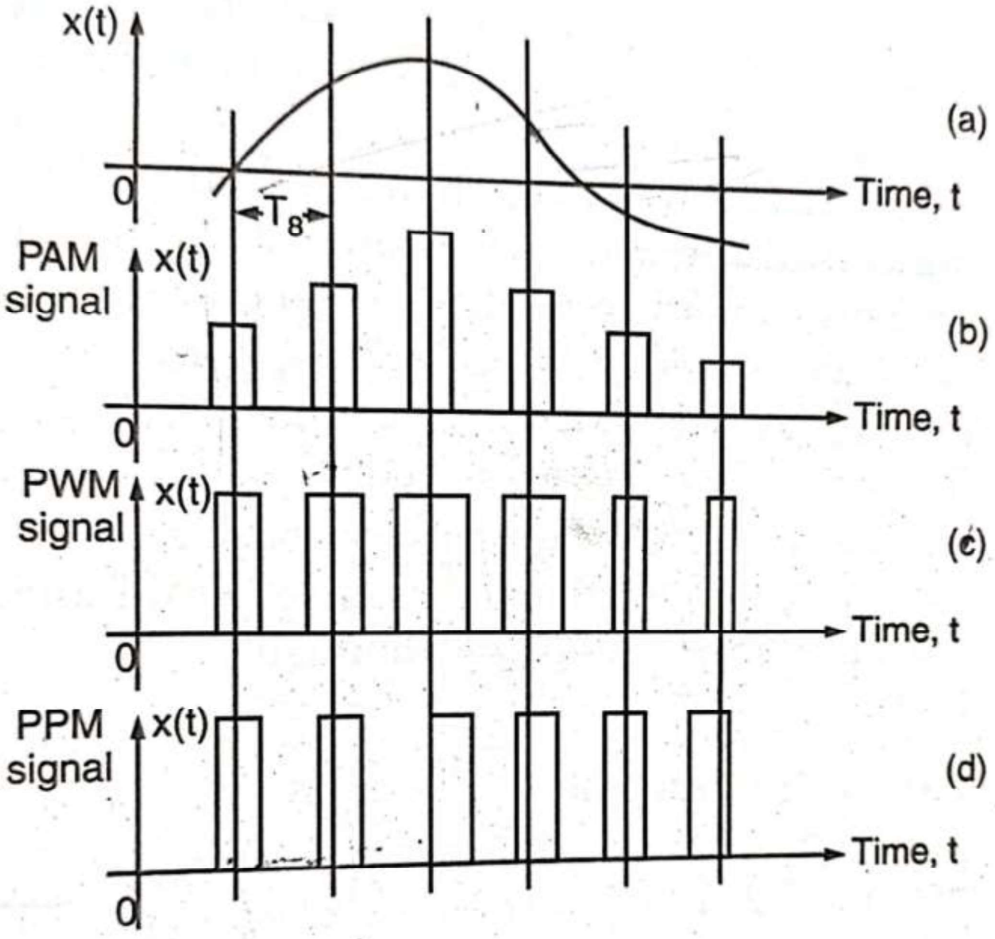


Fig. 5.21. Different types of pulse analog modulation methods

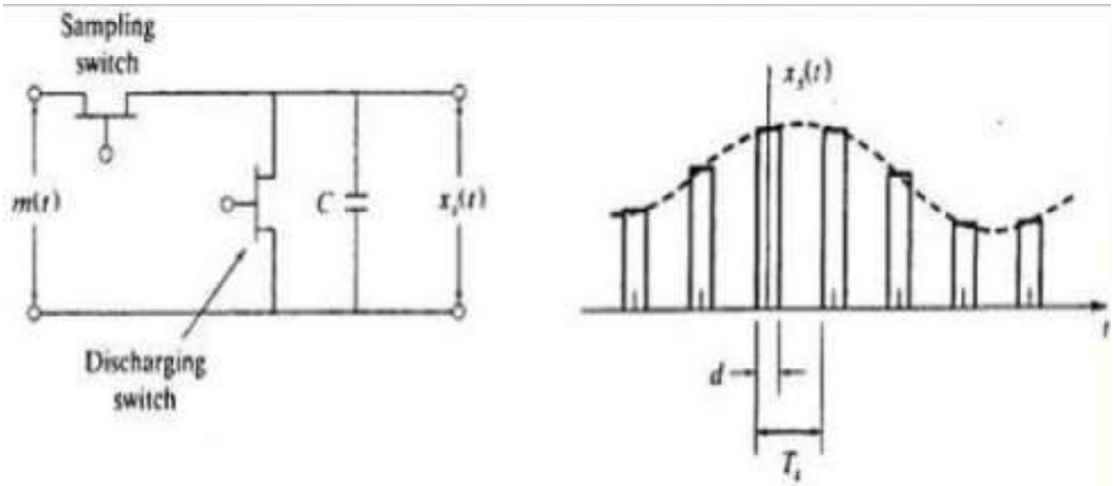
PULSE AMPLITUDE MODULATION(PAM)

In Pulse amplitude modulation, the amplitude of pulses of carrier pulse train is varied in accordance with the modulating signal.

In PAM , the pulses can be flat top type or natural type or ideal type.

Out of these, flat top PAM is widely used because of easy noise removal.

PAM GENERATION

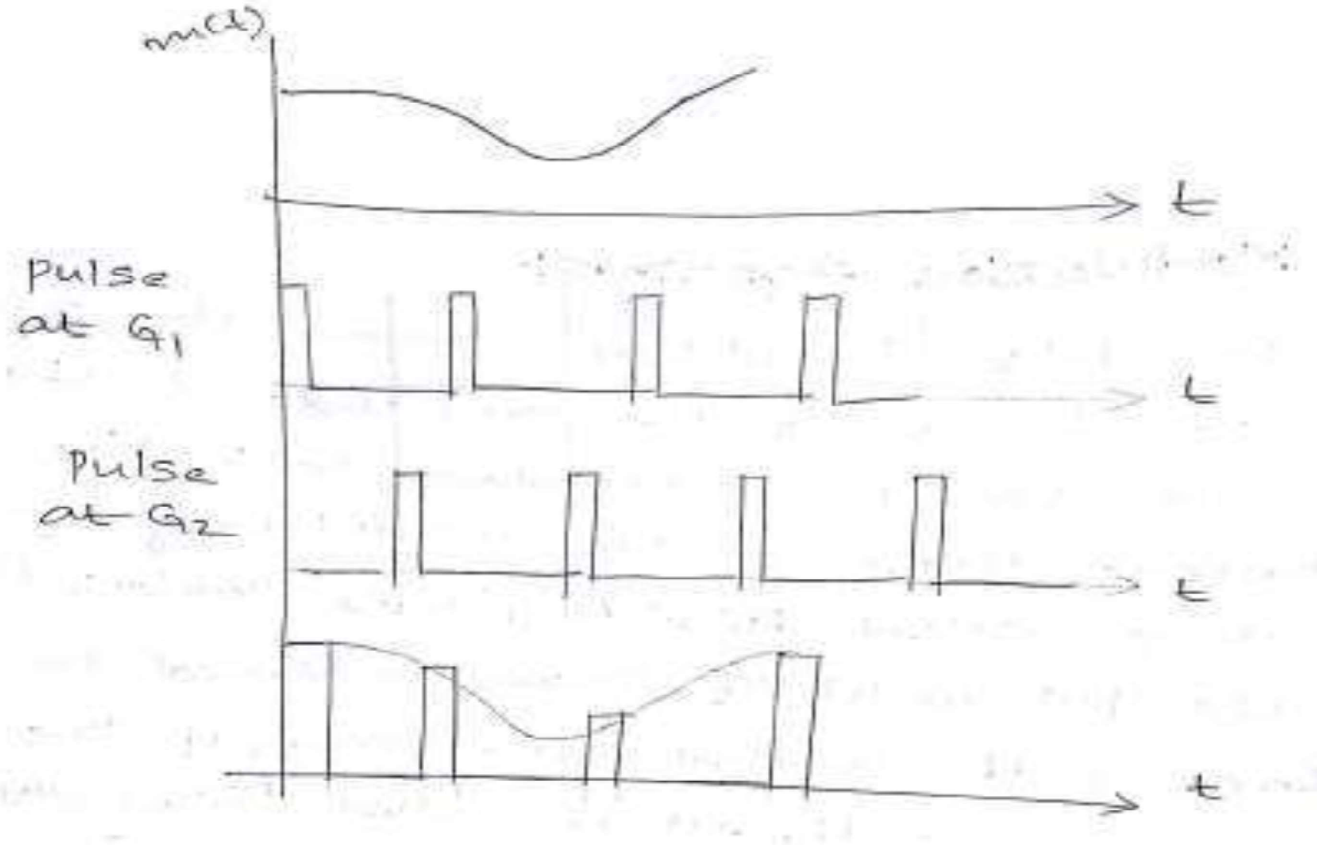


- The sample and hold circuit consists of two FETs and a capacitor.
- The sampling switch is closed for a short duration by a short pulse applied to the gate G1 of transistor.

PAM GENERATION

- During this period, the capacitor is quickly charged to a voltage equal to instantaneous sample value of incoming signal x(t)
- Now the sampling switch is opened and capacitor holds the charge.
- The discharge switch is then closed by a pulse applied to gate G2 of second transistor.
- Due to this the capacitor is discharged to zero volts. The discharge switch is then opened and the capacitor has no voltage.
- Hence the output of sample and hold circuit consists of a sequence of flat top samples.

PAM GENERATION



Transmission bandwidth of PAM

In PAM signal the pulse duration τ is assumed to be very small compared to time period Ts i.e τ< Ts

If the maximum frequency in the modulating signal x(t) is fm then sampling frequency fs is given by fs<=2fm Or 1/Ts <= 2fm or Ts <= 1/2fm

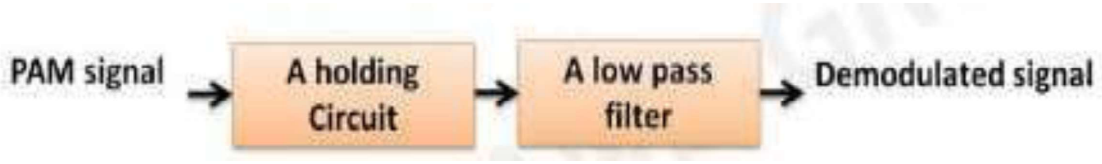
Therefore, τ< Ts <= 1/2fm

If ON and OFF time of PAM pulse is equal then maximum frequency of PAM pulse will be fmax = 1/ τ+ τ = 1/2 τ

Therefore, transmission bandwidth >=1/2 τ >= 1/[2(1/2fm)>= fm

Demodulation of PAM

Demodulation is the reverse process of modulation in which modulating signal is recovered back from the modulated signal.



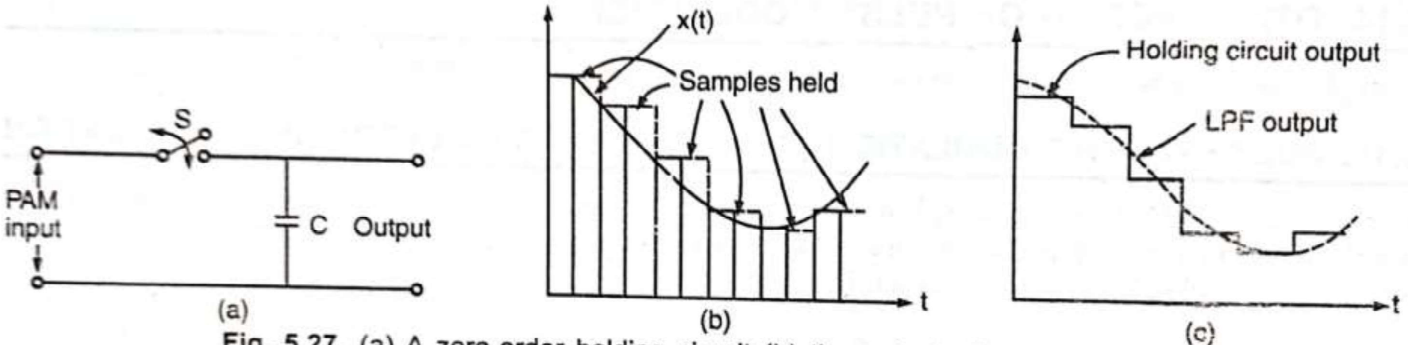


Fig. 5.27. (a) A zero-order holding circuit (b) the output of holding circuit (c) the output of a Low Pass filter (LPF)

Demodulation of PAM

- For PAM signals, demodulation is done using a holding circuit.
- The received PAM signal is first passed through a holding circuit and then through a lowpass filer.
- Switch S is closed during the arrival of the pulse and is opened at the end of the pulse.
- Capacitor C is charged to pulse amplitude value and holds this value during the interval between two pulses.
- Holding circuit output is then passed through a low pass filter to extract the original signal.

Advantages, Disadvantages of PAM

Advantages:

- It is the simple process for modulation and demodulation
- Transmitter and receiver circuits are simple and easy to construct.

Disadvantages:

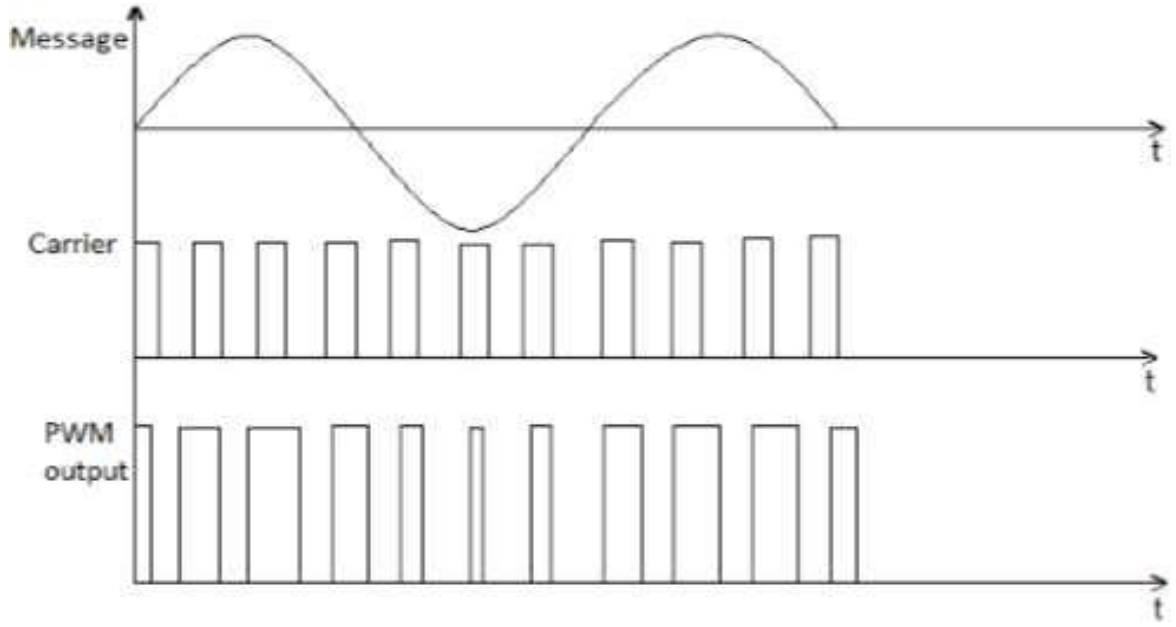
- Bandwidth requirement is high
- Interference of noise is maximum
- Power requirement is high

Applications:

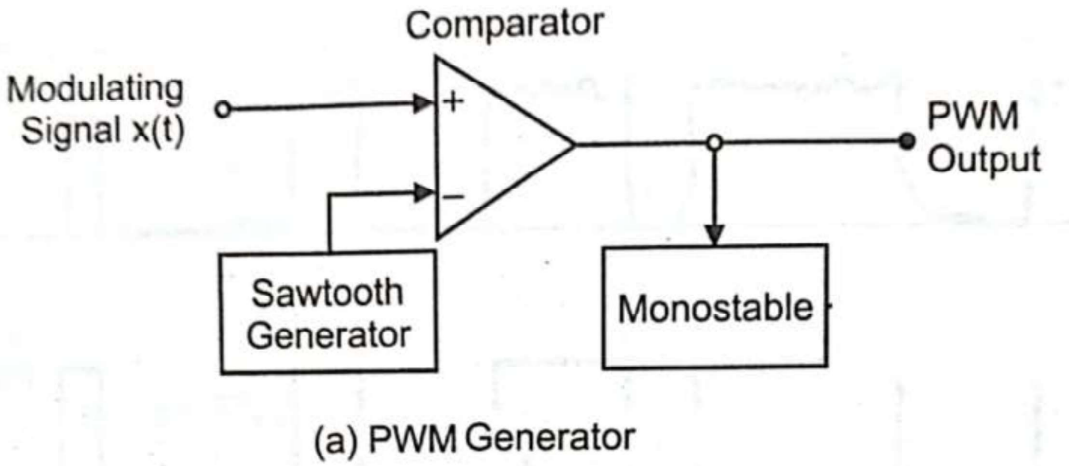
- Used in microcontrollers for generating control signals
- Used as electronic driver for LED lighting

Pulse Width Modulation(PWM)

In PWM, the width of pulses of carrier pulse train is varied in proportion with amplitude of modulating signal.



PWM GENERATION



(a) PWM Generator

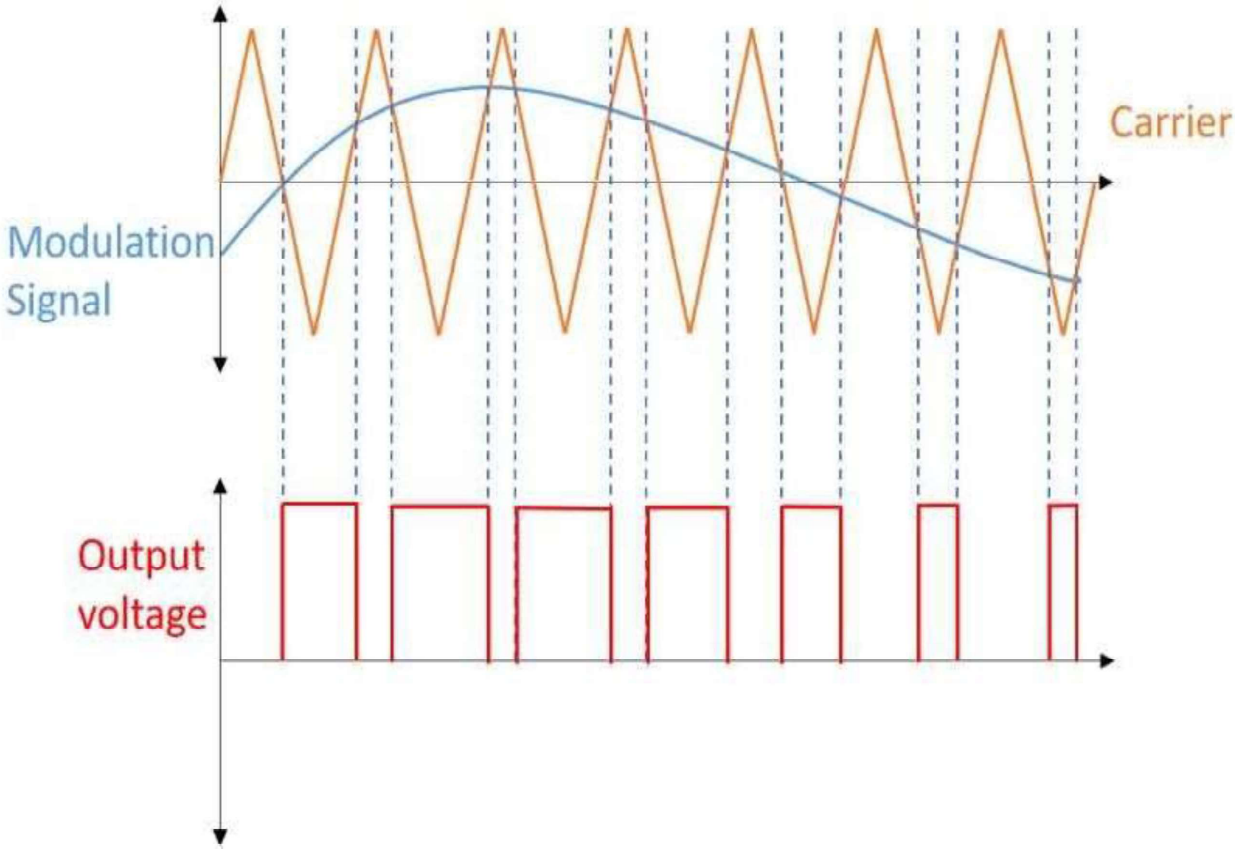
A sawtooth generator generates a sawtooth signal of frequency fs.

This is applied to inverting terminal of comparator.

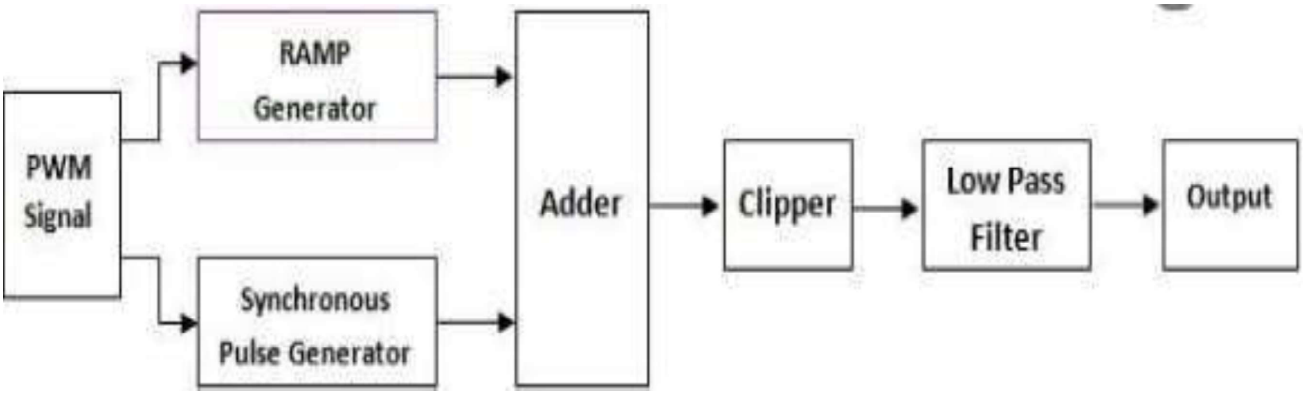
PWM GENERATION

- Modulating signal x(t) is applied to non-inverting terminal of comparator.
- Comparator output remains high as long as instantaneous amplitude of x(t) is higher than sawtooth signal.
- This gives the PWM output at the output of comparator.
- The leading edges of PWM waveform coincide with falling edges of ramp signal
- Therefore, leading edges of PWM signal are always generated at fixed time intervals
- Occurrence of falling edge of PWM signal is dependent on instantaneous amplitude of x(t)

PWM GENERATION



DETECTION OF PWM

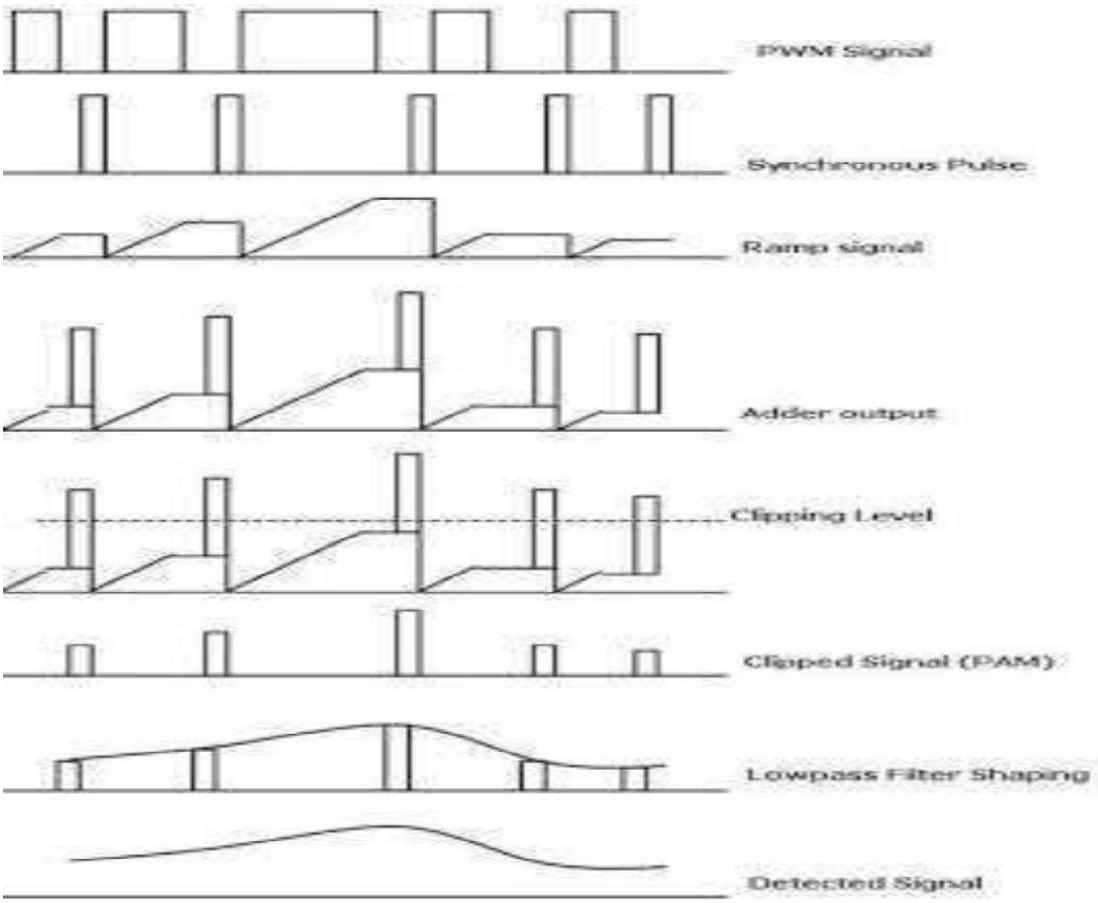


- The PWM signal received at the input of detector circuit will contain noise
- This signal is applied to a pulse generator which regenerates the PMW signal.
- Some of the noise is removed and the pulses are squared up.

DETECTION OF PWM

- The regenerated pulses are applied to a reference pulse generator.
- It produces a train of constant amplitude and constant width pulses.
- These pulses are synchronized to the leading edges of regenerated PWM pulses but delayed by fixed intervals.
- The regenerated PWM pulses are also applied to a ramp generator whose o/p is a constant slope ramp for the duration of the pulse.
- At the end of the pulse a sample and hold circuit retains the final ramp voltage until it is reset at the end of the pulse.
- The constant amplitude pulses at the o/p of the reference generator are then added to ramp signal.
- O/P of the adder is then clipped off at a threshold level to generate a PAM signal.
- A low pass filter is used to recover the original modulating signal back from PAM signal.

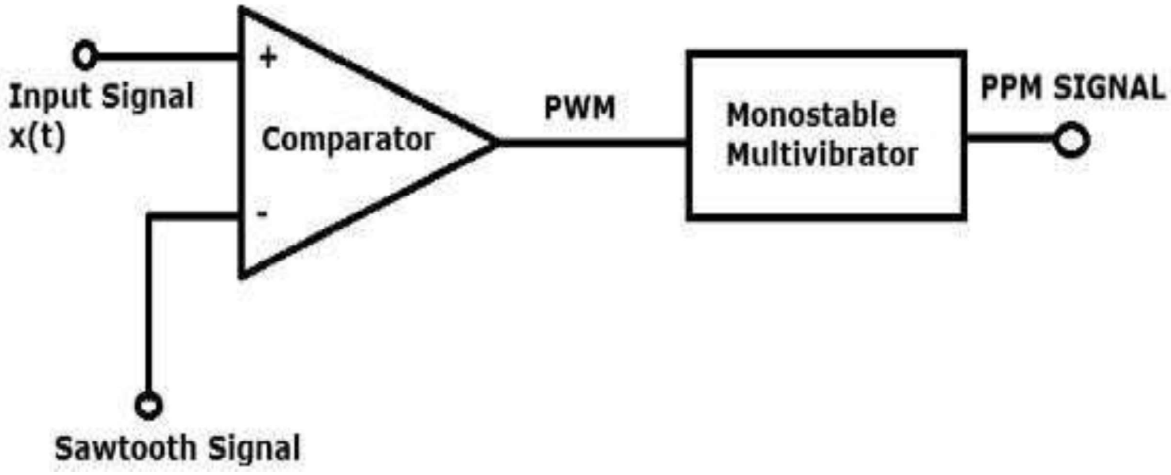
DETECTION OF PWM



PULSE POSITION MODULATION(PPM)

Modulation technique in which position of pulses of carrier pulse train is varied in accordance with amplitude of modulating signal.

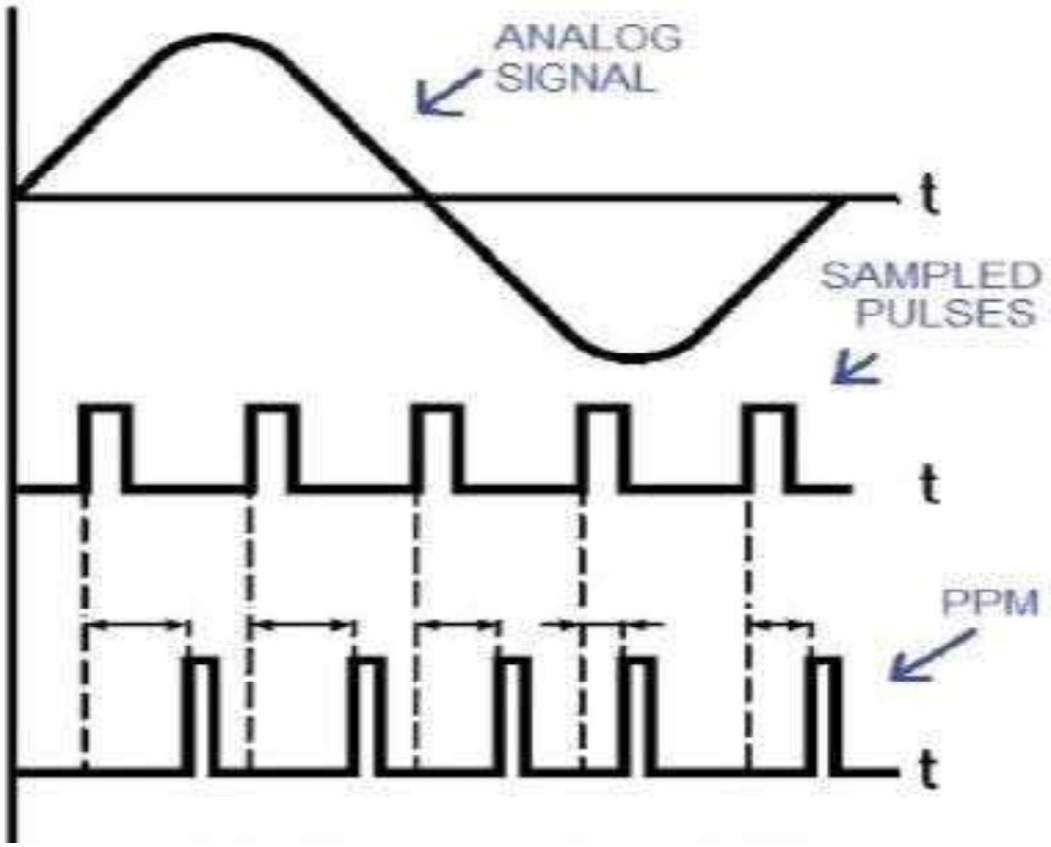
Generation:



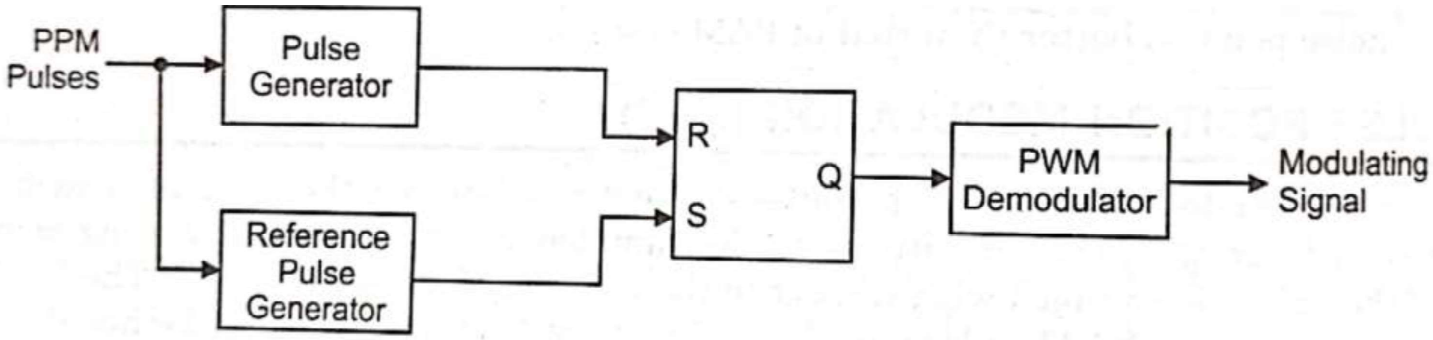
PPM GENERATION

- The block diagram is similar to PWM except monostable multivibrator.
- PWM pulses obtained at the output of comparator are applied to a monostable multivibrator.
- monostable multivibrator is a negative edge triggered circuit. At each trailing edge of PWM signal the monostable output goes high.
- PPM output remains high for a fixed duration from trailing edge of PWM signal.

PPM GENERATION



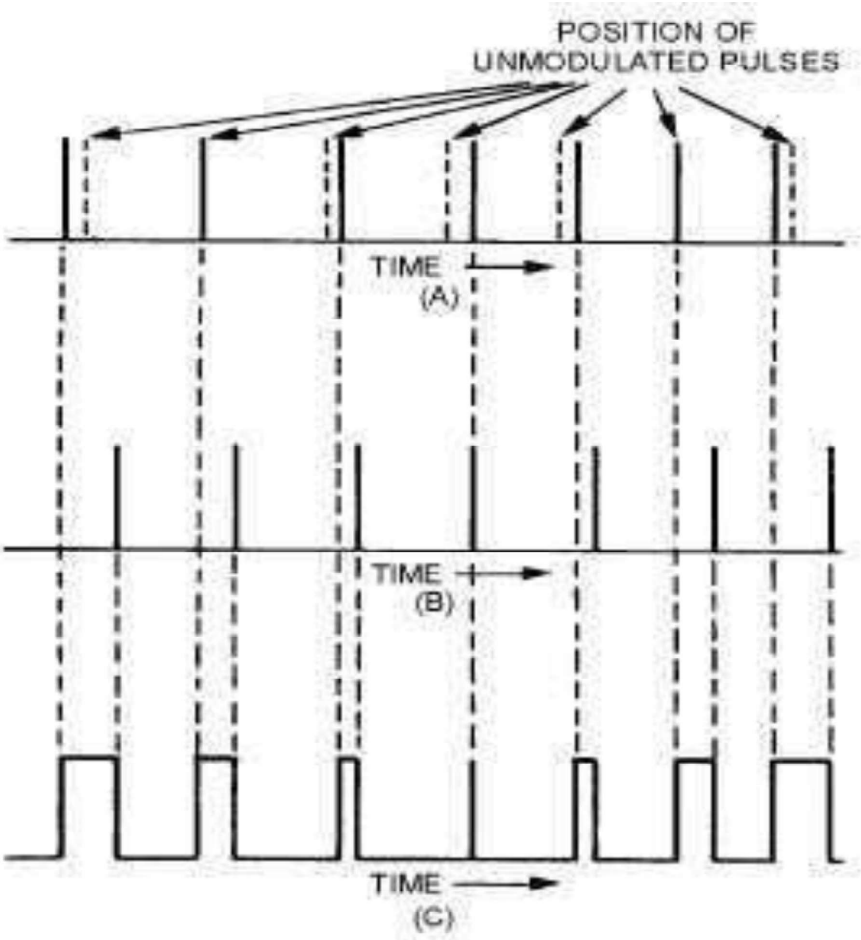
DETECTION OF PPM



DETECTION OF PPM

- The circuit consists of S-R flipflop which is set or gives high output when reference pulses arrive.
- Reference pulses are generated by a reference pulse generator.
- Flip-flop circuit is reset and gives low output at the leading edge of PPM signal.
- The process repeats and we get PWM pulses at the output of flip-flop.
- PWM pulses are then demodulated in a PWM demodulator to get original modulating signal.

DETECTION OF PPM



UNIT VI

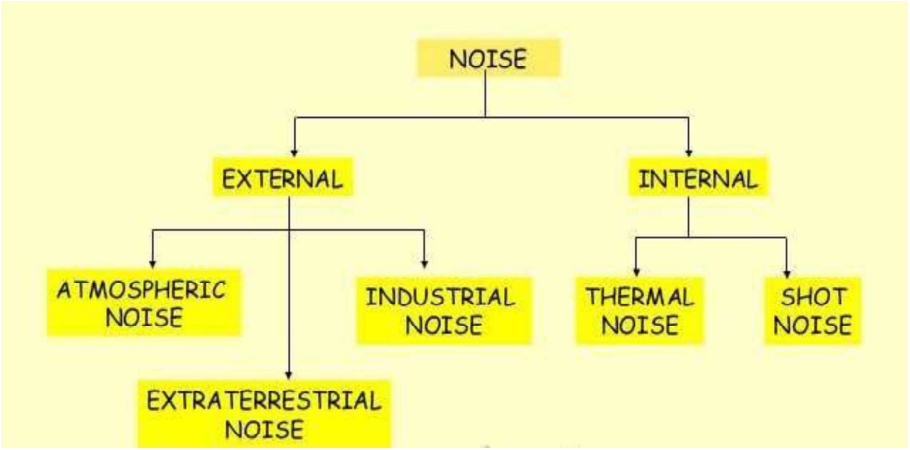
NOISE IN COMMUNICATION SYSTEMS

Noise: It is an unwanted signal which tends to interfere with the modulating signal.

Types of noise:

Noise is basically divided into,

1. External Noise
2. Internal Noise



Classification of Noise

1.External Noise:

- **Atmospheric Noise**: Radio noise caused by natural atmospheric processes, primarily lightening discharges in thunder storms.
- **Extraterrestrial Noise**: Radio disturbances from sources other than those related to the Earth.

  Cosmic Noise: Random noise that originates outside the Earth's atmosphere.

  Solar Noise: Noise that originates from the Sun is called Solar noise.

Classification of Noise

- **Industrial Noise**: Noise generated by automobile ignition, aircrafts, electric motors, Switch gears, welding etc.

2. **Internal Noise**:

- **Shot Noise**: Random motion of electrons in the semiconductor devices generates shot noise.
- **Thermal or Johnson's Noise**: Random motion of electrons in the resistor is called Thermal noise.

$$Vn = KT0BR$$

Where, K= Boltzmann constant, R= Resistance

T0= Absolute temperature B= Bandwidth

Noise Temperature and Noise Figure

Noise temperature(Te): It is a means for specifying noise in terms of an equivalent temperature. It is expressed as ,

$$T_e = (F_n - 1)\, T_0$$

Where, F_n = Noise Figure, T_0 = Absolute temperature

Noise figure(F_n): It is the ratio of output and input noise of an amplifier or network. It is expressed as, $F_n = \frac{KT_0BG + \Delta N}{KT_0BG}$

Where, Δ = Noise added by the network or amplifier.

G = gain of an network or amplifier

Noise Temperature and Noise Figure

Noise Figure of Cascade Amplifier or Network:

Noise Figure of an cascade network or amplifier is expressed as,

$$F_n = F1 + \frac{(F2-1)}{G1} + \frac{F3-1}{G1G2} + \frac{F4-1}{G1G2G3} + \cdots\cdots + \frac{Fn-1}{[G1G2G3\ldots..G(n-1)]}$$

Where, F1= Noise figure of 1st stage

G1= Gain of 1st stage

F2= Noise figure of 2nd stage

G2 = Gain of 2nd stage

Fn = Noise figure of nth stage

Gn = Gain of nth stage

Noise equivalent Bandwidth

When white noise (flat spectrum of frequencies like white light) is passed through a filter having a frequency response , some of the noise power is rejected by the filter and some is passed through to the output.

The noise equivalent bandwidth is defined in the following picture,

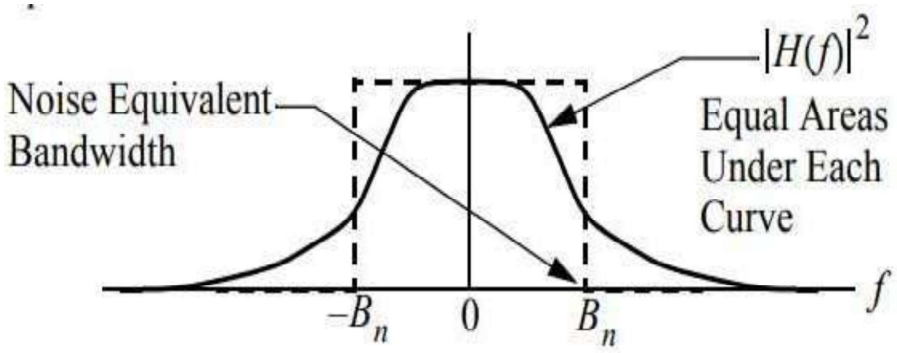


Figure of Merit

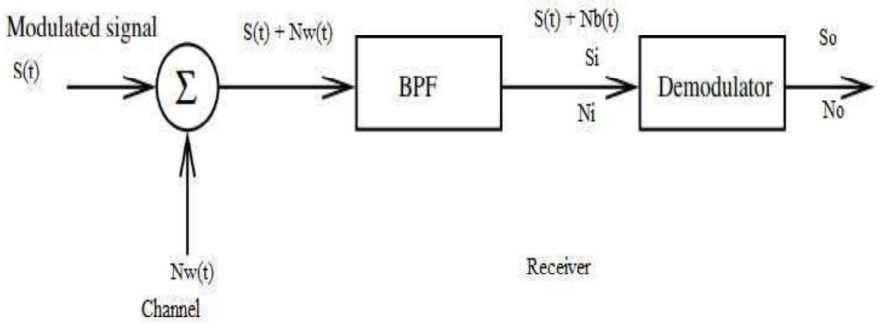
Figure of Merit (FOM): It is ratio of output SNR to input SNR of a communication system.

$$FOM = \frac{S_o/N_o}{S_i/N_i}$$

Where S0= Output Signal Power &N0= Output Noise Power

Si= Input Signal Power &Ni= Input Noise Power

Receiver model for noise calculation:

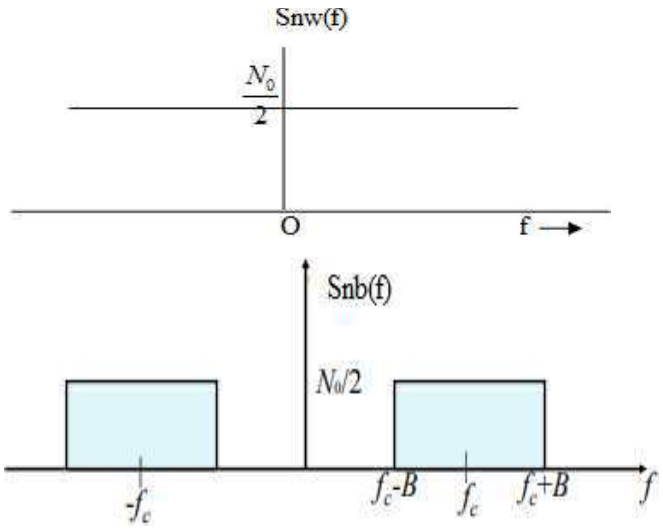


Receiver model for noise calculation

- The receiver is combination of Band Pass Filter (BPF) and Demodulator.
- The BPF is combination of RF Tuned Amplifier, Mixer and Local Oscillator whose band width is equal to band width of modulated signal at transmitter.
- Channel Inter connects transmitter & receiver. Channel adds noise to the modulated signal while transmitting and it is assumed to be white noise whose Power Spectral Density is uniform.
- BPF converts white noise in to color or Band pass noise or narrow band pass noise.

Receiver model for noise calculation

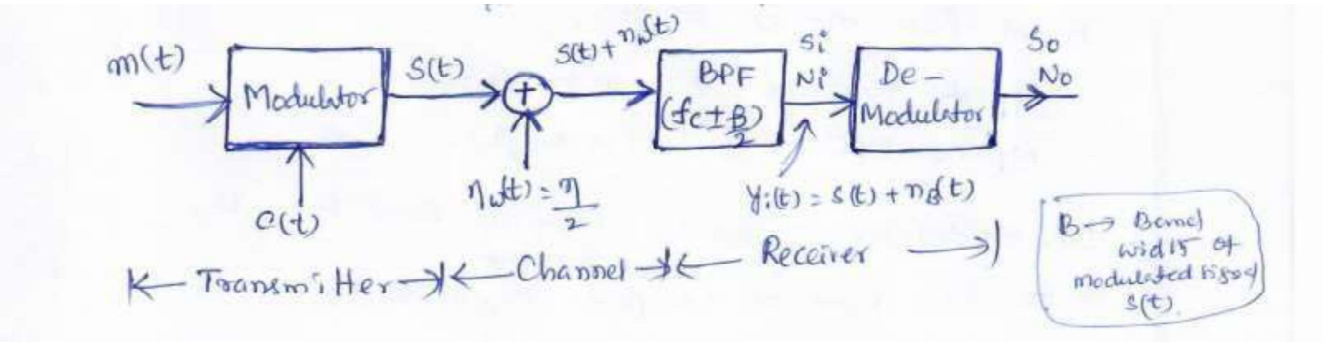
PSD of white noise and Narrow band pass noise are,



Power of band pass noise P = $\int_{f_c-B}^{f_c+B} \frac{N_0}{2}\, df = N_0 B$

Where B = Band width noise.

Communication system model for noise calculation



- The communication system model for noise calculation contains transmitter, channel and receiver.
- Transmitter is replaced by modulator which converts low frequency modulating signal x(t) into high frequency bandpass signal with the help of carrier signal.
- Channel is replaced or modelled as additive noise which adds white noise with PSD η/2 and it contains all frequencies.

Communication system model for noise calculation

- Receiver is modelled as BPF followed by demodulator.
- BPF is combination of RF tuned amplifier, mixer , local oscillator.
- Passband or badnwidth of BPF is equal to bandwidth of modulated signal.
- BPF converts white noise into color or bandpass noise $\eta_B(t)$.

  Input to BPF is $s(t) + \eta_w(t)$

  Output of BPF is $s(t) + \eta_B(t)$
- Demodulator converts high frequency or bandpass signal into low frequency or baseband signal.

Bandpass noise representation

Bandpass noise is represented by,

1. Time Domain representation
   - Quadrature representation
   - Envelope representation
2. Frequency Domain representation

Quadrature representation:

Bandpass noise can be represented as,

$$\eta_B(t) = \eta_i(t) \text{ ocs } W_c t \;\; \eta_q(t) \text{Sin} W_c t$$

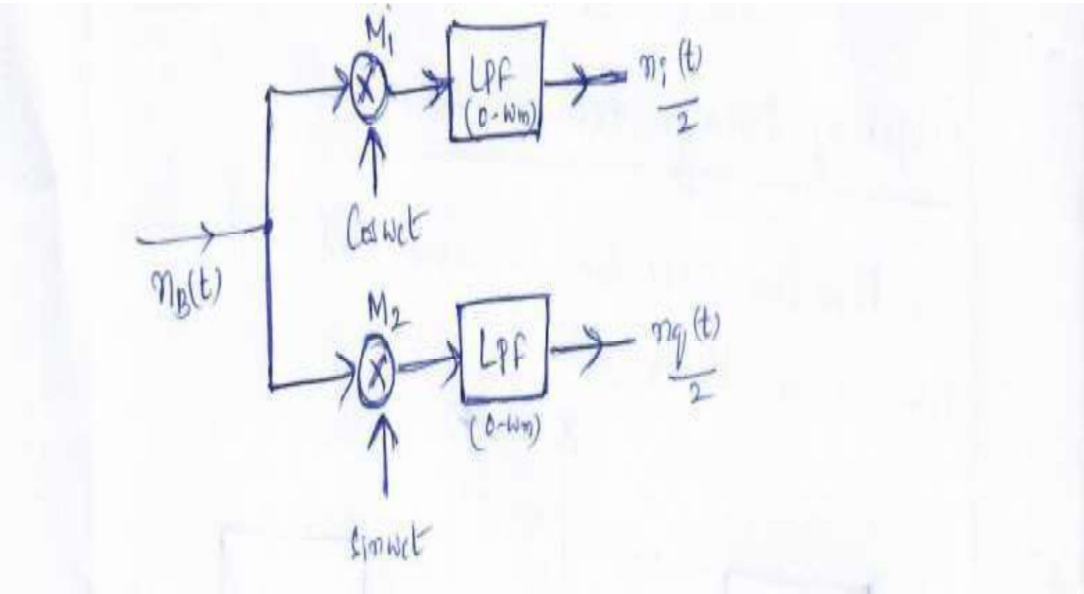
Where, $\eta_B(t)$ = Bnadpass noise

$\eta_i(t)$ = Inphase component of lowpass noise

$\eta_q(t)$ = Quadrature component of lowpass noise

Quadrature representation

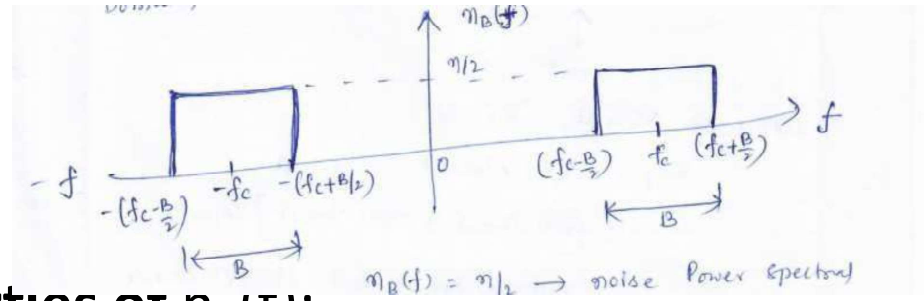
$\eta_i(t)$ and $\eta_q(t)$ can be recovered from $\eta_B(t)$,



Bandpass noise representation

Frequency domain representation:

Bandpass noise can be represented in frequency domain as,



Properties of $\eta_B(t)$:

- $\eta_B(t)$, $\eta_i(t)$, $\eta_q(t)$ will have same power.
- The PSD of $\eta_i(t)$ & $\eta_q(t)$ is,

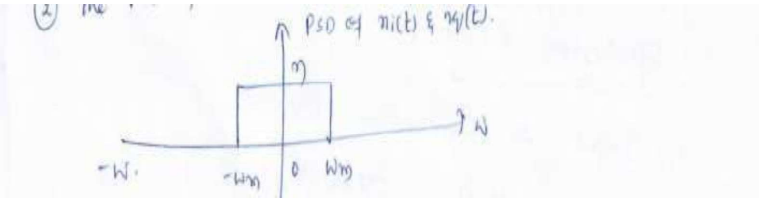
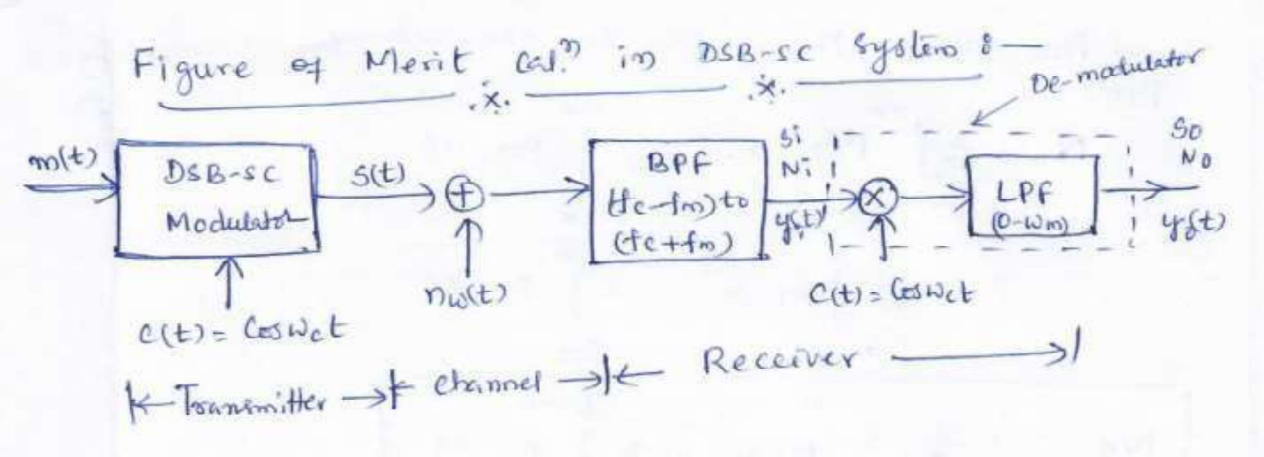


Figure of Merit calculation in DSB-SC



- Transmitter contains DSB-SC modulator, whose output s(t) = m(t) coswct.
- Noise generated by the channel is considered as white noise $\eta_w(t)$ with uniform noise power spectral density $\eta/2$.

Figure of Merit calculation in DSB-SC

- Band pass filter's bandwidth is equal to modulated signal bandwidth.
- BPF allows DSB-SC signal and converts white noise into color noise or bandpass noise $\eta_B(t)$.

  Therefore, o/p of the BPF is $yi(t) = s(t) + \eta_B(t)$.
- Synchronous detector is used to extract modulating signal m(t) which contain multiplier followed by low pass filter.

Input signal power is , $Si = m^2(t)/2$, Input noise power, $Ni = \eta . 2fm$, Output signal power, $S0 = [m(t)/2]^2$, Output noise power, $N0 = \eta . fm/2$

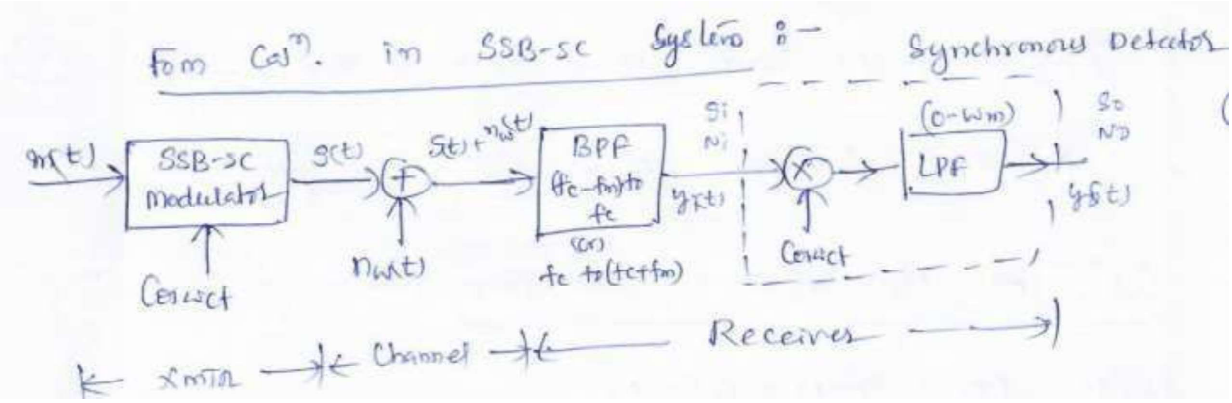
Substituting these values, $FOM=(S0/N0)/(Si/Ni)= 2$

Figure of Merit calculation in SSB-SC

SSB-SC signal, $s(t) = m(t)coswct \pm mh(t)sinwct$

Output of BPF, $y_i(t) = s(t) + \eta_B(t)$

Bandpass noise, $\eta_B(t) = \eta_i(t)\ ocs\ W_ct\ \ \eta_q(t)SinW_ct$



Input signal power is , Si = $m^2(t)$, Input noise power, Ni = η. fm

Output signal power,S0 = $m^2(t)/4$, Output noise power, N0 = η. fm/4

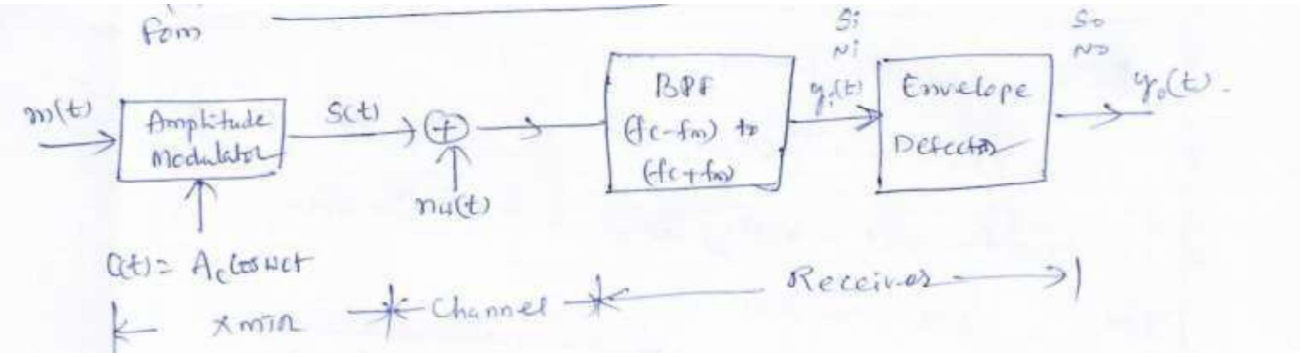
Substituting these values, FOM= (S0/N0)/(Si/Ni)= 1

Noise calculation in AM system

AM signal, $S(t) = [Ac+m(t)] \cos w_c t$

Output of BPF is, $yi(t) = s(t) + \eta_B(t)$

$= [Ac+m(t)] \cos w_c t + \eta_B(t)$



Input signal power is , $Si = [Ac^2/2]+[m^2(t)/2]$

Input noise power, $Ni = 2 \eta . Fm$, Output signal power,$S0 = m^2(t)$

Output noise power, $N0 = 2\eta . Fm$

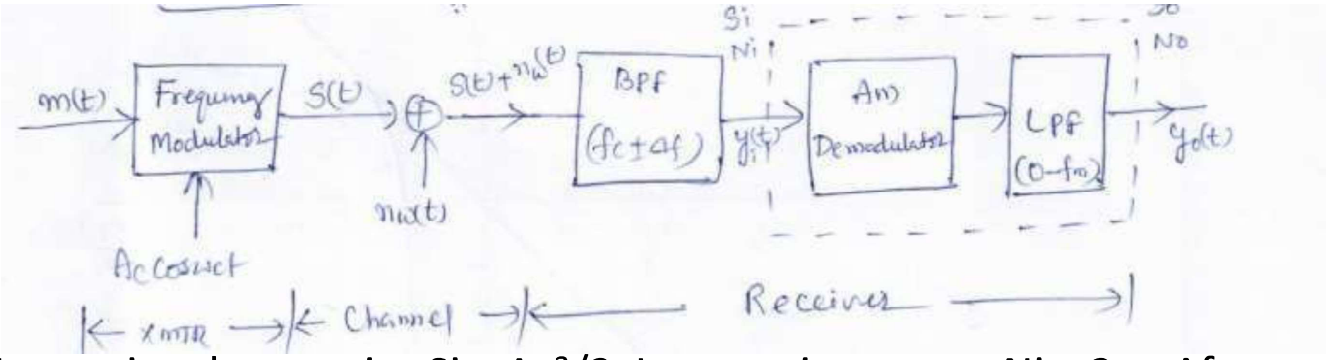
Using these values, FOM= 2.

Noise calculation in FM system

Frequency modulated signal $s(t) = A_c \cos [w_c t + K_f \int m(t)\, dt]$

Output of BPF is, $y_i(t) = s(t) + \eta_B(t)$

$= A_c \cos [w_c t + K_f \int m(t)\, dt] + \eta_B(t)$



Input signal power is , $S_i = A_c^2/2$, Input noise power, $N_i = 2 \eta . \Delta f$

Output signal power, $S0 = \gamma^2 K_f^2 m^2(t)$

Substituting these values, FOM= (S0/N0)/(Si/Ni)

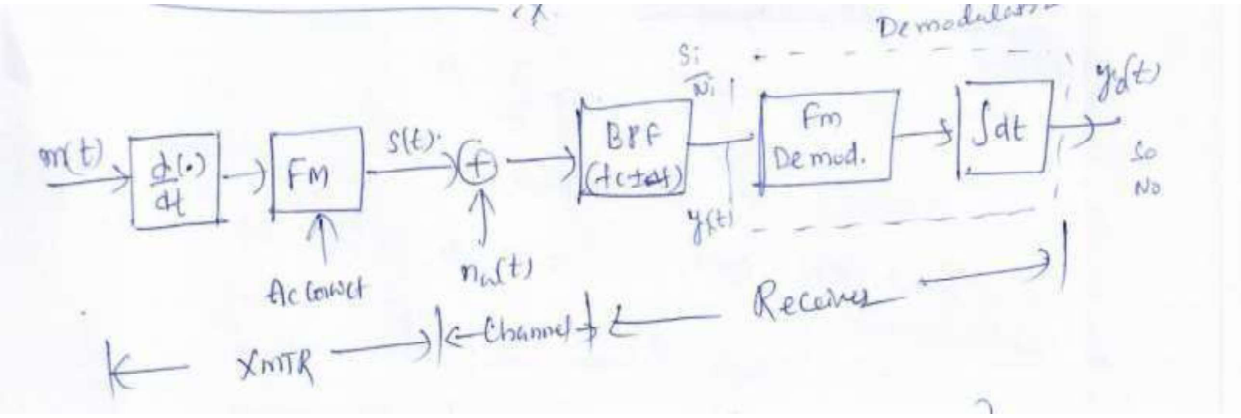
$FOM = (3/4\pi^{2)} m f^{3,}$ Where $m_f = \Delta f / f_m$

Noise calculation in PM system

PM signal $S(t) = A_c \cos[w_c t + K_p m(t)]$

Output of BPF is, $y_i(t) = s(t) + \eta_B(t)$

$= A_c \cos[w_c t + K_p m(t)] + \eta_B(t)$



Input signal power is , $S_i = A_c^2/2$, Input noise power, $N_i = 2\eta . \Delta f$

Output signal power, $S0 = \gamma^2 K_p^2 m^2(t)$, Output noise power, $N0 = 2\eta . Fm$

Substituting these values and substituting $m^2(t) = Am^2/2$

$$FOM = (S0/N0)/(Si/Ni) = m_p^2(\Delta f/ fm)$$

Comparison between different Modulation Systems with respect to FOM

S.No.	Modulation System	FOM
1.	DSB-SC	2
2.	SSB-SC	1
3.	AM with Envelope Detector	$2\frac{\overline{m^2(t)}}{A_c^2+\overline{m^2(t)}}$
4.	AM with Square Law Detector	$2\frac{\overline{m^2(t)}}{A_c^2+\overline{m^2(t)}} \times \frac{1}{1+\frac{\overline{m^2(t)}}{A_c^2}}$
5.	FM	$\frac{3}{4\pi^2}m_f^3$
6.	PM	$m_p^2\frac{\Delta f}{f_m}$

Analog Communication

Assignment 1

1. (a) Derive an expression for single tone AM wave. Assume sinusoidal carrier and modulating signal. Sketch its signal and spectrum. (b) The antenna current of an AM transmitter is 9 A when only the carrier is sent, but it increases to 10.6 A when the carrier is modulated by a single sine wave. Find the percentage modulation. Determine the antenna current when the percentage of modulation changes to 0.8. CO2 & CO3
2. (a) Draw the circuit diagram of Ring modulator and explain about it. (b) An AM transmitter radiates 1 KW when the modulation percentage is 50%. How much Carrier power is required if we want to transmit the same message by an AM DSBSC transmitter. CO2 & CO3
3. (a) Explain the generation FM using direct method. (b) The equation for a FM wave is s (t) = 10sin [15.7 x 10^8 t + 5 sin 12 x 10^3 t]. Calculate: (i) Carrier frequency. (ii) Modulating frequency. (iii) Modulation index. (iv)Frequency deviation. (v) Power dissipated in 100 Ω. CO2 & CO3
4. Write short notes on: (a) VSB. (b) Phase shift method of AMSSBSC generation. CO2
5. (a) Describe Elements of Communication System. (b) What are the types of Communication System? (c) What is the need for modulation? Explain. CO1

Analog Communication
Assignment-1

P. Yaswanth Reddy
20091A0490
ECE-C section 3rd Year

(1) (a) Derive an expression for single tone AM wave from assume sinusoidal carrier and modulating signal and sketch its signal and spectrum.

Ans Let the carrier voltage

$$V_c = V_c \sin \omega_c t \quad -(1)$$

modulating voltage $V_m = V_m \sin \omega_m t \quad -(2)$

modulation index $\boxed{(m) = \frac{V_m}{V_c}} \quad -(3)$

equation for amplitude of AM wave is given by $A = V_c + V_m$

$$A = V_c + V_m \sin \omega_m t$$

$$A = V_c \left(1 + \frac{V_m}{V_c} \sin \omega_m t\right)$$

$$\boxed{A = V_c (1 + m \sin \omega_m t)} \quad -(4)$$

instantenous voltage of resulting AM wave is given as $V = A \sin\theta$

$$V = A \sin \omega_c t$$

$$V = V_c (1 + m \sin \omega_m t) \sin \omega_c t \quad -(5)$$

$$V = V_c \sin \omega_c t + m V_c \sin \omega_m t \sin \omega_c t$$

$$\because \sin A \sin B = \frac{1}{2}\left(\cos(A-B) + \cos(A+B)\right)$$

$$V = V_c \sin \omega_c t + \frac{mV_c}{2}\left[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t\right]$$

$$V = V_c \sin \omega_c t + \frac{mV_c}{2}\cos(\omega_c - \omega_m)t - \frac{mV_c}{2}\cos(\omega_c + \omega_m)t$$

$$\boxed{V = V_c \sin \omega_c t + \frac{mV_c}{2}\cos(\omega_c - \omega_m)t - \frac{mV_c}{2}\cos(\omega_c + \omega_m)t} \quad -(6)$$

↓ Carrier wave ↓ LSB ↓ USB

Time domain :-

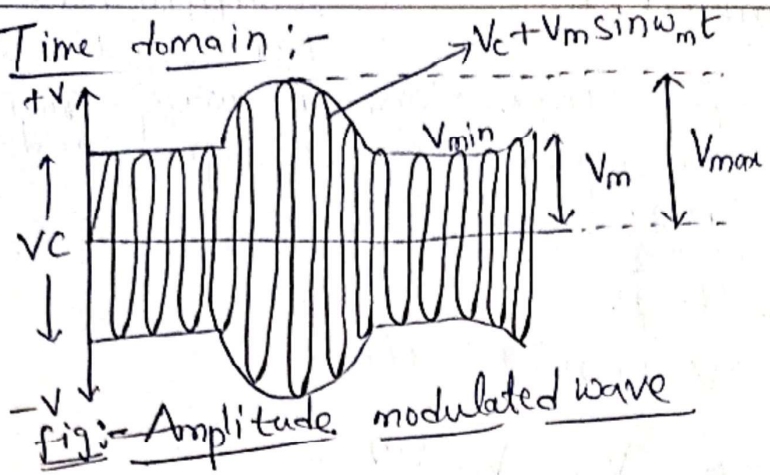


fig:- Amplitude modulated wave

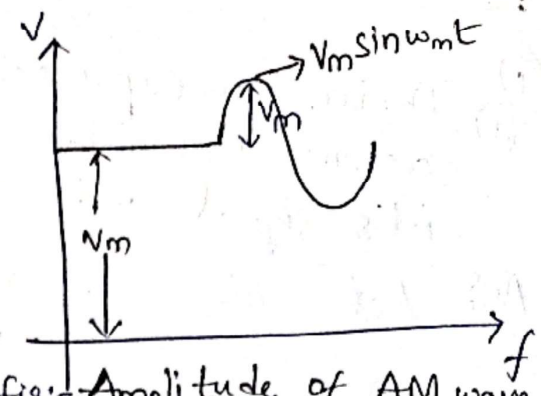
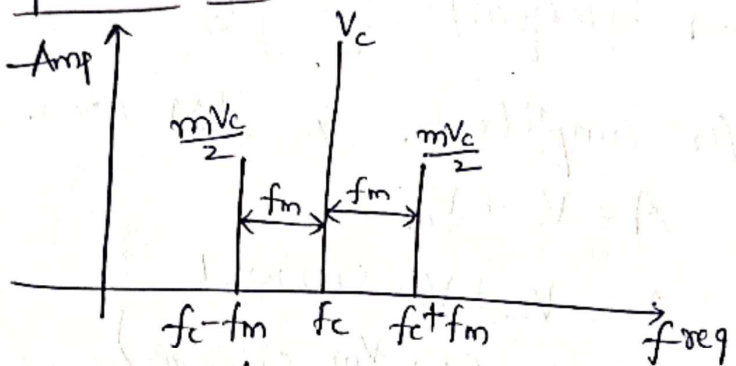


fig:- Amplitude of AM wave

Frequency Spectrum of AM wave :-



(1) (b) The antenna current is 9 Amp when only carrier is sent but it increased 10.6 Amp when the carrier modulated by a single wave. find % modulation. Determine the antenna when the % of modulation changes to 0.8.

Ans Given $I_c = 9\text{Amp}$; $I_t = 10.6$

WKT $\left(\frac{I_t}{I_c}\right)^2 = 1 + \frac{m^2}{2}$

$$m = \sqrt{2\left[\left(\frac{I_t}{I_c}\right)^2 - 1\right]}$$

Now $$m = \sqrt{2 \times \left(\left(\frac{10.6}{9}\right)^2 - 1\right)}$$

$$\boxed{m = 87.99\ \%}$$

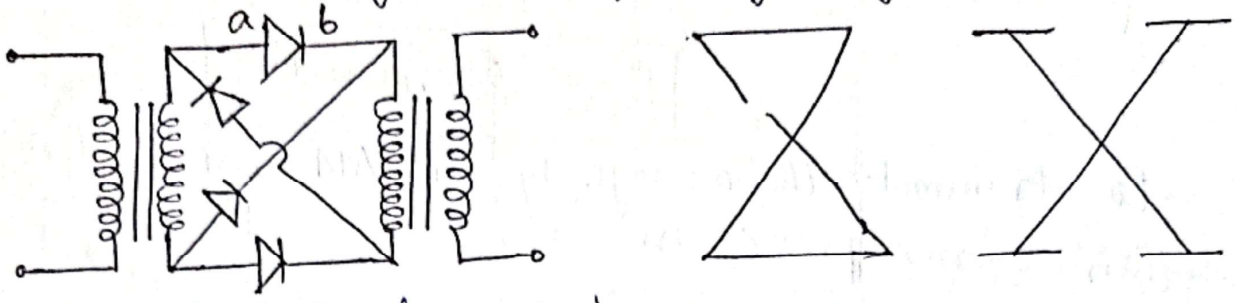
for the second part we have

$$I_t = I_c \sqrt{1 + \frac{m^2}{2}} = 9\sqrt{1 + \frac{(0.8)^2}{2}}$$

$$\boxed{I_t = 10.34\ \text{Amp}}$$

(2) (a) Draw the circuit of ring modulator and explain it.

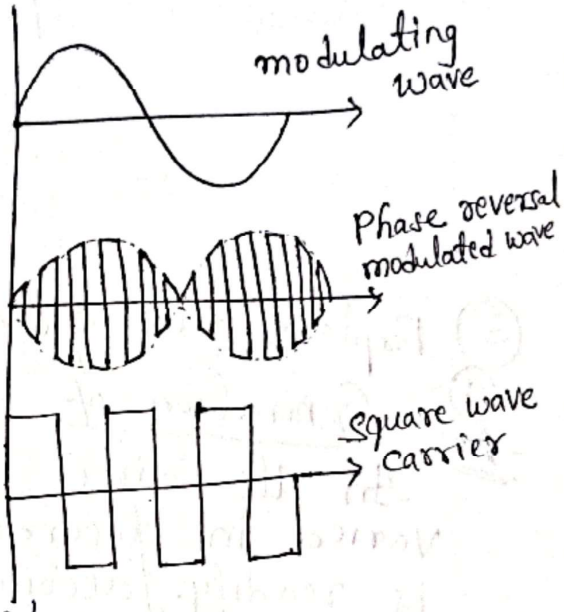
Ans. DSB-SC can be generated by using ring modulator.



Ring modulator is also called as lattice modulator (or) double balanced modulator. It is the one of the most useful product modulator for generic DSBSC modulated wave.

→ The circuit consists 4 diodes which for a ring they all point in the same way the diodes are controlled by square wave c(t) of frequency f(c) by means of 2 centre-tapped transform.

→ We assumed that diodes are ideal the transformer are perfectly balanced.



→ In a carrier supply is +ve, the outer the are switched on present in zero impedance inner diodes are present in infinite input impedance. So that modulator multiply message signal m(t) by +1.

→ When carrier supply is -ve, inner diodes are switched on present in zero impedance outer diodes are switched off present in infinite impedance so that modulator multiplies message signal m(t) by -1.

Therefore the ring modulator in if ideal form is a product modulator for squarewave carrier & message signal.

(2) (b) An AM transmitter radiates 1kW when the modulation % is 50 and how much carrier power is required if we want to transmit the same message by an AM DSBSC transmitter.

Ans Given $P_t = 1kW$; $m = 50\% = 0.5$

$$P_t = P_c\left(1+\frac{m^2}{2}\right)$$

$$P_c = \frac{P_t}{\left(1+\frac{m^2}{2}\right)} = \frac{1k}{1+\frac{0.5^2}{2}}$$

$$\boxed{P_c = 888.88 \text{ watt}}$$

to transmit the message by an AM DSB-SC transmitter then carrier power is

$$P_t = P_c \frac{m^2}{2}$$

$$P_c = \frac{2P_t}{m^2} = \frac{2\times 1\times 10^3}{(0.5)^2}$$

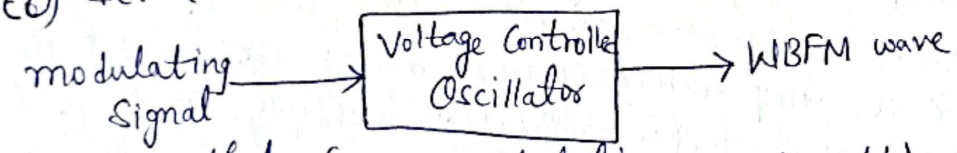
$$\boxed{P_c = 8000 \text{ Watt}}$$

(5) Explain the generation of FM using direct method?

(a) Ans

Generation of FM signal using direct method:

In the direct method the carrier frequency is directly varied in accordance with the input baseband signal which is readily accomplished using voltage controlled oscillator (VCO) device.



Assume that for a modulating wave m(t), the capacitance c(t) is expressed as follows

$$c(t) = C_0 - kCm(t) \quad -(2)$$

$$f_i(t) = \frac{1}{2\pi} * \frac{1}{\sqrt{L_1+L_2 \cdot C(t)}}$$

where C_0 is the total capacitance in absence of modulation, k_c is the variable capacitors sensitivity of voltage change.

Sub eqn(2) in eqn(1)

$$f_i(t) = \frac{1}{2\pi\sqrt{L_1+L_2\ C_0 - k_c m(t)}}$$

$$f_i(t) = \frac{1}{2\pi\sqrt{C_0(L_1+L_2)\left(1-\frac{k_c m(t)}{C_0}\right)}}$$

where $f_0 = \frac{1}{2\pi\sqrt{C_0(L_1+L_2)}}$

$$f_i(t) = f_0\left(1-K\left(\frac{m_t}{C_0}\right)\right)^{-1/2} \quad -(3)$$

where f_0 is unmodulated frequency of oscillation $m(t)=0$

$$f_i = \frac{1}{2\pi\sqrt{C_0(L_1+L_2)}} \quad -(4)$$

provided that the maximum change in capacitor produced by modulating wave is small compared with unmodulated capacitance.

we approximate eqn (3) follows

$$f_i(t) = f_0\left(1-\frac{k_c}{C_0}m(t)\right)^{-1/2}$$

$$f_i(t) = f_0\left(1+\frac{1}{2}\frac{k_c}{C_0}m(t)\right) \quad -(5)$$

$$f_i(t) = f_0 + \frac{f_0}{2}\frac{k_c}{2}m(t)$$

where $k_f = \frac{f_0 k_c}{2w}$; $f_i(t) = f_0 + k_f m(t)$

where K_f = frequency sensitivity

(3)(5) The eqn for a FM wave is $s(t) = 10\sin[(5.7\times10^8)t + 5\sin((12\times10^3 t)]$. Calculate (i) f_c (ii) f_m (iii) m (iv) freq deviation (v) power desipated in 100 Ω resistance.

Ans Given eqn for a FM wave is

$$s(t) = 10\sin\left[(5.7\times10^8)t + 5\sin(12\times10^3)t\right] \quad -(1)$$

original eqn for FM wave is

$$s(t) = A\sin(\omega_c t + m_f \sin\omega_m t) \quad -(2)$$

Compare (1) & (2)

$\therefore A = 10$; $\omega_c = 5.7\times10^8$; $\omega_m = 12\times10^3$

now $2\pi f_c = 5.7\times10^8$

$$f_c = \frac{5.7\times10^8}{2\pi}$$

$$\boxed{f_c = 90.7\,MHz}$$

now $2\pi f_m = 12\times10^3$

$$f_m = \frac{12\times10^3}{2\pi}$$

$$\boxed{f_m = 1.90\,kHz}$$

modulation index $m = 5$

maximum deviation $\delta = V_m f_m = 5 \times 1.90 \times 10^3$

$\delta = 9.5\ kHz$

Power will desipated in 100Ω resistor $P = \frac{V_{rms}^2}{R}$

$$P = \frac{\left(\frac{5}{\sqrt{2}}\right)^2}{100} = \frac{\frac{25}{2}}{100} = 0.125$$

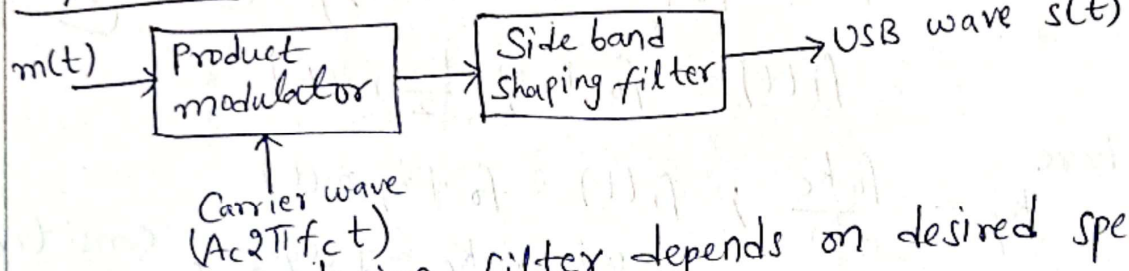
∴ power desipated $P = 0.125$ watt.

(4) Write short notes on VSB.

(a) Ans In VSB modulation one side band is passed almost completely where as a vestige (a small portion) of the other side is retained.

Applications :- TV signal

Generation of USB modulated wave :-



The exact design filter depends on desired spectrum of USB modulation wave.

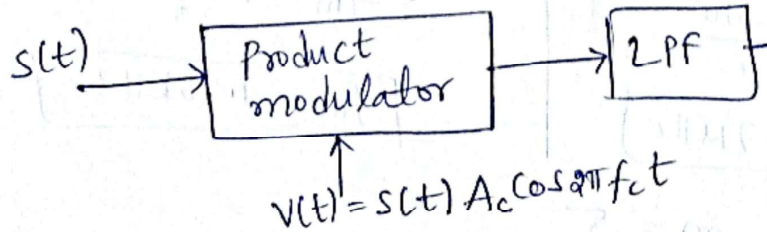
The relation b/w transfer function of their filter and Spectrum of USB modulated wave defined by

$$S(f) = \frac{Ac}{2}\left[m(f-fc) + m(f+fc)\right] H(f) \quad \text{—(1)}$$

where $m(f)$ = message spectrum

DSBFC spectrum T.F = $\frac{o/p}{i/p}$

This filter specification can established by passing $s(t)$ by a coherent detector and the determing the necessary condition for the dector o/p to provide and started of original undistortion message signal.

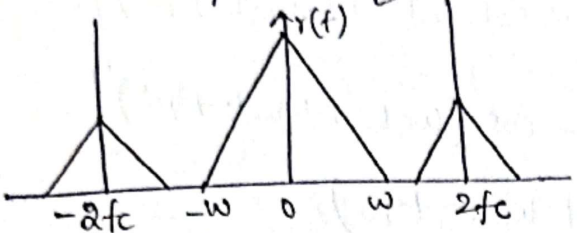


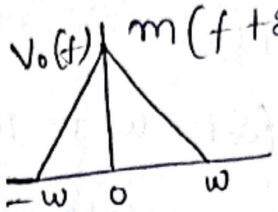
DSBFC spectrum

$$V(f) = \frac{1}{2}[s(f-f_c) + s(f+f_c)] \quad -③$$

Sub eqn ② in eqn ① we get

$$A(F) = \frac{A_c}{4} m(f)[H(F-f_c) + H(F+f_c)] + \frac{A_c}{4}[m(f-2f_c) * H(F-f_c) + m(f+2f_c) H(f+f_c)] \quad -④$$





Bandwidth of USB $B = f_v + w$

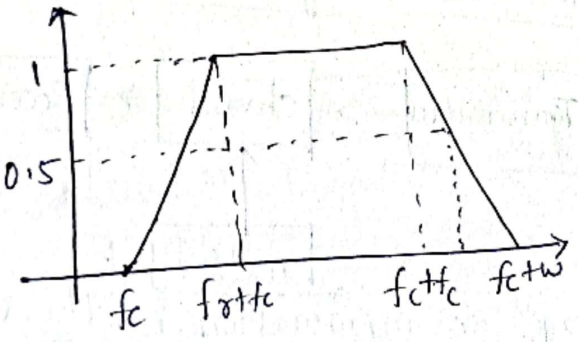
for distortion less original base band $m(f)$ at the coherent detector output we require $V_o(t)$ to the scaled version $m(f)$ this means the transformation $H(f)$ most satisfy the condition

$$H(f-f_c) + H(f+f_c) = 2H(f_c) \quad -⑤$$

where $H(f_c)$ is a constant so this required eqn ⑤ by using filter with a frequency response $H(f)$

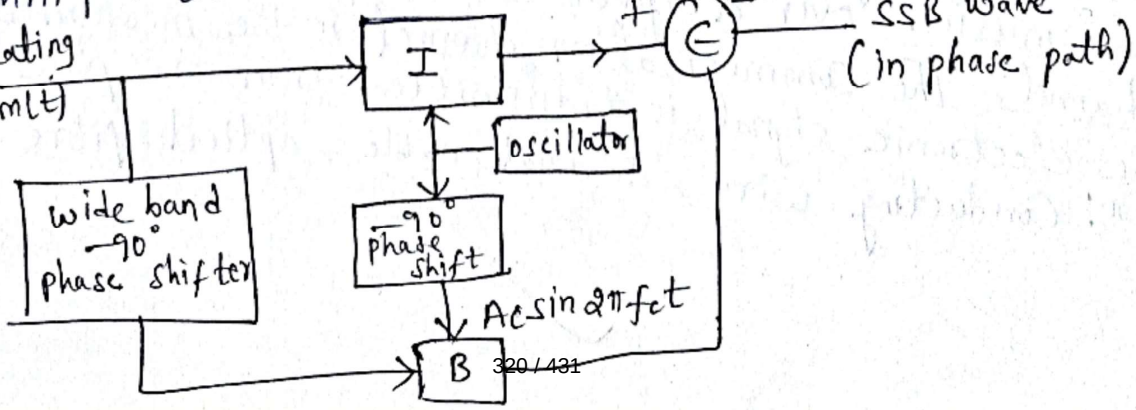
Frequency response of sideband shaping filter:

for a VSB an odd wave sideband shaping of LSB.



(4) (6) Write a short notes on Phase shift method of AM SSBSC generation.

Ans The phasing method of SSBSC generation uses a phase shift technique that causes one of the side band cancelled out.



$$V_1 = \cos[(\omega_c t + 90°) - \omega_m t] - \cos[(\omega_c t + 90°) + \omega_m t]$$

$$V_1 = \cos(\omega_c t - \omega_m t + 90°) - \cos(\omega_c t + \omega_m t + 90°)$$

LSB USB

Similarly

$$V_2 = \cos[(\omega_c t - (\omega_m + 90°) - \cos\omega_c t + (\omega_m t + 90°)]$$

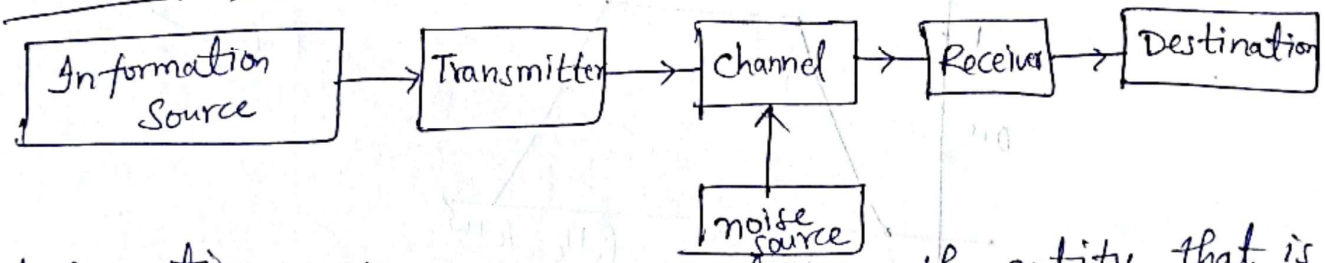
$$= \cos[(\omega_c t - \omega_m t - 90°) - \cos(\omega_c t + \omega_m t + 90°)]$$

$$V_0 = V_1 + V_2 = 2\cos(\omega_c t + \omega_m t + 90°)$$

The first term of the first eqn is 180° out of phase with the first term of the second we have proved one of the sidebands in the adder is cancelled the other is reinforced.

(5) (i) Describe elements in of communication system.
(ii) What are the type of communication system.
(iii) What is need for modulation explain.

Ans Elements of Communication System;-



Information;- Message or information is the entity that is to be transmitted it can be in form of Audio, Video, temp, pressure.

Transmitter;- The transmitter is a collection of electronic circuits designed to convert into a signal suitable for transmission over a given communication.

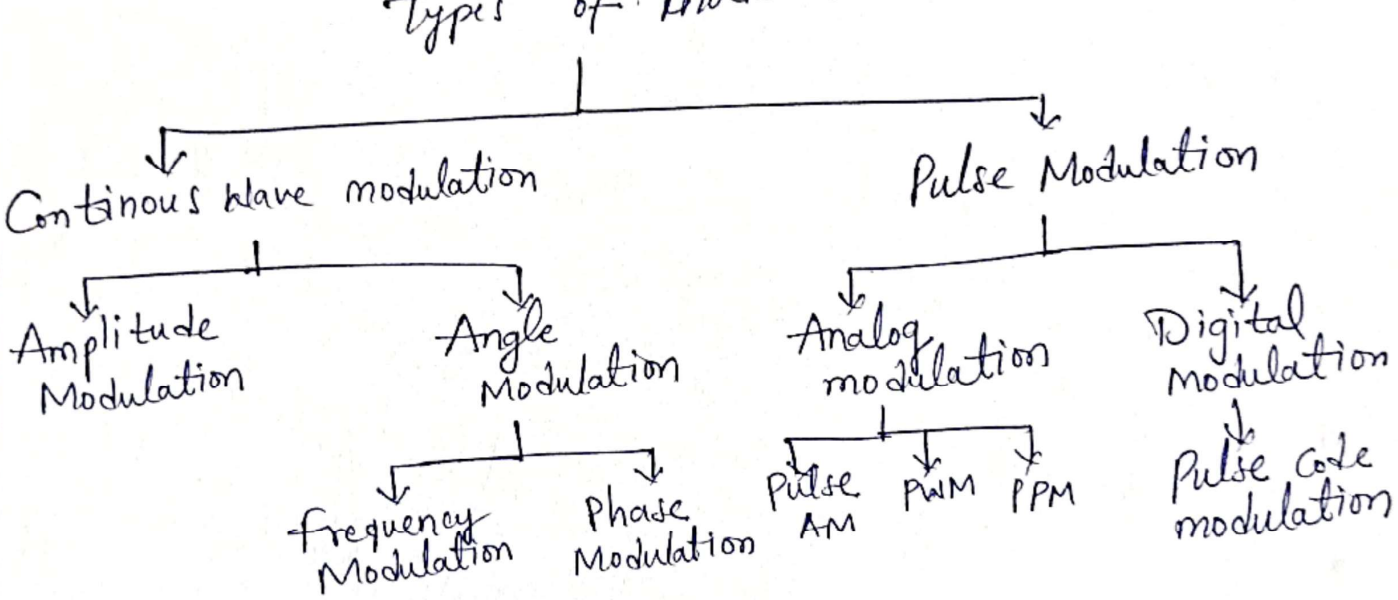
Channel;- The communication channel is the medium by which the electronic signal is transmitted from one place to another Ex: Conducting wire, co-axial cable, optical fibre cable.

Noise :- noise is random, undesirable, electric energy that enters the communication system via the medium and interferes with the transmitted message. Same noise produced in the receiver.

Receivers :- A receiver is collection of electronic Circuits designed to convert the signal back to the original information.

Output transducer :- Convert the electrical signal back to original form.

(ii) **Types of Communication System**



(iii) **Need for modulation :-**

Practically for antenna, Base band are in Communication for direct transmission is used to convey signals from one place to another allow frequency translation.

1. Frequency Multiplexing
2. Reduce the antenna height
3. avoids mixing of signal
4. efficient transmission
5. Reduced noise and interference.

DRS

10

R-20

College Code: 09

Rajeev Gandhi Memorial College of Engineering & Technology

Autonomous

NANDYAL-518501

III B.TechI-Semester- Mid-I *Examinations*

Subject Name: Analog Communications

Branch : ECE

Max. Marks: 20 Date: 27-09-2022 Time: 2 Hours

Note: 1.Answer first question compulsorily. (5 x 1= 5Marks)

2.Answer Any *THREE* from 2 to 5 questions.(3 x 5 = 15 Marks)

Q.No			M	CO	BL
Q.1	a)	What is Carson's rule and write the equation for NBFM?	1M	CO3	BL1
	b)	Write any two differences between DSB-FC,DSB-SC,SSB-SC and VSB	1M	CO1	BL3
	c)	Determine the power content of the carrier and each of the sidebands for an AM signal having a percent modulation of 85% and a total power of 1200W.	1M	CO1	BL3
	d)	Define AM and draw the frequency spectrum of DSB-FC.	1M	CO2	BL2
	e)	What is need for modulation ?	1M	CO2	BL1
Q.2	a)	Define modulation index and draw the diagram of critical and over modulation, mentioning conditions.	2M	CO2	BL2
	b)	Suppose that modulating signal m(t) = Acos(2πfmt), fm<<fc, Determine the DSB-FC signal and its upper and lower sidebands, Draw the spectrum. A 1MHz carrier with amplitude of 1V peak is modulated by a 1KHz signal with modulation index 0.5. Sketch the voltage spectrum.	3M	CO1	BL2
Q.3	a)	Find the frequency deviation and carrier swing for a FM signal which has a resting frequency of 105MHz and whose upper frequency is 105.007MHz when modulated by a particular wave . Find the lowest frequency reached by the FM wave.	2M	CO1	BL3
	b)	With a neat diagram, explain about indirect method of FM generation.	3M	CO2	BL2
Q.4	a)	What is quadrature null effect ?	2M	CO1	BL1
	b)	Draw the block diagram of communication system and explain the function of each block.	3M	CO1	BL2

Q.5	**a)**	Write any two applications of AM and FM.	**2M**	CO1	BL1
	b)	Draw the circuit diagram of ring modulator and explain its operation .	**3M**	CO3	BL2

S.No. 30208

RAJEEV GANDHI MEMORIAL COLLEGE OF ENGINEERING & TECHNOLOGY

AUTONOMOUS

Accredited by NAAC of UGC, New Delhi with 'A' Grade:: Accredited by NBA

Affiliated to JNT University Anantapur

Nandyal - 518 501, Kurnool Dist, A.P.

(ESTD-1995)

INTERNAL EXAMINATIONS ANSWER BOOKLET

NAME OF THE STUDENT: C. Kalyani Reg. No. 2 0 0 9 1 A 0 4 6 3

Principal RGMCET Autonomous NANDYAL

	1	2	3	4	5
A	0	1		0	2
B	0	0		3	1.5
C	0.5				
D	0				
E	1				
Total	1.5	1	–	3	3.5

Grand Total :(In Figures) 09

(in Words): zero nine

NAME OF THE SUBJECT: AC

INTERNAL EXAM : I / II

Date of Exam: 27-09-2022 (FN/AN)

Course : B.Tech. / M.Tech./ MBA / MCA

Year : III rd year Sem.: I sem

Branch: ECE

B.V. Priya 27/09/22

Signature of the Invigilator

(Start Writing From Here)

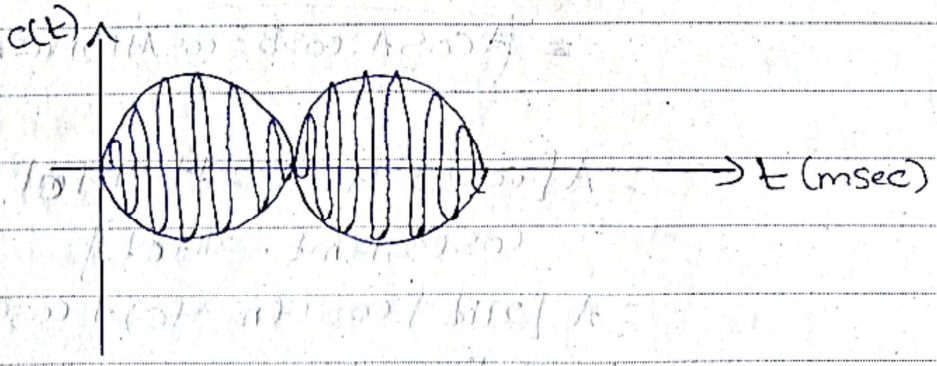
2)a) **Modulation index:-**

The modulation index is the form of carrier & frequency of the modulated signal is called modulation index.

$$\beta = \frac{V_m}{V_C}$$

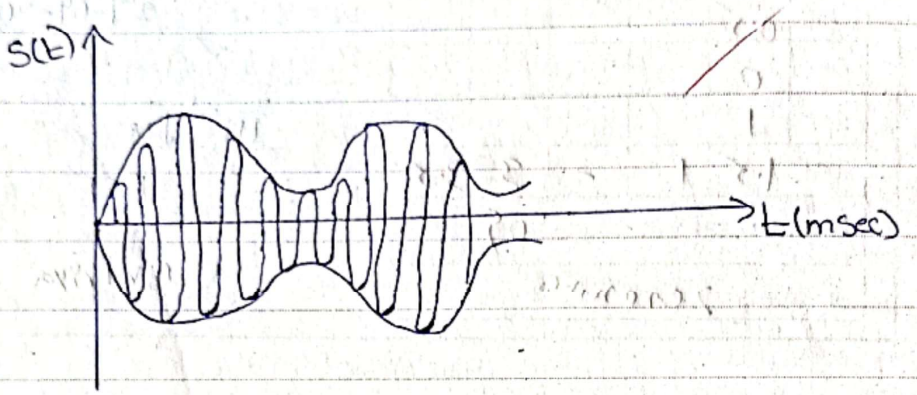
The modulation index is not have a units. It is a efficiency is 0% & 50%.

Critical modulation:-



*The critical modulations checks the signal of frequencies high (or) low.
* The critical modulations of Vmax is amplitude & frequency are there.
* The Vmin of amplitude & frequency signal is zero.

over modulation:-



* The over modulation is a Vmax signal is high frequency & high Amplitude.
* The Vmin of signal is low frequency & low Amplitude.
* This the over modulation.

b) Given that $m(t) = A\cos(2\pi f_m t)$

The DSB-FC signal is

$= m(t) A\cos(2\pi f_c t)$

$= A\cos(2\pi f_m t) A\cos(2\pi f_c t)$

$= \cos A \cos B = \cos(A+B) + \cos(A-B)$

$= A[\cos(2\pi f_m t + 2\pi f_c t) + \cos(2\pi f_m t - 2\pi f_c t)]$

$= A[2\pi t[\cos(f_m - f_c) + \cos(f_m + f_c)]$

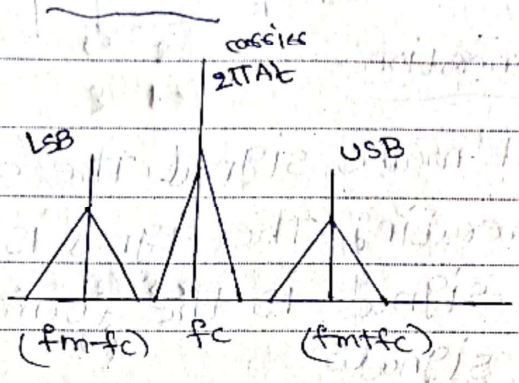
$= A\,2\pi t[\cos(f_m - f_c) + \cos(f_m + f_c)]$

USB = cos(fm+fc)

LSB = cos(fm-fc)

carriers = 2πAt

Spectrum of DSB-FC :-

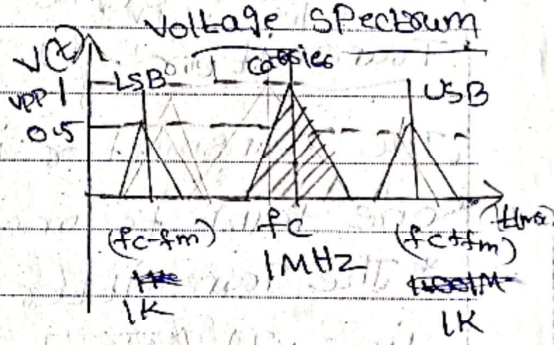


fc = 1MHz

Amplitude (V) = 1V pp

Modulated Signal (fm) = 1kHz

modulation index = 0.5



~~LSB = fc - fm~~ ~~= 1M - 1K~~ ~~= 1K~~

~~USB = fc + fm~~ ~~= 1M + 1K~~ ~~= 1.001M~~

4)a) Quadrature null effect :-

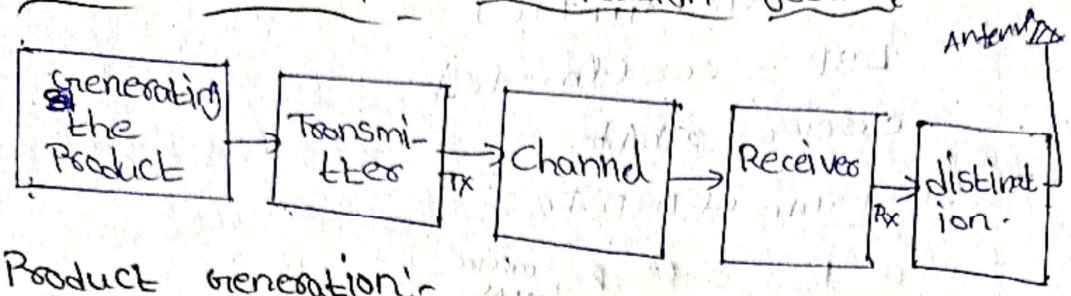
Quadrature null effect is a the signal is a ~~not~~ Amplitude modulation & frequency modulation ~~signal~~ signals are used in this. The Quadrature is a square wave Produces the signal. The Quadrature wave is reduce the frequency & remove the noise in the signals.

The Quadrature null effect is used in the certain signal is forms the waves. The signal is not disturb the noise & distabances in ~~this~~ used in this effect.

null effect is used in the some signals doesn't change the signal waveform.

b) ~~Communication s~~

Block diagram of communication system:-



Product Generation:-

The Product means signal. The signal is first generating. The signal is correct (or) not. The signal is the form of modulation & carrier signals.

Transmitter:-

* The Transmitter is a transmitt the signal one Point to another Point.
* The Transmitter is only transmit only one line. It takes only transmitted.

Channel:-

The channel is a use full to the signal flow of element. It is the channel. The channel is check the Pictures of TV.
* It check the AM & FM signals.

Receiver:-

The Receiver is take the signal from the channel. The transmitter is give signal into channel. The channel is check the condition & go through the Receiver. Receiver is only takes signal.

destination:-

The Receiver is got the destination of

the signal. They reach the destination is give to the signal in Antenna.
It is the Process of the communication system.

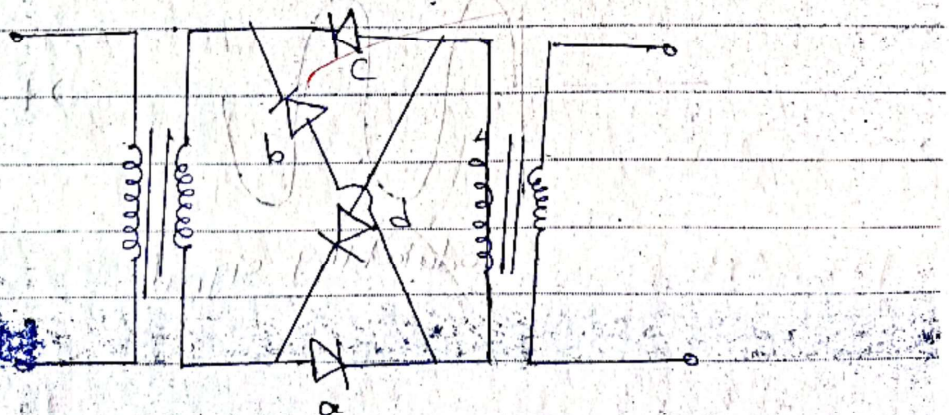
5)a) AM applications:-

* AM signal is the applying for Power modulation.
* It is the reduces high noise.
* It is the applying for low frequency.
* AM is TV communication.

FM applications:-

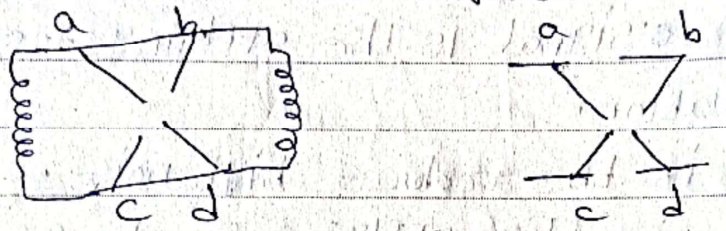
* The FM is applying the angular modulation.
* It is the FM applying the choose various signals.
* FM wave is the Radio communications are used.

b) Ring modulator circuit diagram:-



Ring modulation is the diodes have.

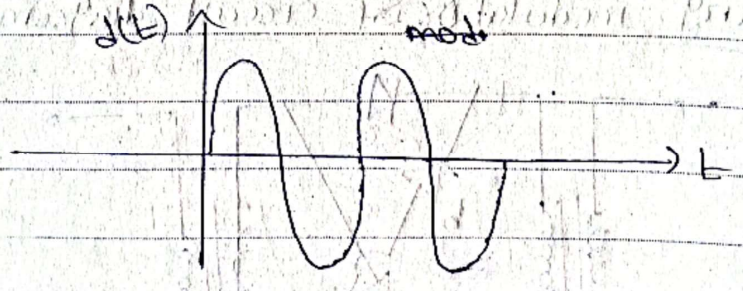
* The diodes are 4.
* The Ring modulation is a transmitter of the signal it form of diodes.
* The AM & FM signals are also Passed.
* The system of the modulation is various of the signals.



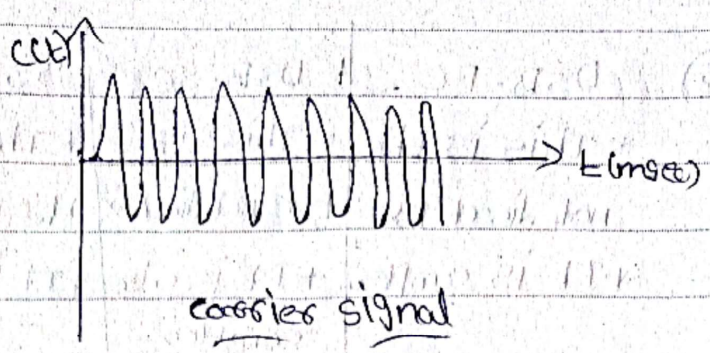
* It is the four diodes are in the form of switches.
* The a & d are one switch.
* The d & c are one switch.
* The a&d are closed the modulation is a signal forms help of c&d.
* The negative wave form is a one.

~~spectrums of th~~

signals of this are

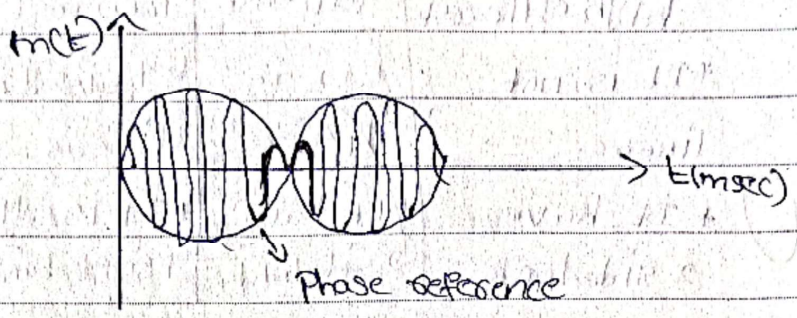


modulating signal.



carrier signal

Modulated signal:-



15

It is gives the phase reference signal.

* The ring modulation is from DSB-SBC form.

1)a) Carson's Rule:-

The carson's rule is a form of angular wave continous in the signals.

The modulation of ~~carr~~ carrier signals are used in the carson's rule.

It is represented by c.n.

The equation for NBFM is

~~NBFM = + j 2π (fm+fn) + 2π (f~~

NBFM = + A cos [j 2π (fm+fc) + 2π (fm+nfc)]

b)

DSB-FC	DSB-SC	SSB-SC	VSB
* This is the not direct Rx	* This is the not direct Rx	* This is the direct Rx	* This is the direct Rx.
* It is complex	* It is also complex	* It is easy	* It is easy.
* It is the high of the cost	* It is the low cost	* It is the low cost	* It is the low cost
* It is not linear.	* It is linear	* It is also linear.	* It is linear
* It have 2 side bands	* It have 2 side bands	* It is only 1 single band	* It is also one band.

c) $P_t = 1200\,W$

Modulation $(m) = 85\% = 0.85$

$P_c = ?$

$$P_t = P_c\left(1+\frac{m^2}{2}\right)$$

$$1200 = P_c\left(1+\frac{(0.85)^2}{2}\right)$$

$$1200 = P_c(1.36)$$

$$\frac{1200}{1.36} = P_c$$

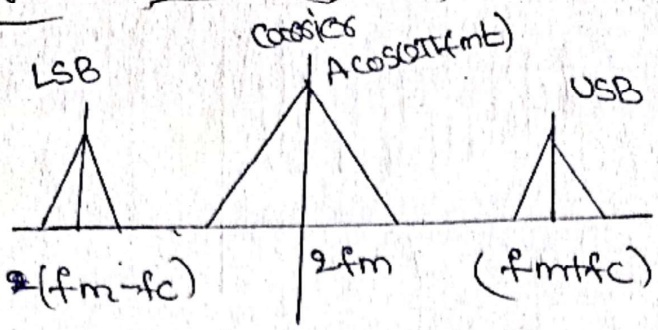
$$P_c = 882.35\ W$$

d) AM :-

The Amplitude modulation is a system in which the modulation of the carrier is proportional to the

Amplitude is called a Amplitude modulation.

frequency spectrum of DSB-FC :-



e) Needs for modulation :-

* Probability of Antenna
* High of Antenna
* Reduce the noise
* Reduction of density
* Disturbances are reduced
* signals are Produces clearly in that.
* we take message signals & carrier signals.

Analog Communication

Assignment 2

1. Explain the working of indirectly modulated FM Transmitter with the help of a neat block diagram. Explain why constant IF is required in super heterodyne receiver. What is image frequency? How it appears in the receiver? CO4
2. Discuss the following terms with regard to receivers. a) Double spotting b) Selectivity c) sensitivity d) Fidelity CO4
3. Determine the nyquist rate for a signal: x (t) = 6 Cos 50πt + 20sin 300πt.Discuss in detail about the following sampling techniques. a) Natural sampling b) flat top sampling. CO1
4. Classify different types of pulse time modulation. Draw the block diagram of typical PWM generator. Define and describe pulse position modulation and explain with waveforms how it is derived from PWM. CO1
5. What are the assumptions to be made in calculating the figure of merit for various communication systems? Define figure of merit of a receiver. When threshold effect starts in an envelope detector? Derive the figure of merit γ of the DSBSC System CO5
6. Explain about the threshold effect in angle modulated system. Prove that the product of the transfer functions of the pre-emphasis and de-emphasis circuits is constant. Derive the expression for figure of merit for FM receiver. CO5

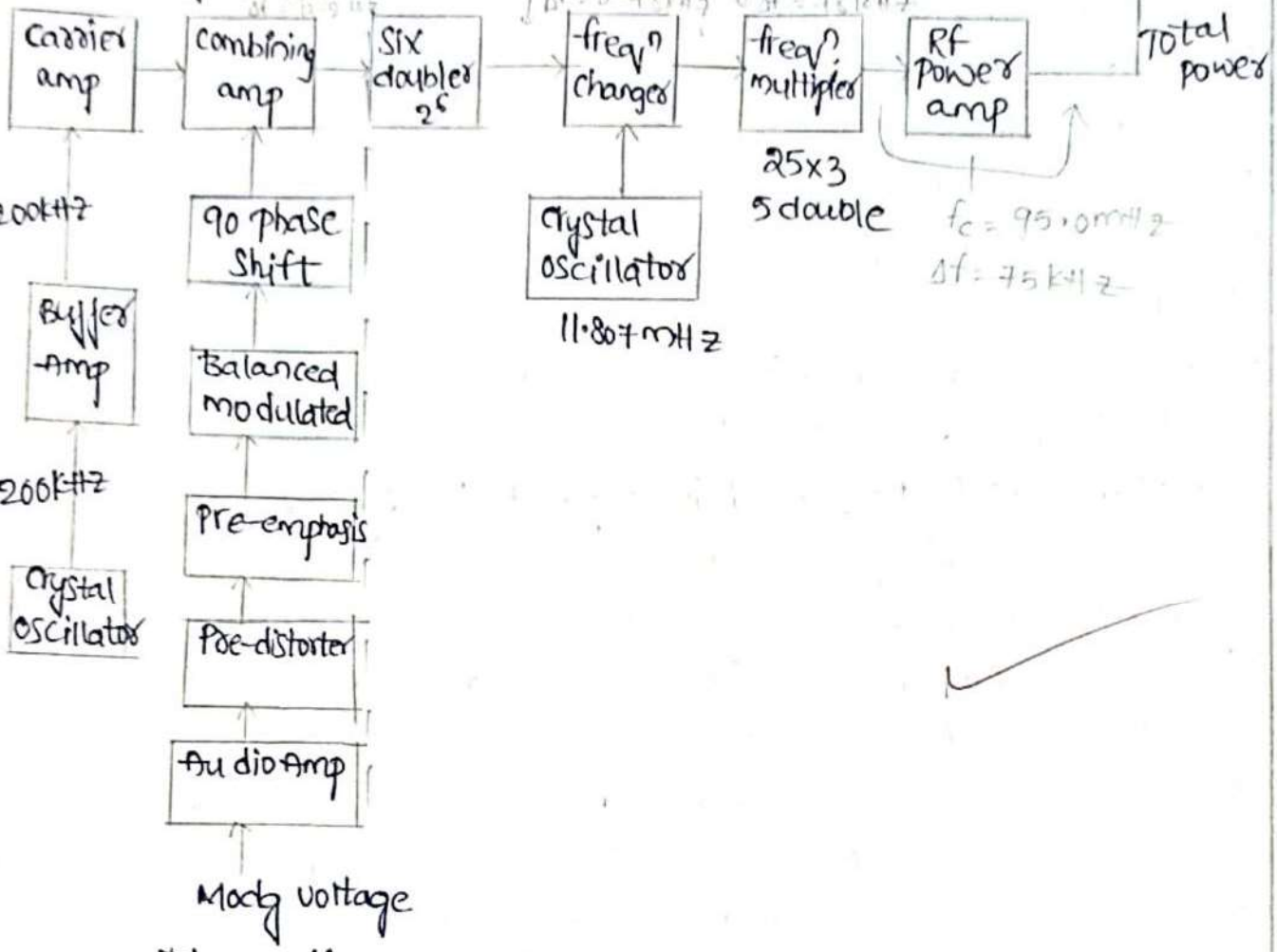
V. Pardhavi
2009A04B6
III. B.Tech ECE-B

Analog communication

Assignment - 2

1. Explain the working of indirectly modulated FM-transmitter with the help of a neat block diagram. Explain why constant IF is required in super heterodyne receiver. What is image frequency? How it appears in the receiver?

Sol Indirectly modulated FM-transmitter:



Block diagram of Indirectly Modulated FM Transmitter.

Working of Indirectly Modulated ~~FM~~ transmitter (Armstrong FM Trans^r)

1. Crystal oscillator is used to provide stable f_c
2. A portion of the carrier voltage is feed to the Supressed carrier balanced modulator
3. The pre-emphasized, pre-distorted & amplified modulating signal is applied to balanced modulated
4. The modulating voltage amplitude is then proportional to $V_m f_m$.
5. Where V_m = original modulated voltage amplitude.

fm = original modulated voltage frequency.

6. The resulting sideband components are shifted with the phase by 90° & are then combined with the amplified carrier voltage in a combining amplifier.

7. The o/p of the combining amp is the desired fm voltage

8. Relative amplitudes of the modulating voltage & carrier voltage are so adjusted that the Max phase deviation is small.

9. This is necessary in order to avoid excessive distortion.

10. As a result the freqn deviation of Δf is small.

11. Here Δf = 12.2Hz at the o/p modulator

12. subsequently freqn multiplier raise fc to desired value of 95.3mHz & Δf to 75kHz

13. After power amplification in a few steps raising the power level to the desired value the o/p is fed to the transmitter antenna.

constant It is Required in super heterdyne receiver.

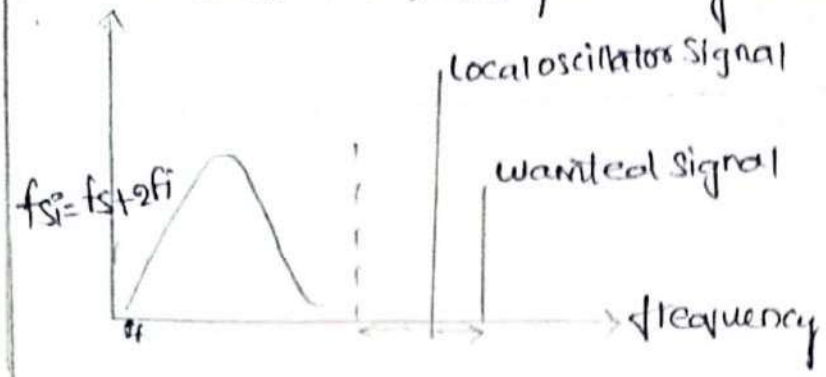
Two different frequencies are to be mixed so as to have another frequency component of lower value Now the thing that first comes to our mind is why the mixer produces a lower frequency value, which is the difference b/w the two frequencies the summation of the carrier & local oscillator frequency at the o/p of mixer will give rise to image frequency which is treated as a type of noise or distortion in the signal this the reason why the mixer generates a frequency difference at its output. This difference frequency is a constant value irrespective of the variations in the i/p known as the 'intermediate frequency'.

Intermediate frequency $f_i = f_o - f_c$

The constant freqn at its output is gained by capacitance tuning. In capacitance tuning, several capacitance are arranged together & operated by a controlling knob. It doesn't Matter what the Incoming signal freqn is the Rf amplifier & local oscillator must be tuned to it.

* In heterodyne receivers, an Image freqn is an undesired i/p freqn equal to the station freqn plus (or minus) twice the intermediate freqn. The image freqn results in two station being received

at the same time, thus producing Interference.



8) Discuss the following terms with regard to receiver
a) Double Spotting b) Selectivity c) Sensitivity d) fidelity

a) Double Spotting:

The phenomenon of double spotting occurs at higher frequencies due to poor front end selectivity of the receivers. In this, reciver picks up same short-wave station at two nearby points on the receiver dial.

When the receiver is tuned across the band, a strong signal appears to be at two different frequencies, once at the desired frequency and again when the receiver is tuned to 2 times. If below the desire frequency. In this second case, the signal becomes the Image reduced in strength by the Image rejection, thus making it appears that the signal is located at two frequencies in the band.

b) Selectivity:

Selectivity refers to the ability of a receivers to a select signal of a desired frequency while reject all other. selectivity in a receiver is obtained by using tuned circuits these are LC-circuits (determines) tuned to resonate at a desired signal frequ the Q of these tuned circuits determines the selectivity. selectivity shows the attenuation that the receiver offers to signal frequencies near to one to which it is tuned. A good receiver isolates the desired signal in the Rf spectrum & eliminate all other sign

$BW = f_0/Q \qquad Q = X_L/R$

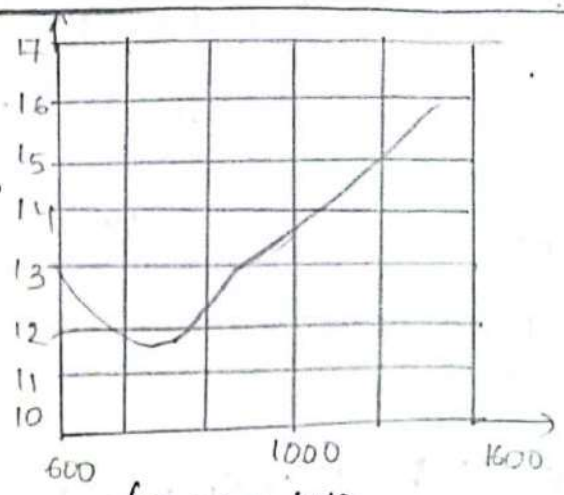
To have narrower bandwidth & better selectivity the Q of the tuned circuit must be high.

Sensitivity:

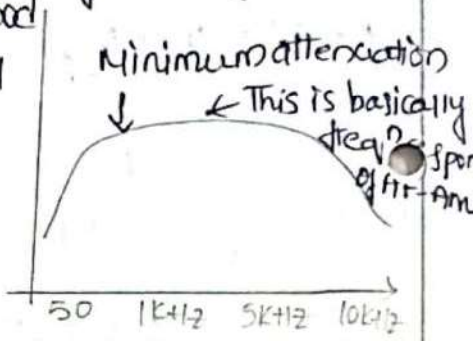
In most simple words we can say it is the ability to amplify a weak or low signal 150μ V is the typical Sensitivity.

value. For small broad cast & curve below is for high quality communication.

Sensitivity curve from good receiver



Fidelity: It is the system's ability to produce on original signal without distortion. It is essential to produce a good quality signal. It ensures that reproduced signal must be an exact copy of the original signal



3. Determine the nyquist rate for a signal $x(t) = 6\cos 50\pi t + 20\sin 300\pi t$. Discuss in detail about the following Sampling a) Natural sampling b) Flat top sampling

Sol) The given signal is expressed as

$$x(t) = 6\cos 50\pi t + 20\sin 300\pi t \longrightarrow (1)$$

Let two frequencies present be ω_1, ω_2

So that the new eqn for signal

$$x(t) = 6\cos\omega_1 t + 20\sin\omega_2 t \longrightarrow (2)$$

Comparing eqn (1) & eqn (2) we have

$\omega_1(t) = 50\pi t$ $\quad\quad$ $\omega_2 t = 300\pi t$

$\omega_1 = 50\pi$ $\quad\quad$ $\omega_2 = 300\pi$

$2\pi f = 50\pi$ $\quad\quad$ $2\pi f = 300\pi$

$f_1 = 25$Hz $\quad\quad$ $f_2 = 150$Hz

Therefore, the minimum frequency present in $x(t) = f_2 = 150$Hz.

Nyquist rate is given as

$$f_s = 2f_m$$

$$f_s = 2\times 150$$

$$= 300\text{Hz}$$

$\therefore$ Nyquist rate $f_s = 300$Hz

Natural Sampling:

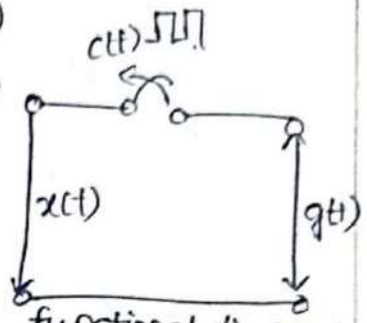
→ Natural Sampling is a practical method

In natural Sampling the pulses has a finite width equal to γ.

→ Let us consider an analog continuous-time signal $x(t)$ to be sampled at the rate of f_s Hz. Here it is assumed that f_s is higher than Nyquist rate such that sampling theorem is satisfied

Again, let us consider a sampling function $c(t)$ which is a train of periodic pulses of width γ & freq^n equal to f_s Hz.

→ When $c(t)$ goes high the switch 's' is closed $g(t) = x(t)$ when $c(t) = A$ & $g(t) = 0$ when $c(t) = 0$

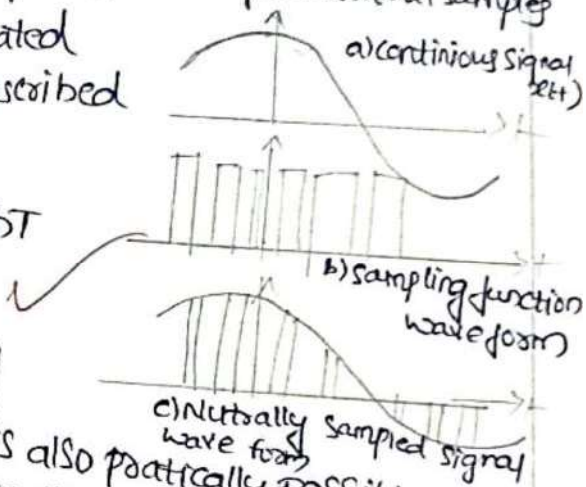


functional diagram of a natural sampler

where A is the amplitude of $x(t)$. The waveforms of signals at $x(t)$, $c(t)$, $g(t)$ have been illustrated

Now, the sampled signal $g(t)$ May also be described mathematically as $g(t) = c(t) \cdot x(t)$

Here, $c(t)$ is the periodic train of pulse width T & freq^n "f_s"

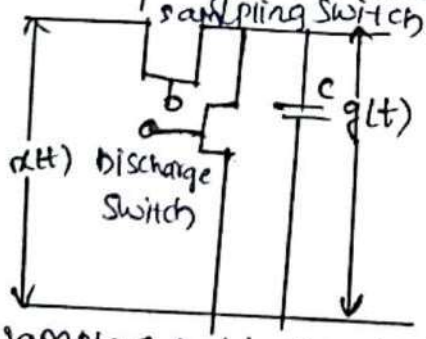


Flat top Sampling or Rectangular Sampling

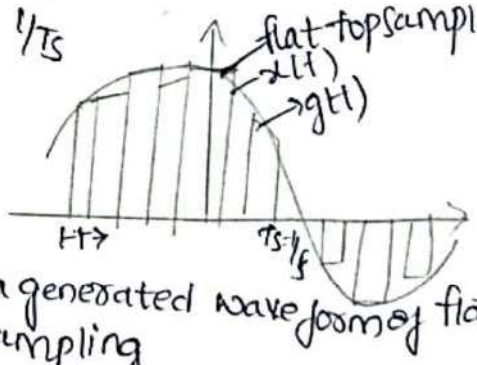
1. Flat top Sampling like natural sampling is also practically possible sampling method. But Natural sampling is little complex whereas it is quite easy to get flat top samples.

In flat-top sampling, the top samples remains constant & is equal to the instantaneous value of the base band signal $x(t)$ at the start of sampling

Sampling rate is equal to $f_s = 1/T_s$



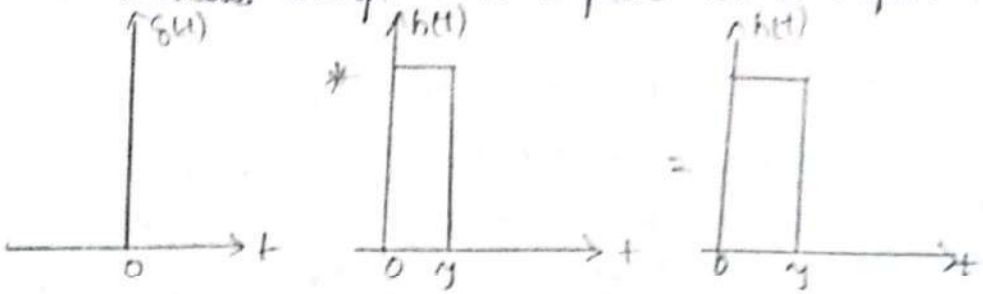
a) sample & hold circuit to generater flattop samples. It may be noted that only starting edge of the pulse represents instantaneous value of the base band signal $x(t)$.



b) a generated wave form of flat top sampling

Flat top pulse $g(t)$ is mathematically equivalent to the convolution

of instantaneous sample and a pulse h(t) is depicted

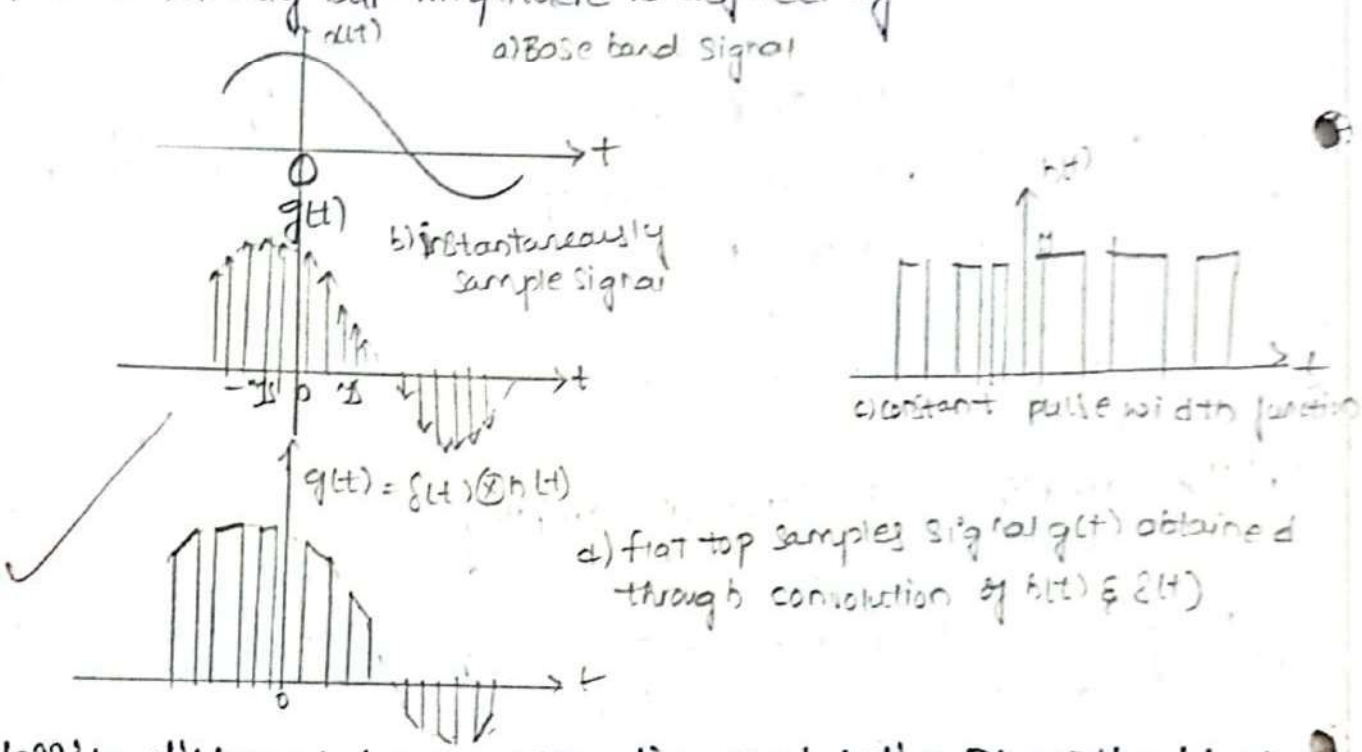


convolution of any function with delta function is equal to that function.

Therefore $g(t) = s(t) \otimes h(t)$

Now, from the property of delta function, for any function $f(t) * \delta(t) = f(t)$

→ ∴ON, convolution of s(t) & h(t) we get a pulse whose duration is equal to h(t) only but amplitude is defined by

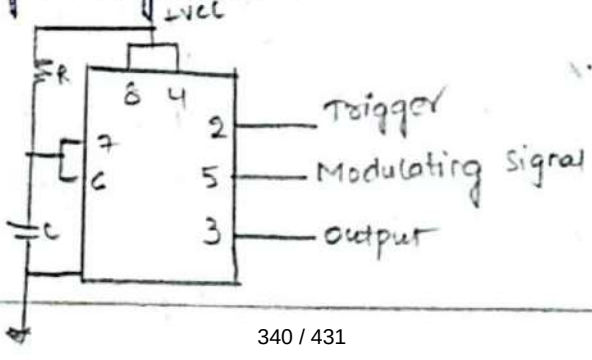


4) classify different types of pulse time modulation. Draw the block diagram of PWM generator. Define & describe pulse position Modulation and explain with waveforms how it is derived from PWM.

Pulse-time Modulation is further divide into analog and digital Modulation. The analog modulation techniques are mainly classified into

1. Pulse-Amplitude modulation 2) Pulse Duration/width Modulation
3. Pulse position Modulation

→ Block diagram of PWM generator



5) What are assumptions to be made in calculating the figure of merit for various communication systems? Define figure of merit of a receiver when threshold effect starts in an envelope detector? Derive the figure of merit of the DSBSC System

Assumption to be made in calculating the figure of merit of various communication systems:-

1. channel noise is additive
2. channel noise is white and gaussian
   total noise power N is obtained simply

$$N = \eta/2 \times \text{Bandwidth}$$

3. Band pass noise at the i/p of detector the power density spectrum of the band pass white noise, power density is given by

$$S_{ni} = \eta/2$$

4. The i/p noise power N_i

$$N_i = \eta/2 \times 2f_m = \eta f_m$$

figure of merit (γ): It is defined as the ratio of output signal noise ratio to input signal ratio

$$\text{figure of merit}(\gamma) = \frac{(SNR)_o}{(SNR)_c}$$

Higher the value of figure of merit indicates better the noise performance of the receiver

Threshold effect in the envelope detector

We have seen that when a noise is large as compared to the signal at i/p of the envelope detector, the detected o/p has a msg signal completely mingled with the noise. It means if the (S/N_i) is below a contain level called threshold level the noise dominate over the message signals. The threshold effect starts in an envelope detector whenever carrier power to noise power ratio approaches unity or level

Threshold effect is the property of an envelope detector or

figure of merit of the DSB-SC system:-

DSB-SC : The DSB-SC system uses synchronous detector at the receiver i/p noise power (N_i) is given as, $N_i = \eta/2 \times 2f_m$

$$N_i = \eta f_m$$

The noise signal at the i/p of this detector is bandpass noise is given by

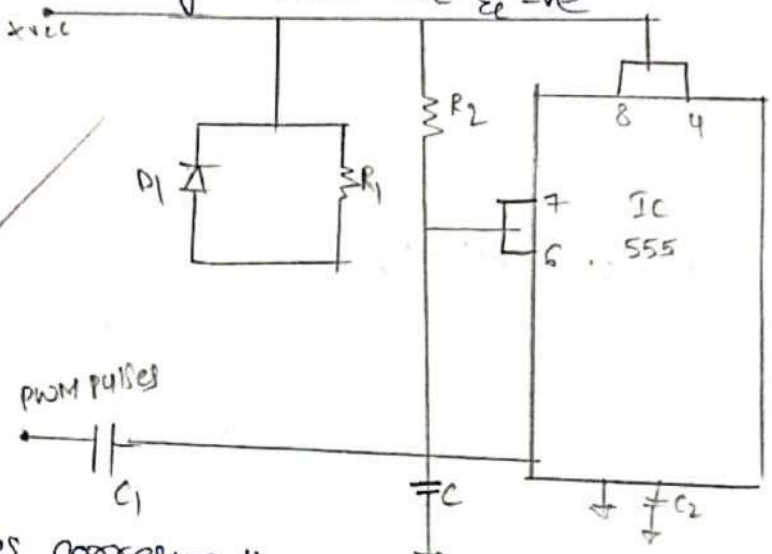
$$n(t) = n_c(t)\cos\omega_c t - n_s(t)\sin\omega_c t$$

Pulse Position Modulation

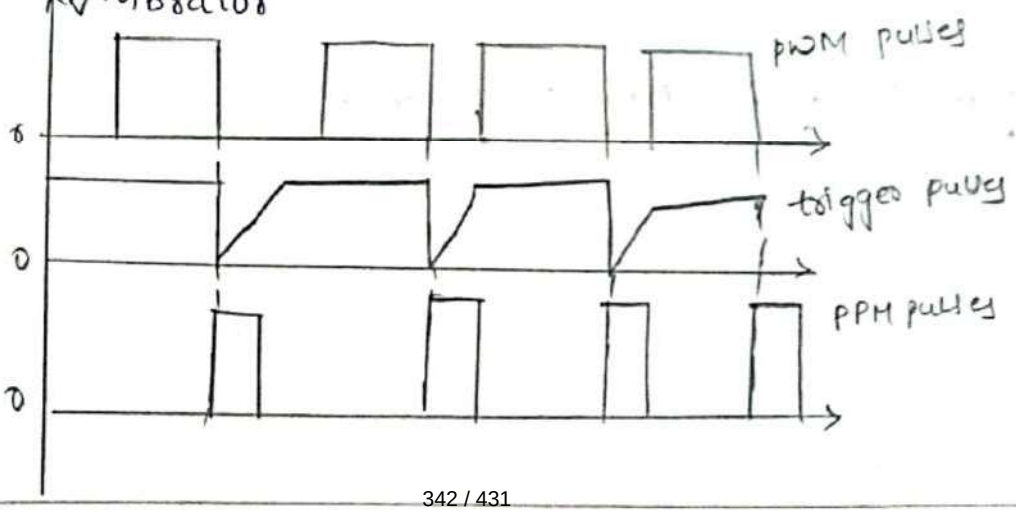
In this System the Amplitude & width of pulses are kept constant, while the position of each pulse, with reference to the position of a reference pulse is changed according to the instantaneous sampled value of the modulating signal. Thus the transmitter has to send Synchronising pulses to keep the transmitter & receiver Synchronism. As the Amplitude and width of the pulses are constant. A transmitter handles constant power output, definite advantage over the PWM. But the disadvantage of the PPM System is the need for transmitter-receiver synchronization. pulse position modulation is obtained from Pulse width modulation. Each trailing edge of PWM pulse is a starting point of the pulse in PPM. Therefore, position of the pulse is 1:1 proportional to the width of pulse in PWM & hence it is proportional to the instantaneous Amplitude of the Sampled modulating Signal

Generation of PPM Signal

PPM generator. It consist of differentiator and a monostable multivibrator. The input to the differentiator in two waveform form. The differentiator generator +ve & -ve



Spikes corresponding to leading & trailing edges of the PWM waveform diode D_1 is used to by Pass the +ve spikes. The -ve spikes are used to trigger monostable multivibrator



This signal is multiplied by $\cos\omega_c t$ in the synchronous detector the multiplied signal

$$n_d(t) = n(t)\cos\omega_c t$$

$$= \frac{n(t)}{2}\cdot(1+\cos 2\omega_c t) - \frac{1}{2}n_s(t)\sin 2\omega_c t$$

$$n_d(t) = \frac{1}{2}n_c(t) + \frac{1}{2}n_c(t)\cos 2\omega_c t - \frac{1}{2}n_s(t)\sin 2\omega_c t$$

→ the signal $n_d(t)$ is passed through LPF. The noise power $n_0(t)$ appearing at the o/p of the synchronous detector is only $\frac{1}{2}n_c(t)$

$$n_0 = \frac{1}{2}n_c(t)$$

∴ The power density spectrum is proportional to the mean square value of this signal. The power density spectrum of $n_0(t)$ related to the power density spectrum of $n(t)$ as follows

$$S_{no}(\omega) = \frac{1}{4}S_{nc}(\omega)$$

for a DSBSC system $S_{nc}(\omega) = \eta_1 \quad |f| \leq f_m$

$$S_{no}(\omega) = \frac{1}{4} \quad \eta = \eta/4$$

The B.W of the message signal at the o/p of the detector is $2f_m$

The o/p noise power n_0 is given by

$$N_0 = S_{no}(\omega) \times B.W$$

$$= \eta/4 \times 2f_m = \frac{\eta f_m}{2}$$

$$\therefore \frac{N_0}{N_i} = \frac{\eta f_m}{2}\cdot\frac{1}{\eta f_m} = 1/2$$

→ the i/p signal power is the mean square value of $f(t)\cos\omega_c t$

the signal voltage at the voltage of the o/p of the detector is $\frac{1}{2}f(t)$

∴ the o/p signal power is the rms value of $\frac{1}{2}f(t)$

$$S_0 = \overline{(\tfrac{1}{2}f(t))^2} = \frac{1}{4}\overline{f^2(t)}$$

$$\frac{S_0}{S_i} = \frac{\overline{f^2(t)}}{4}\cdot\frac{2}{\overline{f^2(t)}} = 1/2$$

∴ Figure of merit for a DSB-SC system is unity

∴ the SNR at the i/p & o/p are identical & there is no improvement in SNR

4) Explain about the threshold effect in angle modulated system prove that the product of the transfer functions of the pre-emphasis & de-emphasis circuit is constant. Derive the expression for figure of merit for FM receiver.

Ans **FM threshold effect:**

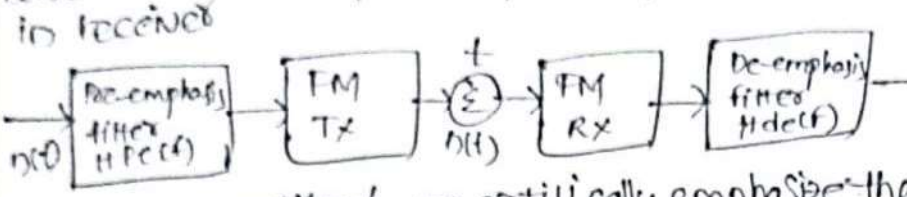
The $(S/N)_0$ of an FM signal is indicated by

$(S/N)_0 = \frac{3A_c^2 K_f^2 P}{2N_0\omega^3}$ is valid only if (S/N) measured at discriminator

S/P is high compared with unity
→ It is observed that the i/p noise power, the (S/N) ratio decreases & Rx breaks.
→ Initially individual clicks are heard in Rx o/p & (S/N) still further, click rapidly merge into a crackling or sputtering noise.
→ At breaking point (S/N) begins to fall by predicting values of o/p (S/N) larger than the actual ones this phenomena is known as threshold effect

Pre-emphasis and De-emphasis:

1. In this method simple RC network are used to improve the threshold
2. A more satisfactory approach to efficient utilization of allowed freq is based on the use of pre-emphasis of the transmitter & de-emphasis in receiver



3. In this method, we artificially emphasise the high frequency components of message signal prior to modulation in the Tx, & therefore before the noise is introduced in the Rx

Improvement FM analysis w.r.t avg o/p noise power without pre-emphasis & de-emphasis

$$= 2N_0W^3/3A_c^2$$

$$D = \frac{\text{avg o/p noise power without pre-emp \& de-emp}}{\text{avg o/p noise power with pre-emp \& de-emp}}$$

$$D = \frac{2N_0W^3/3A_c^2}{\frac{N_0}{A_c^2}\int_{-W}^{W} f^2|H_{de}(f)^2|df} = \frac{3N_0W^3}{3A_c^2} \times \frac{A_c^2}{N_0\int_{-W}^{W} f^2|H_{de}(f)|df}$$

Derivation for figure of merit for FM reciever

$$S(t) = A_c\cos(2\pi f_c t + 2\pi k_f \int m(t)dt)$$

$$S_i = A_c^2/2 \quad ; \quad m = WN_0$$

$$(S/N)_i = \frac{A_c^2}{2W_0N_0}$$

I/P to FM demod = FM signal + Noise

$$= A_c\cos(2\pi f_c t + 2\pi k_f \int m(t)dt) + n_c(t)\cos 2\pi f_c t + n_q(t)\sin 2\pi f_c t$$

Assume that carrier is not modulated $m(t) = 0$

$$= A_c\cos 2\pi f_c t + n(t)\cos 2\pi f_c t + n_q(t)\sin 2\pi f_c t$$

$$= A_c + n_c(t)\cos 2\pi f_c t + n_q(t)\sin 2\pi f_c t$$

$$= \sqrt{(A_c + n_c(t))^2 + n_q(t)}\cos(2\pi f_c t + \phi) \rightarrow (1)$$

→ In order to produce undistorted version of original message at the Rx o/p, pre-emphasis filter in Tx & de-emphasis filter in Rx must ideally have transfer function that are the inverse of each other

$H_{pe}(f)$ – TF of pre-emphasis

$H_{de}(f)$ – TF of de-emphasis

$H_{de}(f) = \frac{1}{H_{pe}(f)}$ $-w \le f \le w$

This choice of TF makes the avg msg power of Rx o/p independent of pre-emphasis & de-emphasis process

we have power spectral density of noise of discriminator o/p is

$$S_{Nd}(f) = \frac{N_0 f^2}{A_c^2}$$

∴ Modified power spectral density of the noise at the de-emphasis filter o/p

$$= |H_{de}(f)|^2 S_{Nd}(f)$$

∴ Avg o/p noise power with de-emphasis $= \frac{N_0}{A_c^2} \int_{-w}^{w} f^2 |H_{de}(f)|^2 df$

$\phi = \tan^{-1}[n_q(t)/(A_c + n_c(t))]$

$S(t) = A_c \cos\theta(t)$

$\theta(t) = 2\pi f_c t + \phi$

$\phi \simeq \tan^{-1}(n_q(t)/A_c)$ neglecting $n_c(t)$

$\theta = \phi = \frac{n_q(t)}{A_c}$ So, $\tan^{-1}\left(\frac{n_q(t)}{A_c}\right) < 1$

Substitute ϕ in ①

$$= \sqrt{A_c + (n_c(t))^2 + n_q^2(t)}$$

$\cos\left(2\pi f_c - \frac{n_q(t)}{A_c}\right) = \theta(t)$

$f_i = \frac{1}{2\pi} \frac{d\theta}{dt}$

$f_i = \frac{1}{2\pi}\left[\frac{d}{dt}\left[2\pi f_c t - \frac{n_q(t)}{A_c}\right]\right]$

$= \frac{1}{2\pi}\left[2\pi f_c - \frac{1}{A_c}\frac{d}{dt} n_q(t)\right]$

$f_i = f_c - \frac{1}{2\pi A_c}\frac{d}{dt} n_q(t)$

$f_i = f_c + k f m(t) - \frac{1}{2\pi A_c}\frac{d}{dt} n_q(t)$

f_i due to f_m – AM signal; f_i' due to noise signal

o/p of demodulator $V_0 = k f_i$

$V_0 = k f_c + k k_f m(t) - \frac{K}{2\pi A_c}\frac{d}{dt} n_q(t)$

$= 0$ when $(f_i = f_c)$

$V_0 = K \cdot k_f \cdot m(t) - \frac{K}{2\pi f_c}\frac{d}{dt} n_q(t)$

$K \cdot k_f \cdot m(t) \rightarrow K^2 k_f^2 P$

$S_0 = K^2 k_f^2 P$

$n_q(t) \xrightarrow{PSD} N_0$; $-w < f_c < w$

$\frac{K}{2\pi A_c}\frac{d}{dt} n_q(t) \rightarrow \frac{K}{2\pi k_c} j w N_0$

$= \frac{K^2}{4\pi^2 A_c^2}(2\pi f)^2 N_0$

$= \frac{K^2 f^2 N_0}{A_c^2}$

$PSD = \frac{K^2 f^2 N_0}{A_c^2}$

Noise power $= \int_{-w}^{w} \frac{K^2 N_0}{A_c^2} f^2 df$

$= \frac{K^2 N_0}{A_c^2} \int_{-w}^{w} f^2 df$

$= \frac{K^2 N_0}{A_c^2} \left(\frac{f^3}{3}\right)$

$= \frac{K^2 N_0}{A_c^2} \left[w^2 - (-w^3)\right]$

$n_0 = \frac{K^2 N_0}{3A_c^2} (2w^3)$

$(S/N)_0 = K^2 k_f^2 \cdot P 3A_c^2 / K^2 N_0 \cdot 2w^3$

$(S/N)_0 = \frac{\frac{3}{2} k_f^2 P A_c^2}{N_0 w^3}$ ...

$f_0 m(t) = \frac{3/2\, k_f^2 P A_c^2 / N_0 w^3}{A_c^2 / 2wN_0}$

$f_{om} = \frac{3/2 k_f^2 P}{w^2}$

$= \frac{3 k_f^2 A_m^2}{2 f_m^2}$

for single tone mod of FM

$f_{om}(t) = \frac{3}{2} \beta^2$

(10)

DRS

R-20

College Code: 09

Rajeev Gandhi Memorial College of Engineering & Technology

Autonomous

NANDYAL-518501

III B.Tech I-Semester- Mid-II *Examinations*

Subject Name: Analog communication

Branch: ECE

Max. Marks: 20 **Date: 08-12-2022** **Time: 2 Hours**

Note: 1.Answer first question compulsorily. (5 x 1= 5Marks)

2. Answer Any *THREE* from 2 to 5 questions. (3 x 5 = 15 Marks)

Q.No			M	CO	BL
Q.1	a)	Write the expressions of in phase and quadrature phase noise components.	**1M**	CO5	BL1
	b)	What is blocking in a receiver ?	**1M**	CO4	BL1
	c)	Determine the nyquist rate for a signal: x(t) = $(1/2\pi)$Cos (4000πt) cos (1000πt)	**1M**	CO1	BL5
	d)	Differentiate between PAM and PWM.	**1M**	CO1	BL3
	e)	Define figure of merit of a receiver.	**1M**	CO5	BL1
Q.2	a)	Explain the working of indirectly modulated FM Transmitter with the help of a neat block diagram.	**3M**	CO4	BL2
	b)	Discuss the following terms with regard to receivers. i) selectivity ii) sensitivity	**2M**	CO4	BL6
Q.3	a)	Draw the block diagram of tuned radio frequency Receiver and explain the function of each block.	**3M**	CO4	BL2
	b)	In a broadcast super heterodyne receiver having no RF amplifier, the loaded Q of the antenna coupling circuit is 100. The IF is 455 KHz. Determine the image frequency and its rejection ratio for tuning at 25 MHz.	**2M**	CO4	BL5
Q.4	a)	With a neat block diagram, explain the operation of pulse position modulation.	**3M**	CO1	BL2

	b)	Derive the transmission bandwidth in PAM.	**2M**	CO1	BL3
Q.5	**a)**	Derive the figure of merit γ of the DSBFC System.	**3M**	CO5	BL3
	b)	What is capture effect ?	**2M**	CO5	BL1

S.No. 42764

RAJEEV GANDHI MEMORIAL COLLEGE OF ENGINEERING & TECHNOLOGY

AUTONOMOUS

Accredited by NAAC of UGC, New Delhi with 'A' Grade:: Accredited by NBA
Affiliated to JNT University Anantapur
Nandyal - 518 501, Kurnool Dist, A.P.

Principal
Autonomous
RGMCET, NANDYAL.

INTERNAL EXAMINATIONS ANSWER BOOKLET

NAME OF THE STUDENT: S Naveed ahammed Reg. No. 20091A04A5

	1	2	3	4	5
A	1	3	3	2.5	0
B	0	1.5	1	2	
C	0				
D	1				
E	1				
Total	3	4.5	4	4.5	0

Grand Total :(In Figures) 16

(in Words): one six

NAME OF THE SUBJECT: AC

INTERNAL EXAM : I / II

Date of Exam: 08/12/2022 (FN/AN)

Course : B.Tech. / M.Tech./ MBA / MCA

Year : III Sem.: I

Branch: ECE

M. Naveed 8/12/22
Signature of the Invigilator

(Start Writing From Here)

Q.2. Working of indirectly modulated FM Transmitter

2.a) → It is also called Armstrong FM Transmitter

→ The block diagram of indirectly modulated FM is given as

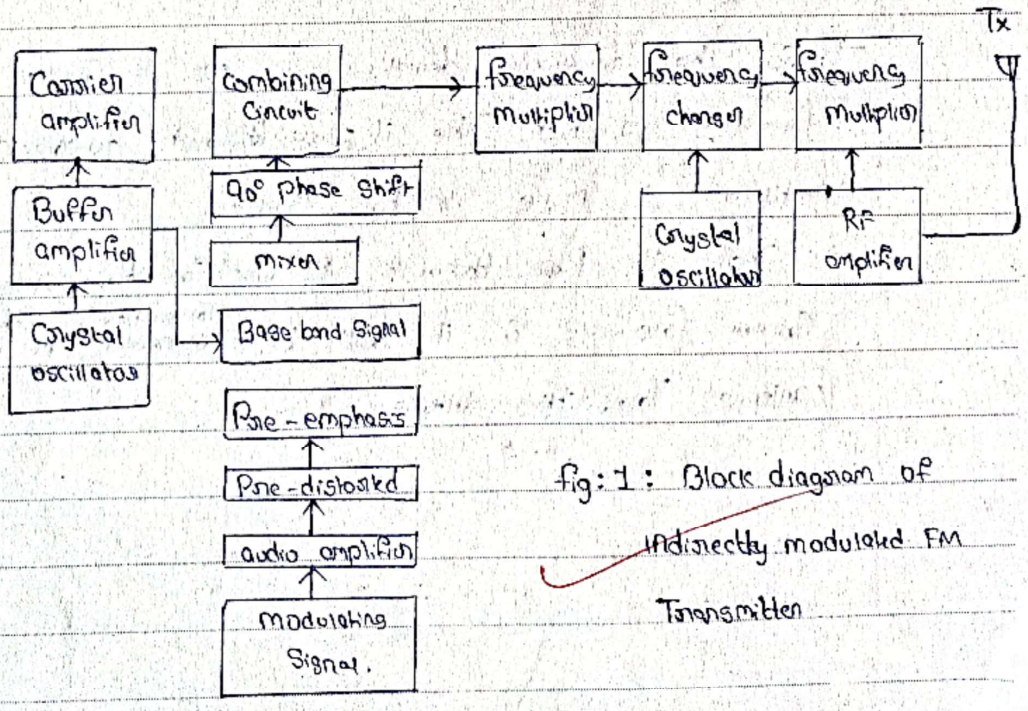


fig: 1: Block diagram of indirectly modulated FM Transmitter

→ The Crystal oscillator produces oscillations to the Carrier amplifier and through that a buffer amplifier is placed to not to come variations in frequency.

→ Then a Carrier signal is generated through Carrier amplifier.

→ Then the pre-emphasis and pre-distorted and audio amplifier is fed to the baseband eg. through modulating signal

→ Then the modulating signal is to do 90° phase shift and then goes to mixer.

→ The Carrier signal & modulating signal are combining in combinational circuit.

→ By that it become to know the modulating frequency and Carrier frequency are less in power along with frequency deviation (Δf).

→ Then it can be pass through frequency multiplier to increase the frequency then it passes through frequency chanser & it already provide with Crystal oscillator.

→ So the frequency deviation is less & Carrier frequency is high. The it pass through frequency mutiplier & make Δf to 75KHz.

→ Then it Pass through RF amplifier to increase the Carrier frequency & it will be Transmitted through (antenna) Transmitting antenna

2.b)

i) Selectivity :-

Selectivity is defined as to select the station among the all station of Am bandwidth is called Selectivity

ii) Sensitivity :-

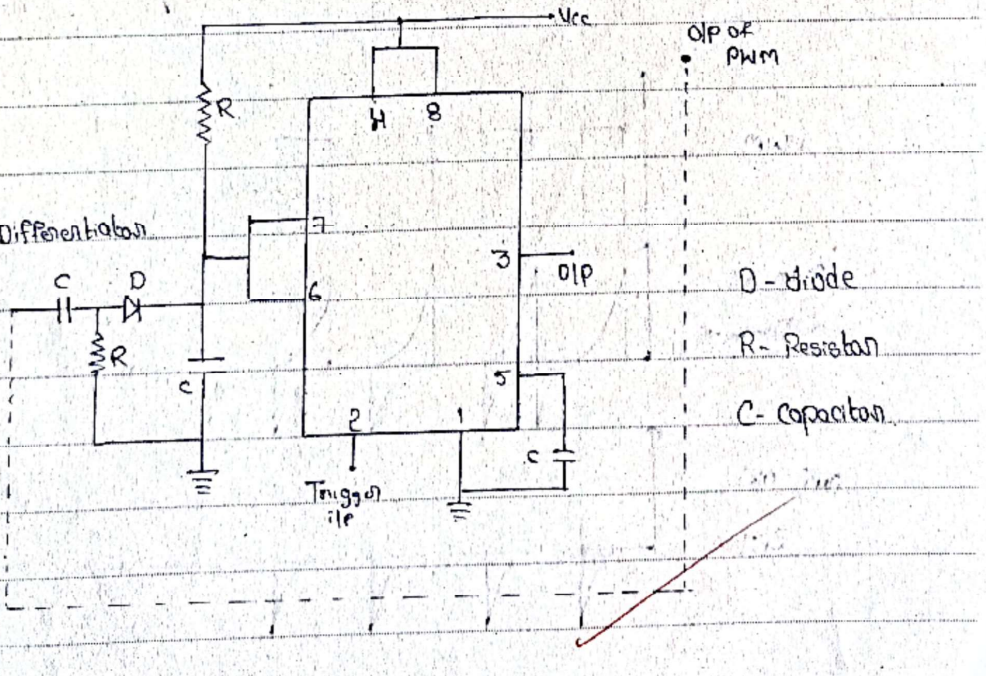
Sensitivity is defined as to take the weak signal & make it as strong signal is called sensitivity

- It is used to control the gain also.

Q.4. Pulse Position Modulation :-

a) The Position of the pulses in the Carrier Pulse train is varied in accordance of modulating signal is called Pulse Position Modulation (PPM).

Operation of PPM :



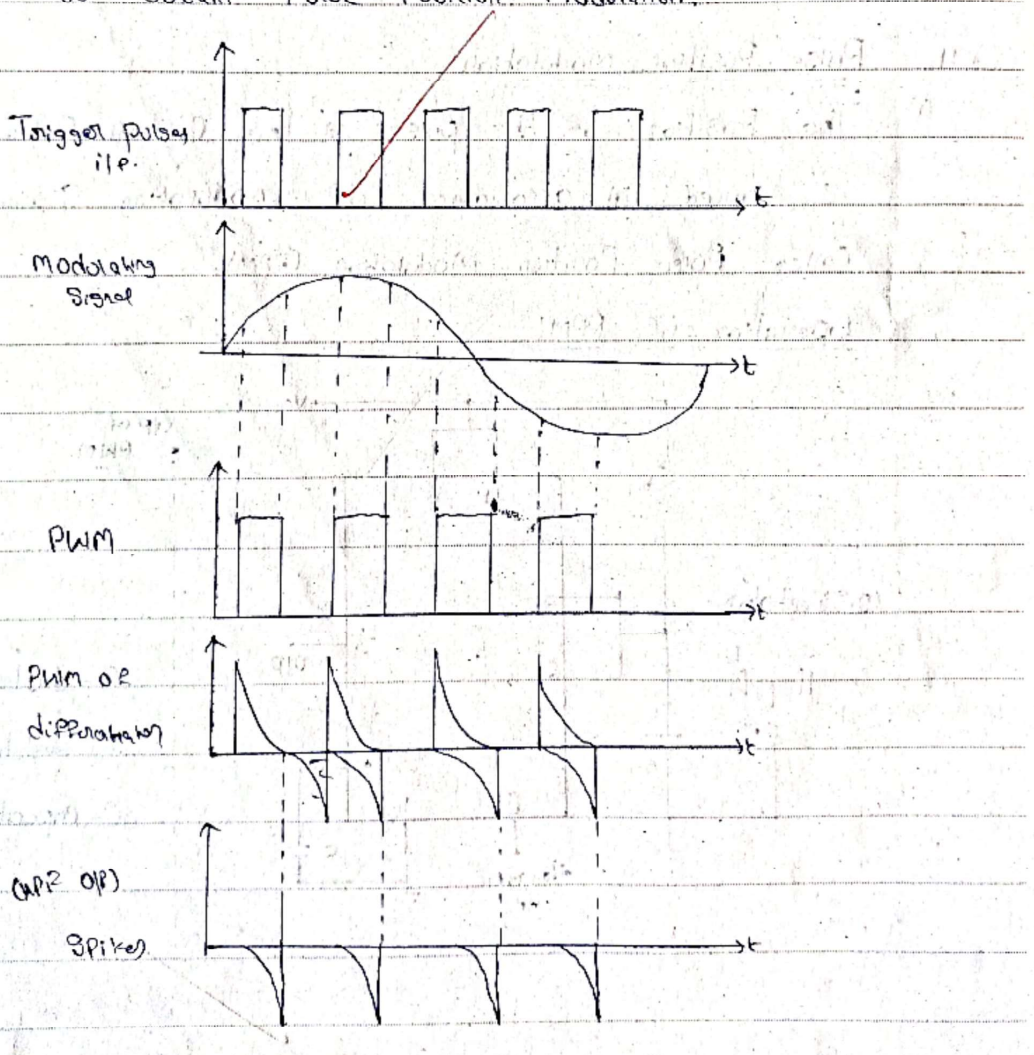
→ PPM is obtained by PWM & monostable multivibrator.

→ The output of PWM is connected as differentiator to the another monostable multivibrator to obtain Pulse Position modulation.

→ First we take a trigger pulses as input & it is multiplied with modulating signals to generate Pulse width modulation

→ Then the sampled pulses of PWM is again goes to pass through differentiator & it can give spikes as output.

→ Then the spikes are again with change in operation & pass through (low) pass filter, (High) HPF to obtain Pulse Position Modulation.



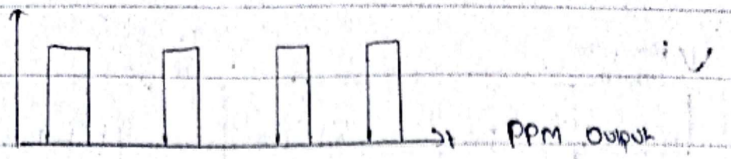


fig (2): Output waveforms of generation of PPM

2.

4.b) Transmission bandwidth of PAM:

→ The transmission bandwidth of Pulse Amplitude modulation is given as:

2

We know that the frequency spectrum f_s

$$f_s = 1/T_s$$

Since $f_s \gg 1/T_s$

We know $T_s \gg \tau$

$$f_s \ll \frac{1}{\tau}$$

Since $f_s = 2f_m$

$$2f_m \gg \frac{1}{\tau}$$

$$f_m \ll \frac{1}{2\tau}$$

Now, the max frequency is given as

$$f_{max} \ll \frac{1}{2\tau}$$

Since Bandwidth BW

$$BW \ll \frac{1}{2\tau}$$

$$\frac{1}{2\tau} = f_m$$

$$BW \gg f_m$$

∴ Bandwidth $BW \gg f_m$ or f_s.

Q.3.

3.a) Tuned Radio frequency Receiver:-

→ It is one type of a Receiver.

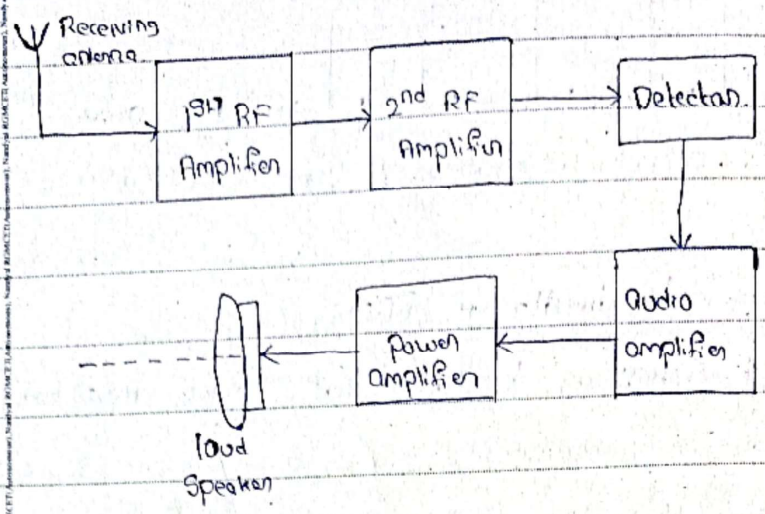


fig 3.1:- Block diagram of Tuned Radio frequency Receiver.

Receiving Antenna:- It is the antenna which receives the signal what we want

1st RF amplifier : RF amplifier means Radio frequency amplifier. In this amplifier, it receives (or) takes the signal & increases the power levels.

2nd RF amplifier :- It also same, it signal pass through second RF amplifier to increase power level of the signal

3

Detector:- Detector is a device which detects the signal. In this, it can take out the unwanted signal & detect the noise in signal

Audio & power amplifier :-

These audio & power amplifiers are the amplifiers when the signal detects noise & remove then its level of power be less & it again pass through this amplifier to increase signal strength.

loud speaker:- It is a device which receive & gives the signal. In this it gives out the received signal

3.b) Given IF = 455KHz

the Quality factor,

Q = 100

& given fi = 25MHz

Now, to find image frequency, we find ρ

ρ is given as $\frac{f_{si}}{f_s} + \frac{f_s}{f_{si}}$

$f_{si} = f_s + 2f_i$

$= 455 + 50$

$= 505$

$\rho = 2.010$

Now image frequency $= \sqrt{1+Q^2\rho^2}$

$= \sqrt{1+(100)^2(2.010)^2}$

$= \sqrt{1+40401} = \sqrt{40402}$

$= 201.002$

$= 201\ KHz.$

Q.1.

1.a) The expressions of inphase and quadrature phase noise components are

$n_{ich} = n_c \cos\omega t$

$n_{qch} = n_c \cos\omega t - n_c \sin\omega t$

1.b) blocking in a receiver:

Blocking in a receiver means, it is detector it blocks the noise & unwanted signal & gives only wanted signal. It detects only block noise.

1.c) Given $x(t) = \frac{1}{2\pi} \cos 4000\pi t \cdot \cos(1000\pi t)$

On comparing with eqn

$$f_1 = \frac{4000}{2} = 2000$$

$$f_2 = \frac{1000}{2} = 500$$

Now 2000 is greater than, then Nyquist rate

$$= 2f_m$$

$$= 2 \times 2000$$

$$= 4000.$$

1.d)

PAM	PWM
→ PAM means pulse amplitude Modulation	→ PWM means Pulse width Modulation.
→ In this, the amplitude (Pulses) is varied in accordance of modulating Signal	→ In this, the width of Pulses is varied in accordance of Modulating Signal
→ It is obtained by Flat top Sampling	→ It is obtained by Sending Modulating Signal in Control voltage of Monostable Multivibrator.

1.e) Figure of Merit: It is defined as the ratio of output SNR to channel SNR

it is given as (γ) - gamma

$$\text{figure of merit } (\gamma) = \frac{(SNR)_o}{(SNR)_c} \text{ (or) } \frac{So/No}{Si/Ni}$$

CHOICE

Q.5. a) figure of merit γ of DSBFC System:

→ We know that input Signal Noise power $N_i = \eta f_m$ (from noise in AM)

Now, the noise equations is given as

$$n_c(t) = n_c \cos \omega_c t$$

$$n_s(t) = n_c \cos \omega_c t - n_s \sin \omega_c t$$

Now, $n_s(t)$ is product with $n(t)$ to get $n_o(t)$

$$n_o(t) = n_s(t) \times n_c(t)$$

$$n_o(t) = (n_c \cos \omega_c t - n_s \sin \omega_c t)\, n_c \cos \omega_c t$$

$$= n_c \cos^2 \omega_c t - n_s \cos \omega_c t \sin \omega_c t$$

$$= n_c \left(\frac{1+\cos 2\omega_c t}{2}\right) - n_c \frac{\sin 2\omega_c t}{2}$$

$$= \frac{1}{2} n_c + \frac{1}{2} n_c \cos 2\omega_c t - \frac{1}{2} n_c \sin \omega_c t.$$

Now $S_{No}(\omega) = \frac{1}{4} \times \eta$

$$N_o = \tfrac{1}{4} \times \eta \times B\omega \Rightarrow \frac{\eta}{4} \times 2 f_m = \frac{\eta}{2} f_m.$$

Now, Signal Power $S_i = \tfrac{1}{2} \overline{f^2(t)}$ [From mean square value of $\cos \omega_c t$]

$$S_o = \tfrac{1}{4} \overline{f^2(t)}$$

Now, figure of merit:

$$\gamma = \frac{S_o/N_o}{S_i/N_i}$$

$$\gamma = \frac{\frac{1}{4}\overline{f^2(t)}}{\eta/2 f_m} \times \frac{\eta f_m}{\frac{1}{2}\overline{f^2(t)}}$$

$$\gamma = \frac{1/4}{1/2} \times 1/2$$

$$\gamma = 1$$

∴ figure of merit of DSBFC System is unity (1).

Code: A0413205R1222

RGM COLLEGE OF ENGINEERING & TECHNOLOGY (AUTONOMOUS)

22nd December 2022

III B.Tech. I Sem. (R20) End Examinations (Regular)

ANALOG COMMUNICATION

ECE

Time: 3 Hrs **Total Marks: 70**

Note 1:Answer Question No.1 (Compulsory) and 4 from the remaining

2:All Questions Carry Equal Marks

1a Draw the frequency spectrum of VSB, where it is used.

b What are the functions performed by the carrier in radio communications?

c What is Threshold effect in an envelope detector?

d What are the disadvantages of PAM?

e Explain why constant IF is required in super heterodyne receiver.

f What is meant by indirect FM generation?

g Draw the circuit diagram of an envelope detector.

2 a) Describe the general process of frequency changing in a super heterodyne Receiver. (9)

b) List out the functions of RF amplifier in the receiver. (5)

3 Draw and explain about

a) PPM modulation and

b) PPM demodulation

4 a) What is pre-emphasis? Why is it used? Sketch the circuit of pre-emphasis. Explain why de-emphasis must also be used. (9)

b) A modulating signal $5\cos(2\pi 15\times 10^3)t$, angle modulates a carrier $A\cos\omega_c t$. (5)

i. Find the modulation index &bandwidth for both FM &PM.

ii. Determine the change in the bandwidth and the modulation index for both FM &PM, if modulating signal frequency reduces to 5kHz.

5 a) With the help of typical values explain about the various RF bands and state their applications. (10)

b) Explain the Communication Process. (4)

6 a) Prove that the Product of the transfer functions of the pre-emphasis and De-emphasis circuits is Constant.

b) Define and explain the Capture effect.

7 a) Explain the generation of AM signals using square law modulator. (9)

b) Derive the Power relations in AM technique. (5)

- xxx -

1/11

RGM College of engineering & Technology (AUTONOMOUS)

22nd December 2022

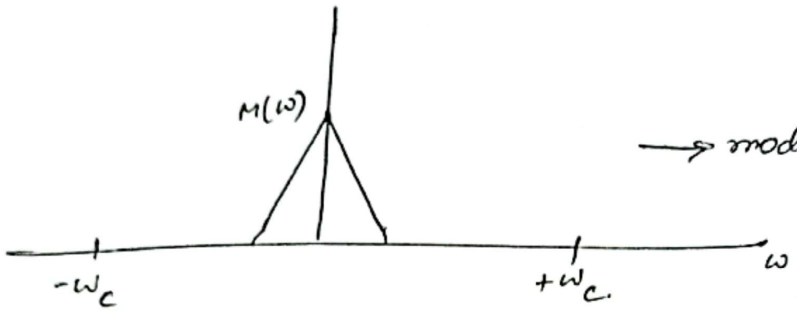
III B.Tech. I sem. (R20) End Examinations (Regular)

ANALOG COMMUNICATION Code: A04137205R12222

ECE

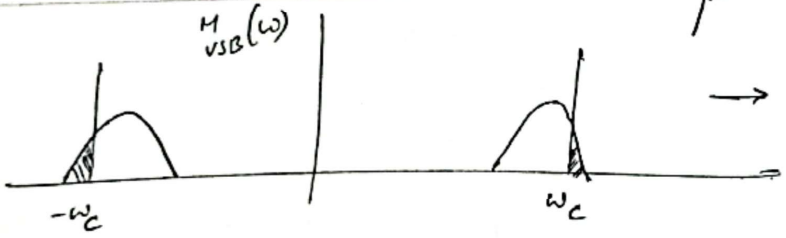
Scheme of evaluation.

(1) (a) Frequency spectrum of VSB (Vestigial side band) :-



→ modulating signal.

(1 Mark).



→ frequency spectrum of VSB.

VSB modulation is used for the transmission of TV signals because the video signals require the larger transmission bandwidth, if transmitted using VSB-FC (or VSB-SC techniques ——— (1 mark).

(1) (b) The purpose of the carrier is usually either to transmit the information through space as an electromagnetic wave (as in radio communication), or to

to allow several carriers at different frequencies to share a common physical transmission medium by frequency division multiplexing (as in a cable television system.

- Carrier does not contain any information, it consists of constant parameters like amplitude, phase and frequency. ——— (2 marks).

(1)(c) The loss of message signal in the output of the envelope detector, due to low carrier-to-noise ratio is called as threshold effect.

- It means that there is a massive drop in output SNR, if input SNR drops below threshold.
- At breaking point $(S/N)_o$ begins to fall by predicting values of o/p (S/N) larger than the actual ones. This phenomenon is known as threshold effect. ——— (2M)

(1)(d) Disadvantages of PAM:-

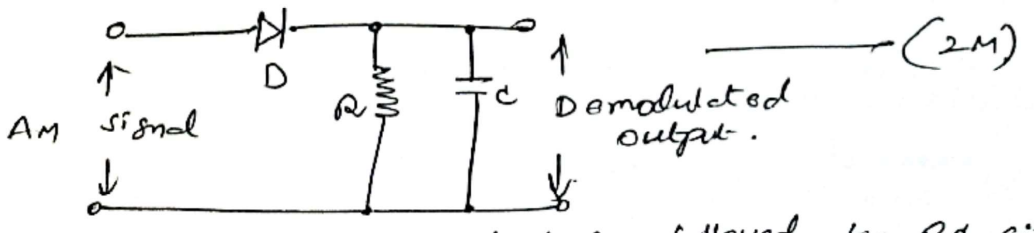
- Large bandwidth is required for transmission.
- More noise. ——— (2M).
- Transmitted power is not constant.
- Since PAM (pulse amplitude modulation) does not utilize constant amplitude pulse, output is distorted due to additive noise so that it is infrequently used.

2-/11

(1e) The main reason for using an intermediate frequency is to improve frequency selectivity. IF is a fixed RF frequency. The receiver after the IF section has to deal with only one carrier frequency of IF. All the different received station carrier frequencies are down converted with this single IF carrier in which the receiver can demodulate easily and present it to the output. ——— (2M)

(1) (f) In the indirect method of producing frequency modulation, the modulating signal is first integrated and then phase modulates a crystal controlled oscillator signal to produce a narrow band FM signal. Frequency multiplication is then used to increase the resulting frequency deviation to the desired level. ——— (2M)

(1) (g) Circuit diagram of envelop detector:-

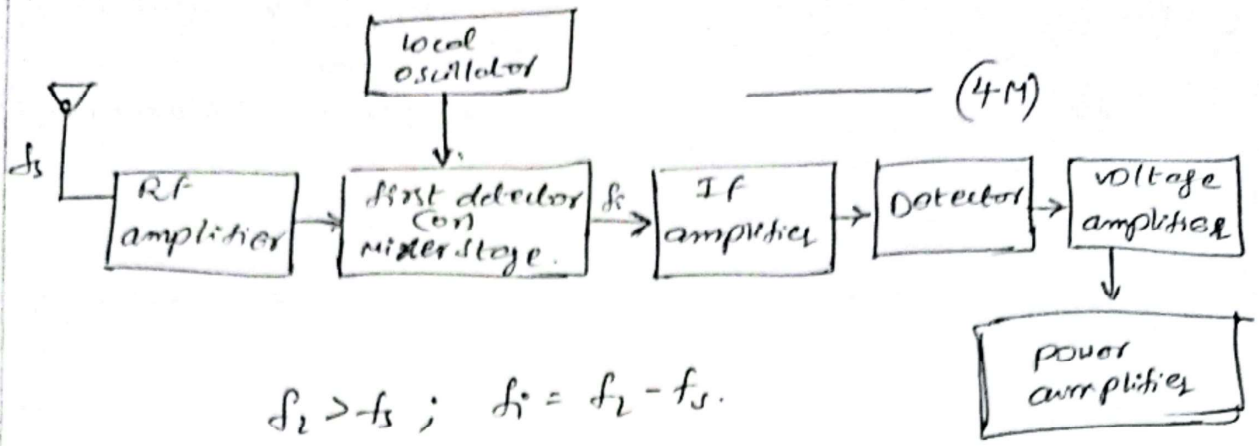


——— (2M)

Envelop detector consists of diode (D) followed by RC circuit.

(2)

(a)



——— (4M)

$f_L > f_s$; $f_i = f_L - f_s$.

fig: Block diagram of superheterodyne receiver.

superheterodyne receiver is the most popular receiver which has far better performance than TRF receivers. Significant feature of this receiver is that all incoming radio frequencies are converted into a single intermediate frequency "f_i" by the heterodyning process. The incoming carrier and the locally generated signal "f_L" are mixed in mixer also referred to as first detector. Mixer generates sum and difference frequencies at the output. $f_L - f_s$ is selected by a properly tuned circuit. As $f_L > f_s$, the receiver is called superheterodyne receiver.

The intermediate frequency in commercial radio receivers is fixed to 455kHz. All the incoming carrier frequencies are converted in to fixed IF frequency. Hence the circuit design becomes simple with improved performance. ——— (3M)

The IF signal is amplified in to an intermediate frequency amplifier. This is used to be designed for a fixed frequency band centered at 455KHz. Actually it is easy to design and develop a high gain amplifier at fixed carrier frequency. The sensitivity of superheterodyne receiver is primarily decided by IF amplifier stage. The highrate IF - amplifier provide better sensitivity than TRF receivers. ——— (2M). (3/11)

---

(2) (b) Function of RF amplifier in the receiver:-

A RF amplifier essentially is a tuned amplifier that enables the input signal of broadcast (or transmitted information to control an output signal. The RF amplifier uses frequency determining networks to convert the input signal in to an output signal, providing the required response at a given frequency — (2M). (1M)

Main function of the RF amplifier in a superhetero-dyne receiver is to

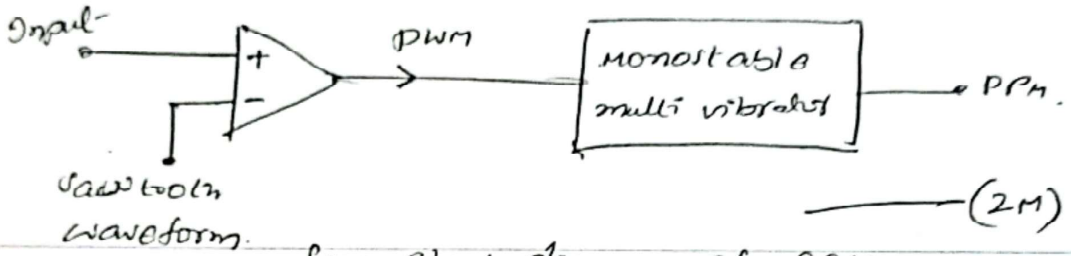
- Amplify the signal.
- Reject unwanted signal.
- Discriminate against image frequency signal and IF signal.

} — (3M). (2M)

The selected carrier is amplified by a class-C tuned amplifier. This is necessary to raise the carrier voltage level so that the linear diode detector following this stage may operate in linear region. The increased carrier voltage also suppresses the effect of noise. ——— (2M)

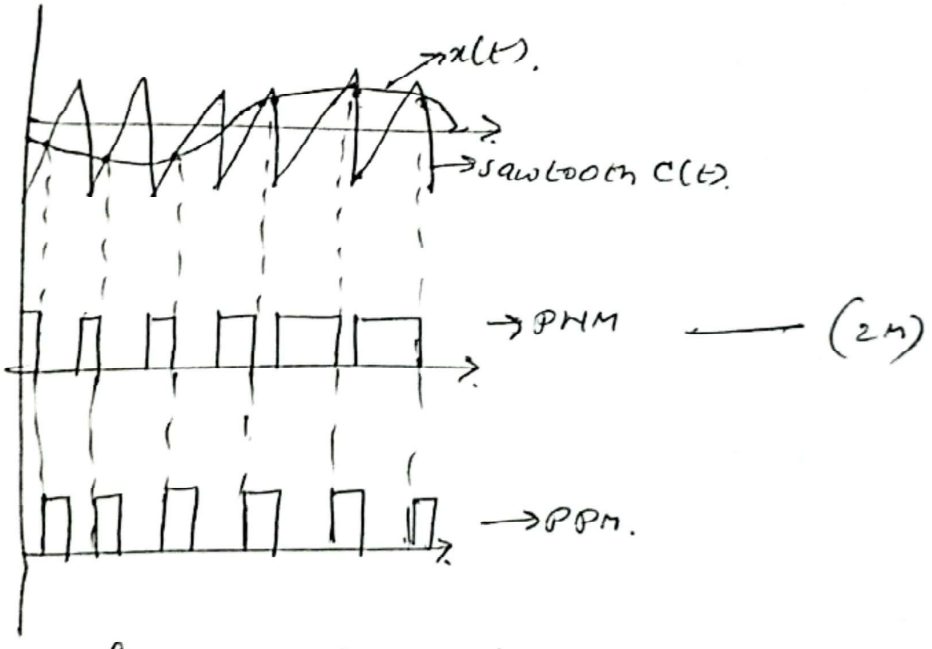
(3)

(a) PPM modulation:-



——— (2M)

fig:- Block diagram of PPM.



——— (2M)

fig:- waveforms of PPM.

PPM stands for pulse position modulation, the position of the pulse varies in accordance with m(t).

4/11

In the generation of PPM, modulating and sawtooth signals are given as inputs to comparator, where PPM can be generated from PWM through monostable - multivibrator.

The behaviour of the monostable multivibrator is, it will be in one stable state (low) and the otput becomes high when circuit is triggered. In the generation of PPM, PWM act as triggered pulse, such that wherever the PWM pulse appears, the output will become high. For the monostable multivibrator which is the output represented as PPM.

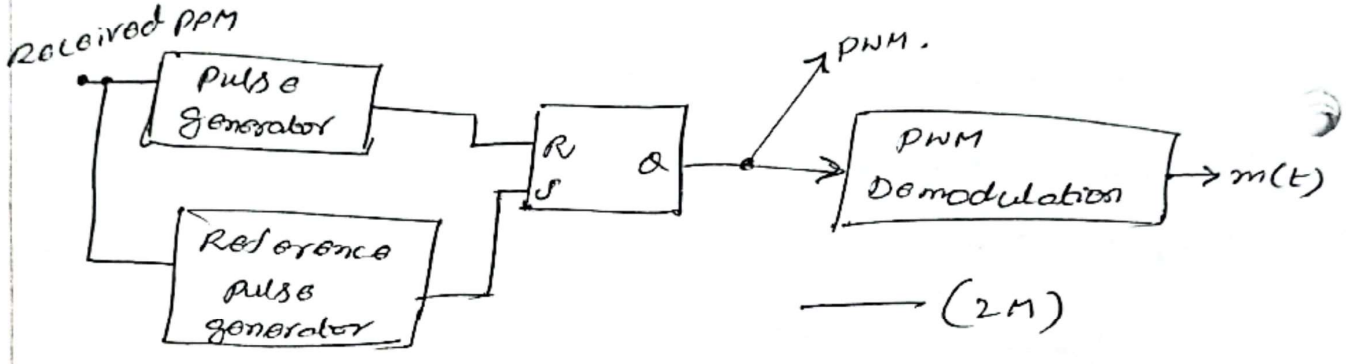
The starting of PPM will be the ending point of PWM. The comparator having reference level, this reference level crosses the sawtooth waveform. Second crossing of the reference modulating signal gives the starting of PPM and ending of PWM. Ending of PPM is decided by the time interval of sample pulses. The corresponding waveforms and block diagram of PPM is shown in the figure. It consists of the following conditions i.e.

If $x(t) > c(t) \longrightarrow$ Positive. ——— (3M)

$x(t) < c(t) \longrightarrow$ Negative.

when ever amplitude of m(t) is increased, the width of the pulse also increases. In the modulation of PPM when triggered the output becomes high, which is up to some duration, that depends on RC components of mono stable multivibrator.

(3)
(b) PPM Demodulation :-



(2M)

fig:- block diagram of PPM Demodulation.

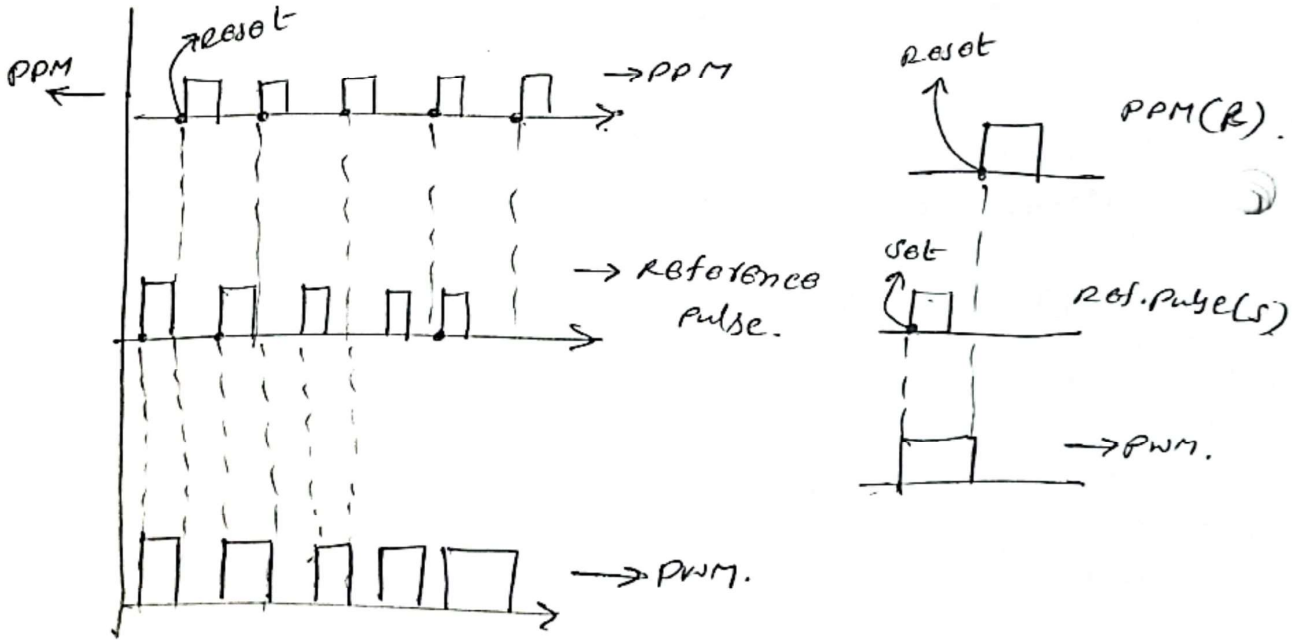


fig:- waveforms of PPM demodulation. —— (2M)

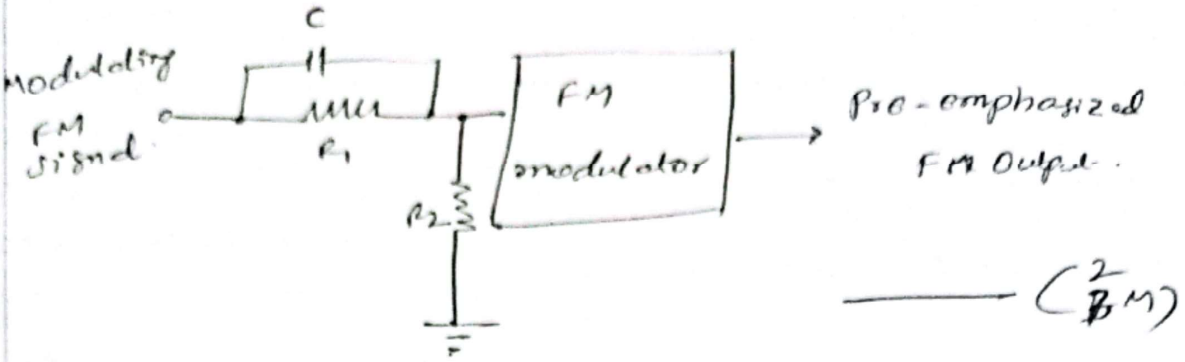
(5/17)

In the demodulation of PPM, the outputs of pulse generator(R) and reference pulse generator given to SR flipflop. The output of flipflop is PWM, which further given to PWM demodulator, where finally demodulated signal will be the final output ——(2M)

From the waveforms from the PPM pulse, reference pulses are generated. Depending on "set" and "Reset" levels corresponding PWM generated. From PWM, again demodulation done using PWM demodulator. Finally, modulating signal is retained from the modulated signal. ——→ (1M)

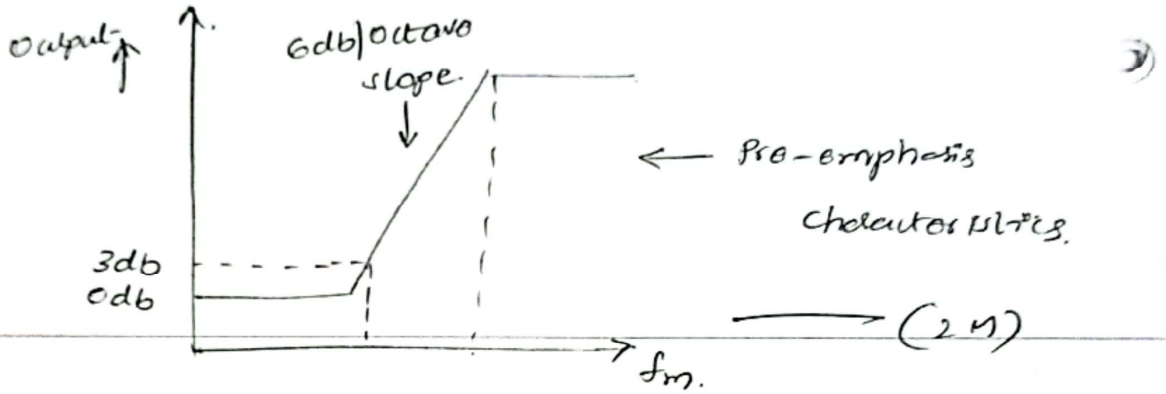
---

(4) (a) The artificial boosting of higher modulating frequencies is called as Pre-emphasis. The modulating AF signal is passed through a high pass RC filter, before applying to the FM modulator. As fm increases, reactance of 'C' decreases and modulating voltage applied to FM modulator goes on increasing. Pre-emphasis circuit is basically a high pass filter. It is carried out at the transmitter. —— (1M)



(2M)

fig:- Pre-emphasis circuit.



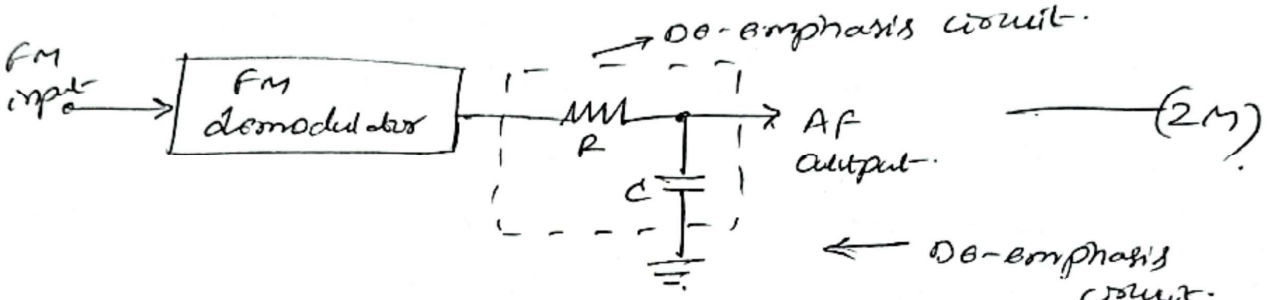
(2M)

The frequency response characteristics of the RC high-pass network is shown in the figure. Boosting is done acordig to the pre-arranged curve.

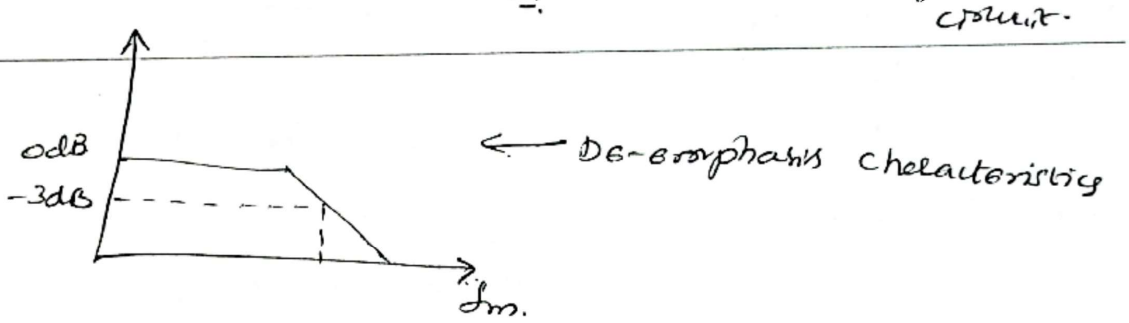
The combined effect of pre-emphasis and de-emphasis is to increase the high frequency components duriy the transmission so that they will be stronger and not masked by noise. As a result, the pre-emphasis at the transmitter is exactly offset by the de-emphasis circuit in the receiver, providig a normal frequency response. —— (2M)

6/11

The process that is used at the receiver end to nullify (or) compensate the artificial boosting given to the higher modulating frequencies in the process of pre-emphasis is called De-emphasis. Artificially boosted high frequency signals are brought to their original amplitude using the de-emphasis circuit.



(2M)



(4)

(b) Given that $V_m = 5V$; $f_m = 15KHz$

(a) For FM system

Frequency deviation $\Delta f = k_f \cdot V_m$.

$$m_f = 15 \times 10^3 \times 5 = 75KHz.$$

Hence modulation index, $m_f = \frac{\Delta f}{fm} = \frac{75}{15} = 5.$ — (1M)

According to carson's rule, the bandwidth is

$$BW = 2(\Delta f + fm) = 2(75+15) = 2 \times 90 = 180KHz.$$

$\therefore$ Bandwidth (FM) = 180 KHz.

(b) For PM system

Frequency deviation $\Delta f = k_p \cdot V_m \cdot f_m = 15 \times 10^3 \times 5 \times 15 \times 10^3$.

$\therefore \Delta f = 1125$ MHz.

$\therefore$ Bandwidth $= 2(\Delta f + f_m) \cong 2\Delta f = 2 \times 1125$ MHz. $= 2250$ MHz.

$\therefore$ Bandwidth (PM) = 2250 MHz

It is clear that the bandwidth is quite large as compared to FM. This is due to the fact that the modulation index in PM is quite large as

$$m_p = k_p \cdot V_m = 15 \times 10^3 \times 5 = 75{,}000.$$

This very large value of modulation index m_p produces a large number of sidebands and thus this is the reason for very large bandwidth. ——— (2M)

(ii) In this case $f_m = 5$ KHz.

(a) For FM; since frequency deviation Δf is independent of f_m and hence remains 75 KHz. Thus, $m_f = \frac{\Delta f}{f_m} = \frac{75}{5} = 15$.

$\therefore$ Bandwidth will be BW $= 2(\Delta f + f_m) = 2(75+5) = 2 \times 80$

$= 160$ KHz.

This means that in FM, modulation index changes sufficiently with a change in f_m, however, bandwidth changes only slightly

(b) For PM,

$\Delta f = k_p \cdot V_m \cdot f_m = 15 \times 10^3 \times 5 \times 5 \times 10^3 = 375$ MHz. $\therefore \Delta f = 375$ MH.

$\therefore$ Bandwidth = BW $= 2(\Delta f + f_m) \cong 2\Delta f = 2 \times 375 = 750$ MHz.

Thus, hence the bandwidth has changed compared to 2250 MH

modulation index $m_p = k_p \cdot V_m = 15 \times 10^2 \times 5 = 75$ KHz.

Which is same as previous one i.e. 75 KHz.

——— (2M)

7/11

(5)

(a) The entire radio frequency (RF) range is classified in to several bands as given below (10M) (9M) 9 bands.

SNO	Class	Frequency range	Wavelength	Application
1	very low frequency (VLF)	3KHz - 30KHz	$10^5 - 10^4$ m	Submarines.
2	Low frequency (LF)	30KHz - 300KHz	$10^4 - 10^3$ m	Navigation purpose.
3	Medium (MF) frequency	300KHz - 3MHz	$10^3 - 10^2$ m	Analog cordless Phones.
4	High frequency (HF)	3MHz - 30MHz	$10^2 - 10$ m	Aeronautical applications.
5	very high frequency (VHF)	30MHz - 300 MHz	10 - 1m	FM broadcasting
6	Ultra high frequency (UHF)	300MHz - 3GHz	< 1m	satellite communication
7	super high frequency (SHF)	3GHz - 30GHz	—	TV broadcasting.
8	Extremely high frequency (EHF)	30GHz - 300GHz	—	Multimedia point-point.
9	Extremely low frequency (ELF)	30Hz - 300Hz	-	simple half-duplex.

(5)
(b) Communication is a process by which the information is transmitted from one point to another. Thus, the procaa of communication involves sending, receiving and processing information in electrical form which is previously in non-electrical form. (1M)

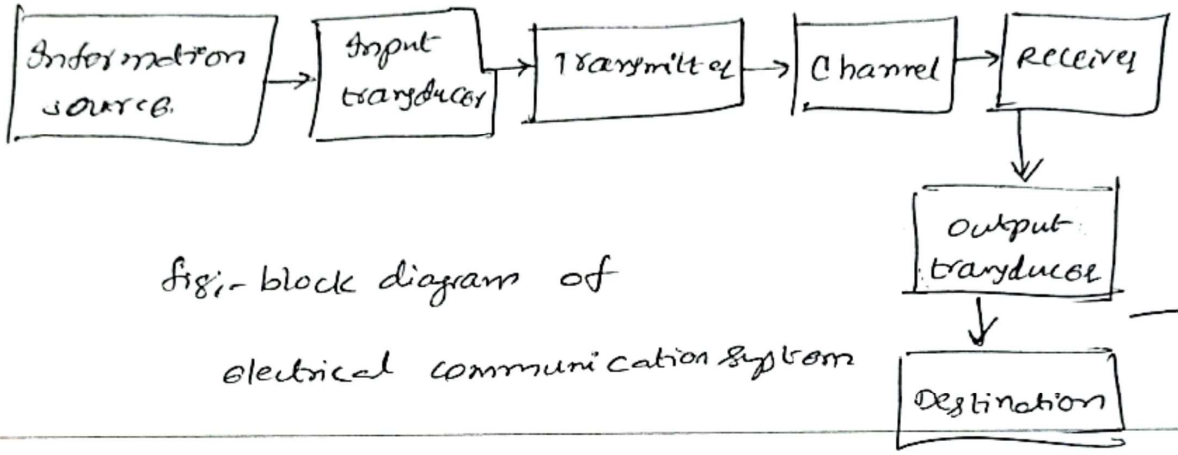


fig:- block diagram of electrical communication system (2M)

Transducer is to convert from electrical to non-electrical at receiver and from non-electrical to electrical at transmitter. Modulation is done at transmitter section, where as demodulation is done at receiver. Noise interferes with the modulated signal at Channel. Modulation is a process, some parameters of the carrier varies in accordance with the instantaneous value of the modulating signal. The process of recreating the original baseband signal is called demodulation. (1M)

(6)
(a) The transfer function $H_p(\omega)$ of pre-emphasis circuit is given by

$$H_p(\omega) = \frac{V_o(\omega)}{V_i(\omega)}.$$

Assuming R<<r, the current $I_1(\omega)$ is given by

$$I_1(\omega) = V_i(\omega)\left(\frac{1}{r} + j\omega C\right)$$

Hence $V_o(\omega) = R\left(\frac{1}{r} + j\omega C\right) V_i(\omega)$

The transfer function is,

$$H_p(\omega) = \frac{V_o(\omega)}{V_i(\omega)} = \frac{R}{r}(1 + j\omega Cr) = \frac{R}{r}\left(1 + j\frac{\omega}{\omega_1}\right)$$

where $\omega_1 = \frac{1}{rC}$. ——— (3M).

similarly, for the deemphasis circuit, the transfer function $H_d(\omega)$ is given by

$$H_d(\omega) = \frac{V_o(\omega)}{V_i(\omega)}.$$

$$\Rightarrow V_o(\omega) = I_2(\omega) \times \frac{1}{j\omega C} = \frac{V_i(\omega)}{r + \frac{1}{j\omega C}} \times \frac{1}{j\omega C}.$$

Hence, $H_d(\omega) = \frac{V_o(\omega)}{V_i(\omega)} = \frac{1}{1 + j\left(\frac{\omega}{\omega_1}\right)}$.

where $\omega_1 = \frac{1}{rC}$.

Therefore, the product of transfer function is given by.

$$H_p(\omega) \cdot H_d(\omega) = \frac{\gamma}{\beta}$$ which is constant with frequency. ——— (2M)

(6)

(b) The phase deviation produced by the noise signal in FM is much smaller than the phase deviation produced by the modulating signal, provided that noise is smaller than the carrier. Thus, in the low-noise case, the distortion produced by noise at the output of the FM detector is negligible in comparision to the desired modulating signal. In other words, noise is almost suppressed by the signal. The noise suppression characteristic of FM can also be applied to the common channel interference. This phenomena is known as capture effect. ——— (2M)

"When FM signals from two transmitters operated on the same (or nearly same carrier frequency reach the receiver simultaneously, the signal of a weak magnitude is suppressed by a

strong signal, and the FM receiver reproduces only the strong signal." ——— (3M)

This effect is similar to noise-suppressing action, with the weaker signal playing the role of the noise. The weak signals due to common channel and adjacent channel interferences are suppressed. ~~Am~~ AM lacks this feature, where FM system contains a very useful feature. ——(2M)

(7) Generation of AM signals using square law modulator:-

(a)

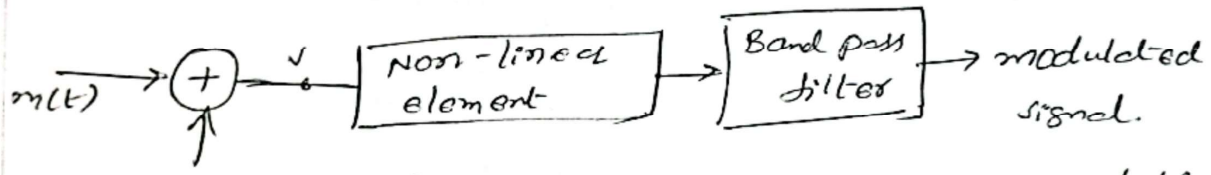
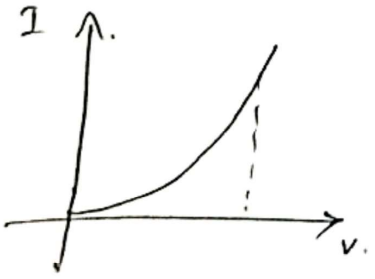
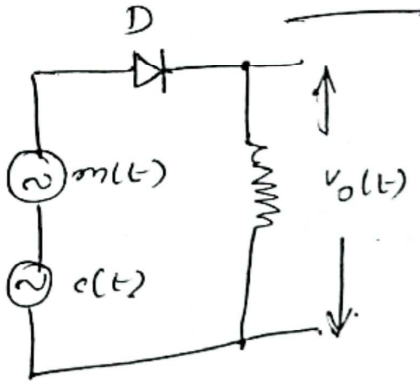


fig:- block diagram of square law modulator.

——— (3M)

Non-linear region of diode.





square law modulator consists of three elements, they are summer, non-linear device and band pass filter.

(i) Summer :- It adds the carrier $c(t)$ and modulating signal $m(t)$.

(ii) Non-linear device :- It has non-linear input and output relation.

ex:- semiconductor transistor, Diode.

(iii) Band pass filter :- (BPF)

It extracts desired signal from modulated output because of non-linearity of transfer characteristics of diode and the magnitude of carrier component is greater during positive half cycle of modulating signal and lesser in magnitude during negative half cycle of modulating signal.

The equation of squale law is $i_0 = a_1 v + a_2 v^2$ — (1)

The non-linear V-I characteristics of diode can be expressed by a squale law equation i.e. eq (1), where a_1 and a_2 are taylor series constants. — (3M)

Let $V_c(t) = V_c \sin \omega_c t$; $V_m(t) = V_m \sin \omega_m t$.

$V = V_m(t) + V_c(t)$ — (2).

Substitute eq (2) in (1)

$\Rightarrow i_0 = a_1 (V_m(t) + V_c(t)) + a_2 (V_m(t) + V_c(t))^2$.

$$\Rightarrow i_0 = a_1(V_m \sin\omega_m t + V_c \sin\omega_c t) + a_2(V_m \sin\omega_m t + V_c \sin\omega_c t)^2.$$

$$= a_1 V_m \sin\omega_m t + a_1 V_c \sin\omega_c t + a_2 V_m^2 \sin^2\omega_m t + a_2 V_c^2 \sin\omega_c^2 t + 2a_2 V_m V_c \sin\omega_m t \sin\omega_c t.$$

$$= a_1 V_m \sin\omega_m t + a_1 V_c \sin\omega_c t + a_2 V_m^2 \sin^2\omega_m t + a_2 V_c^2 \sin^2\omega_c t + a_2 V_m V_c [\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t.]$$

Neglecting the second order and higher order elements terms.

$$i_0 = a_1 V_m \sin\omega_m t + a_1 \sin\omega_c t\, V_c + a_2 V_m V_c (\cos(\omega_c - \omega_m)t) - a_2 V_m V_c \cos(\omega_c + \omega_m)t.$$

When the band pass filter is tuned to the carrier frequency it allows only ω_c, $\omega_c - \omega_m$ and $\omega_c + \omega_m$ and it eliminates all other terms (It allows only carrier frequency components).

$$i_0 = a_1 V_c \sin\omega_c t + a_2 V_m V_c \cos(\omega_c - \omega_m)t - a_2 V_m V_c \cos(\omega_c + \omega_m)t.$$

carrier | LSB | USB

(or)

(3M)

$$i_0 = a_1 V_c \sin\omega_c t + 2a_2 V_m V_c \sin\omega_c t \sin\omega_m t.$$

$$\Rightarrow i_0 = a_1 V_c \left[1 + \frac{2a_2}{a_1} V_m \sin\omega_m t\right] \sin\omega_c t.$$

$$= a_1 V_c [1 + m \sin\omega_m t] \sin\omega_c t.$$

where modulation index $\boxed{m = \frac{2a_2}{a_1}}$.

(7) (b) Power relations in AM Techniques :-

The total power required to transmit AM signal is the carrier power and sideband power i.e. $P_T = P_c + P_{SB}$ ——— (1M)

$= P_c + P_{LSB} + P_{USB}$.

If resistance is 'R' and voltage is 'V', then $P = \frac{V^2}{R}$

and $V_{rms} = \frac{V_{max}}{\sqrt{2}}$.

$\therefore P_c = \left(\frac{E_c}{\sqrt{2}}\right)^2 / R = \frac{E_c^2}{2R}$.

$P_{LSB} = \left(\frac{m_a E_c}{2\sqrt{2}}\right)^2 / R = \frac{m_a^2 E_c^2}{8R}$.

$P_{USB} = \left(\frac{m_a E_c}{2\sqrt{2}}\right)^2 / R = \frac{m_a^2 E_c^2}{8R}$.

$\therefore P_T = \frac{E_c^2}{2R} + \frac{m_a^2 E_c^2}{8R} + \frac{m_a^2 E_c^2}{8R}$.

$= \frac{E_c^2}{2R} + \frac{m_a^2 E_c^2}{4R}$.

$= \frac{E_c^2}{2R}\left[1 + \frac{m_a^2}{2}\right]$.

$$\Rightarrow \boxed{P_T = P_c\left[1+\frac{m_a^2}{2}\right]} \quad \because P_T \propto m_a \;\&\; 0 < m_a \leq 1.$$

(2M)

At $m_a = 1$ i.e. when modulation index = 1

$$\Rightarrow P_T = P_c\left[1+\frac{1}{2}\right] = 1.5P_c \Rightarrow \boxed{P_T = 1.5P_c}$$

It is maximum power required to transmit which is 1.5 times of carrier. In this power required for information transmission is equal to the power of sidebands which is equal to the sum of LSB and USB power.

At $m_a = 1 \Rightarrow P_{SB} = P_{LSB} + P_{USB}$.

$$= \frac{m_a^2 E_c^2}{2R \cdot 2} \quad \text{---------- (1M)}$$

$$= \frac{1}{2} P_c$$

$$\therefore \boxed{P_{SB} = 0.5P_c}.$$

Prepared by

S.V. Ratan Kumar,

Asst. Professor,

ECE Dept,

RGMCET,

Cell No: 9949671517.

27/12/22

(ESTD-1995)

1-a) VSB : Vestigial side Band is used in Video signals where where bandwidth is large for AM and DSB-SC and SSB-SC

→ It is also used in Radio Navigation.

→ Bandwidth is slightly increment than SSB-SC.

→ But power requirement is minimum.

Amplitude.

Subject : Analog communication

Q.P. Code : A04132005R1222

12413022

For Office Use

VALUATION-II

Q.No.	1	2	3	4	5	6	7
A							
B							
C							
D							
E							
F							
G							
TOTAL							

GRAND TOTAL

(in words)

Signature of the Examiner

Signature of the Scrutinizer

Subject : Analog Communication.

Q.P. Code : A04132005R1222

For Office Use 12413022

VALUATION-I

Q.No.	1	2	3	4	5	6	7
A	2	8	6	8	8	4	8
B	2	1	7	2	4	2	4
C	2						
D	2						
E	1						
F	2						
G	2						
TOTAL	13	9	13	10	12	06	12

GRAND TOTAL 60

(in words) SIX ZERO.

Signature of the Examiner

Signature of the Scrutinizer

For Office Use 12413022

INSTRUCTIONS TO THE STUDENTS

1. The student is prohibited from :

   - Writing his / her Hall Ticket Number in any part of his / her Answer Booklet
   - Writing his / her name or any other matter including God's name say Om, Rama etc. or any symbol which may lead to his / her identification, in any part of the Answer Booklet
   - Addressing the Examiner in any manner what-so-ever in his / her Answer Booklet.

2. Every student has to :

   - Enter the Question Paper Code Number received by them in the Block specified on the first Page of the answer booklet.
   - Write the Correct number and sub-number of the question, answered
   - Write answers legibly on both sides of the page.

3. No loose sheets or paper will be allowed into the examination hall and no paper must be detached from the Answer Booklet.

4. Students should not write more than 25 lines per page. It is not necessary to begin each answer on a fresh page.

5. Students will not be allowed into the Examination Hall after the commencement of the Examination and not permitted to leave the Examination Hall till the completion of the duration.

6. Carrying / Keeping Cell Phones in the examination hall will amounts for malpractice.

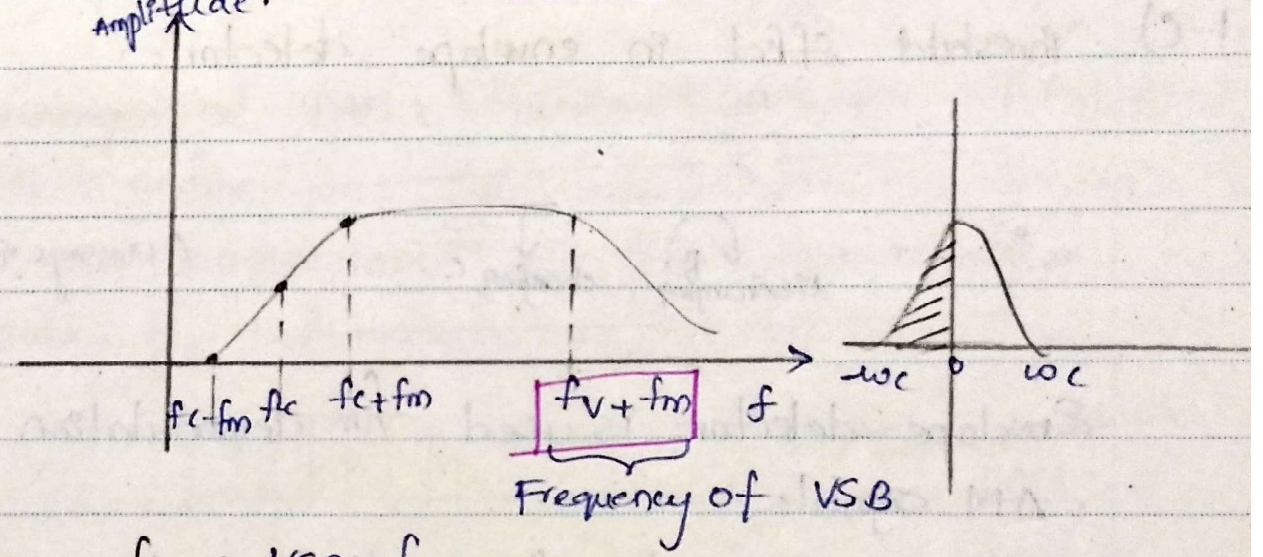
(ESTD-1995)

1-a) VSB : Vestigial side Band is used in Video signals where where bandwidth is large for AM and DSB-SC and SSB-SC

→ It is also used in Radio Navigation.

→ Bandwidth is slightly increment than SSB-SC.

→ But power requirement is minimum.



where fv = VSB frequency

fm = modulating signal frequency

1-b) Functions performed by Carrier in Radio Communication:

Carrier is the highest frequency signal, which doesnot contain any information.

→ Carrier function is reduce/~~increase~~ the Bandwidth for efficient transmission and Reception.

For the use of Evaluator Only

Q.No	Marks
(a)	2

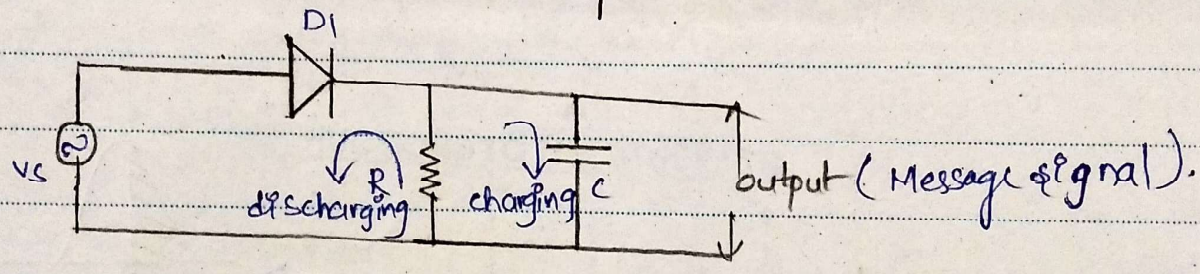
Q.No	Marks
1(b)	2

Q.No	Marks

(ESTD-1995)

Carrier is used for frequency multiplexing when the frequency components present in audio signal are same then at Receiver side used to modulate with different carrier frequency to differentiate the signals at Receiver.

1-c) Threshold effect in envelope detector:



Envelope detector is used for demodulation of AM signal.

⇒ The carrier pulse follows the envelope of modulating signal. So it is called envelope detector.

Threshold effect: It is the minimum value of input that gives the output not less than predicted value of output.

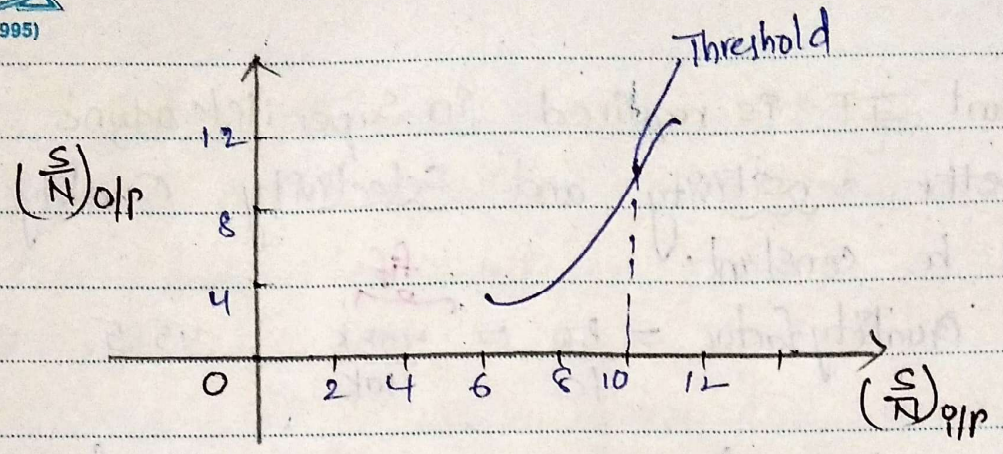
Threshold effect in FM: Minimum value of $(\frac{S}{N})$ that gives the output $(\frac{S}{N})_o$ not less than the predetected value of $(\frac{S}{N})$ using formula.

For the use of Evaluator Only

Q.No	Marks
1(c)	2

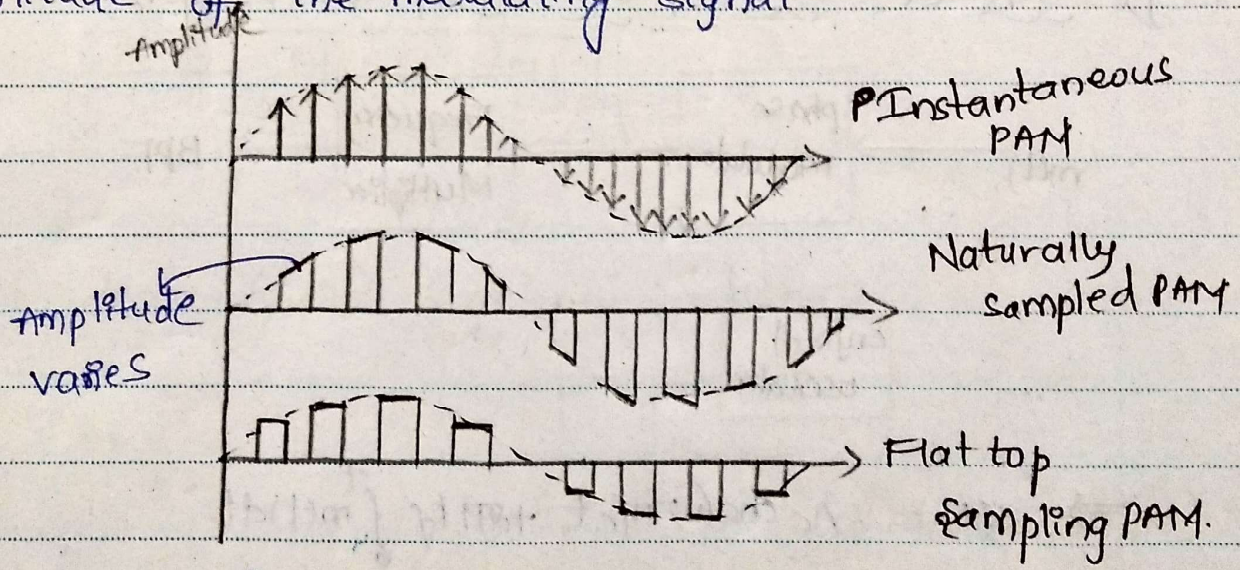
Q.No	Marks

Q.No	Marks



1-d) Disadvantages of PAM: (Pulse Amplitude Modulation)

PAM: It is defined as the amplitude of the carrier pulse varies in accordance with the instantaneous amplitude of the modulating signal.



Disadvantages: Due to the variation of amplitude, power required to transmit the signal is not constant. The transmitter should responds to the change in amplitude to corresponding power requirement.

→ Bandwidth is large.

For the use of Evaluator Only

Q.No	Marks
1(d)	2

Q.No	Marks

Q.No	Marks

(ESTD-1995)

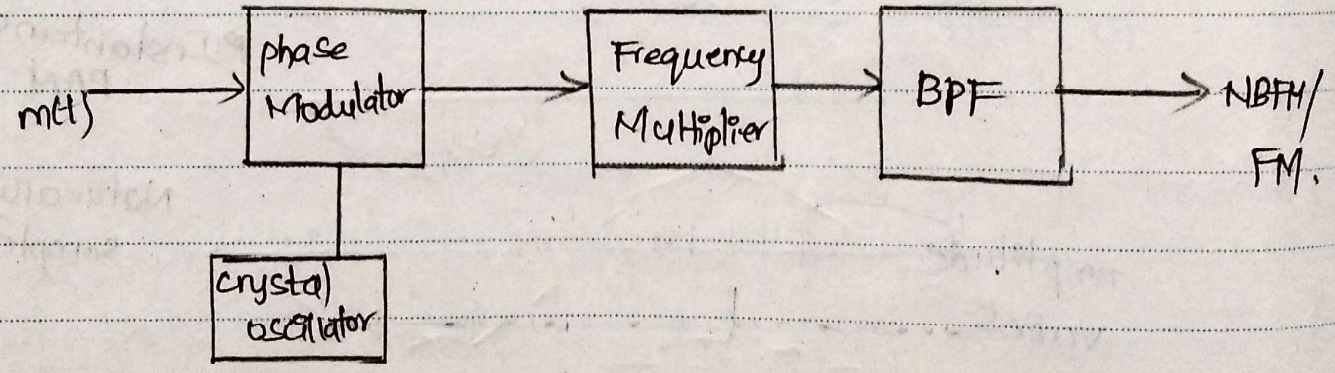
1-e) Constant IF is required in Super Heterodyne Receiver: For Better sensitivity and selectivity Quality factor should be constant.

$\Rightarrow$ For Quality factor $= \frac{B.W}{f_c} = \frac{455K}{10K} = 45.5$ (fif)

$\rightarrow$ Intermediate frequency is a constant frequency, it generates single frequency (ie 455K).

$\rightarrow$ Superheterodyne Receiver has constant Quality factor.

1-f) Indirect FM Generation: / NBFM Generation.



$\Rightarrow s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(t)dt\right)$

$m(t) = A_m \sin 2\pi f_m t$

$s(t) = A_c \cos\left[2\pi f_c t + 2\pi k_f A_m \frac{\cos 2\pi f_m t}{2\pi f_m}\right]$

$s(t) = A_c \cos\left[2\pi f_c t + \Delta f \cos 2\pi f_m t\right]$

For the use of Evaluator Only

Q.No	Marks
1(e)	1

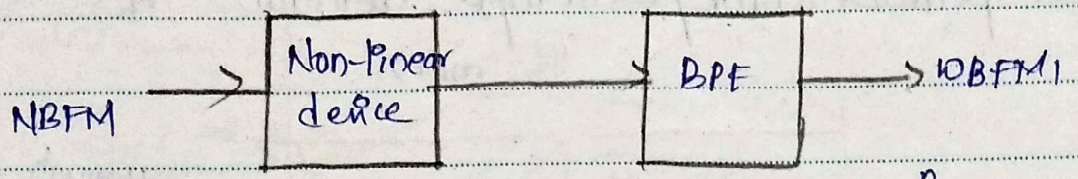
Q.No	Marks
1(f)	2

Q.No	Marks

(ESTD-1995)

BPF function is when it passes through Non-linear device and BPF. It allows band of frequencies.

ie $f_i(t) = f_c + K_f m(t)$.

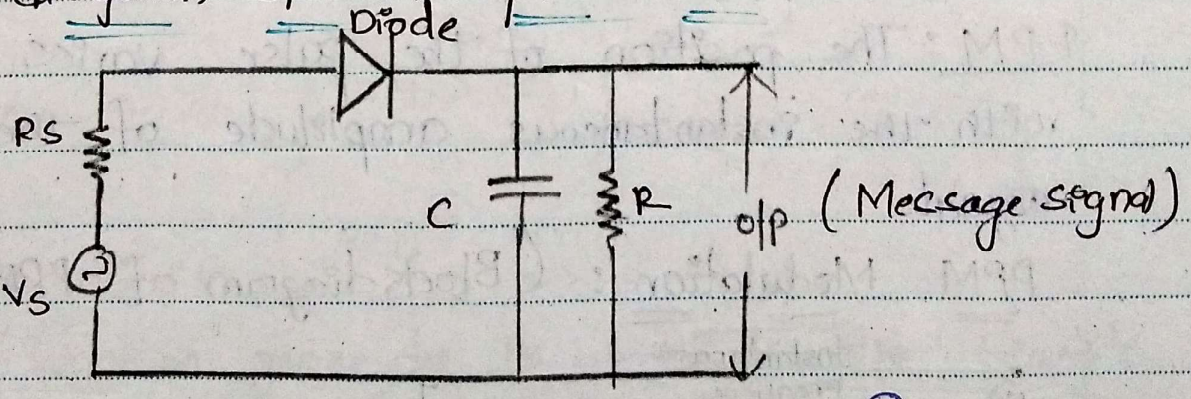


$$V(t) = V_1 s_1(t) + V_1 (s_1(t))^2 + \dots + V_n (s_n)^n(t).$$

By substituting s(t) we get carrier frequencies f_{c1}, f_{c2}, f_{c3} -- and frequency deviation $\Delta f_1, \Delta f_2, \dots \Delta f_n$

ie $\boxed{f_i(t) = nf_1 + K_{fn} m(t)}$

g) Circuit diagram of Envelope detector:



→ During (+)ve half cycle of input: Diode gets forward bias and capacitor charges upto peak of the modulating signal. $R_L C << \frac{1}{f}$

For the use of Evaluator Only

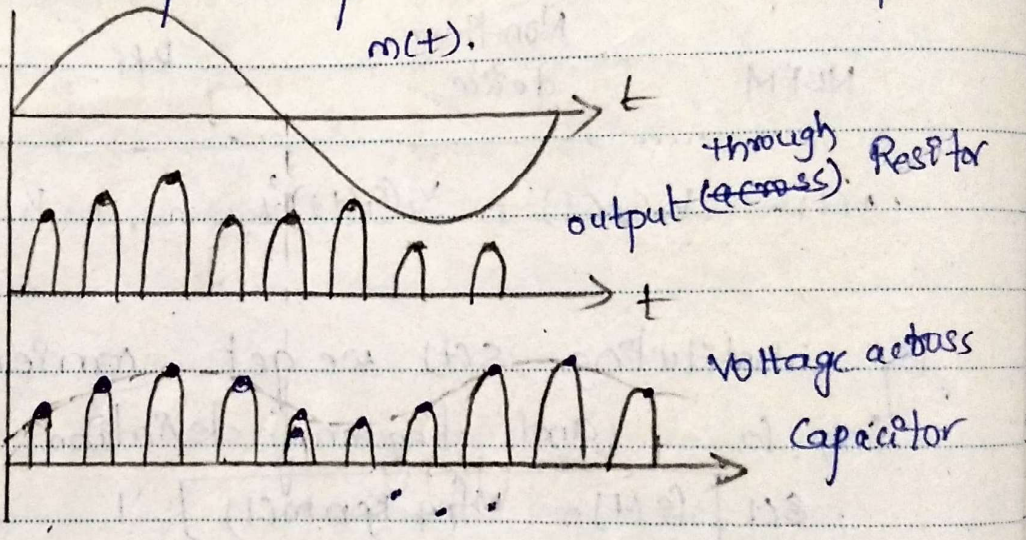
Q.No	Marks
11(g)	2

Q.No	Marks

Q.No	Marks

(ESTD-1995)

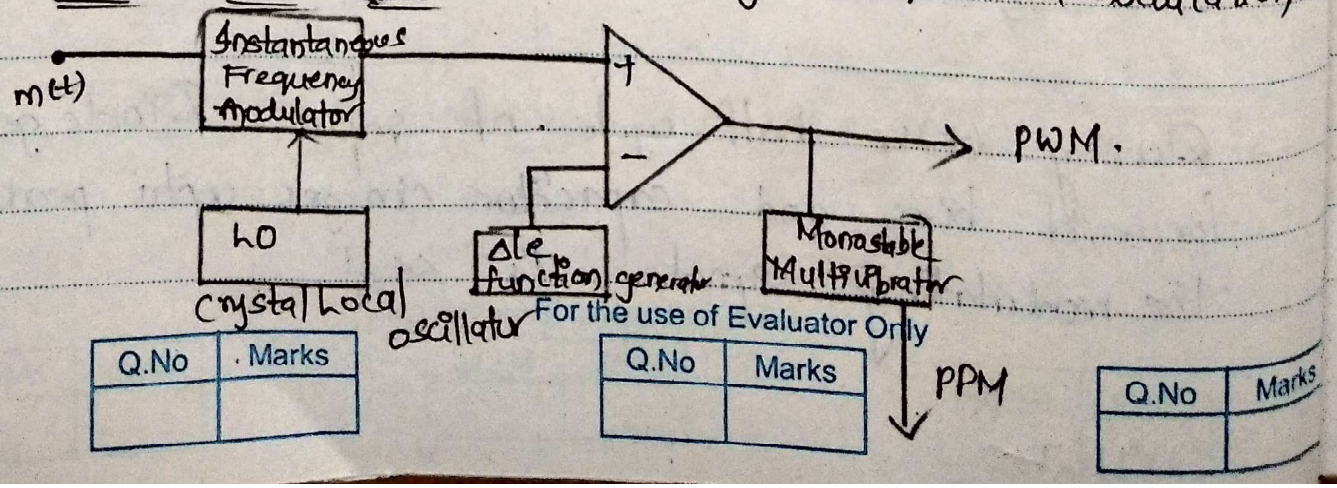
During (-)ve half cycle: capacitor discharges and capacitor output follows the envelope and it detect the peak of modulating signal. so it is called peak Detector / Envelope detector. $R_h C >> \frac{1}{f}$



3a) & PPM Modulation and b) PPM demodulation.

PPM: The position of the pulse varies inaccordance with the instantaneous amplitude of the modulating signal.

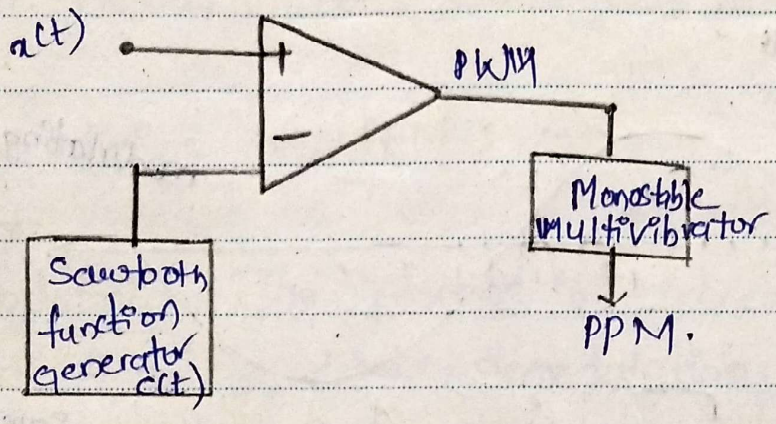
PPM Modulation: (Blocks diagram of PPM modulation)



(ESTD-1995)

→ We cannot generate PPM directly, with the help of PWM we generates PPM.

PWM Generator:



→ When modulating signal x(t) is greater than c(t) then output is (+)ve (High)

→ When modulating signal x(t) is less than c(t) then output is (-)ve.

$$V_o = (+)ve \quad \{x(t) > c(t)\}$$

$$V_o = (-)ve \quad \{x(t) < c(t)\}$$

The Ale function generator is used to the triangle pulses for PWM.

For the use of Evaluator Only

Q.No	Marks

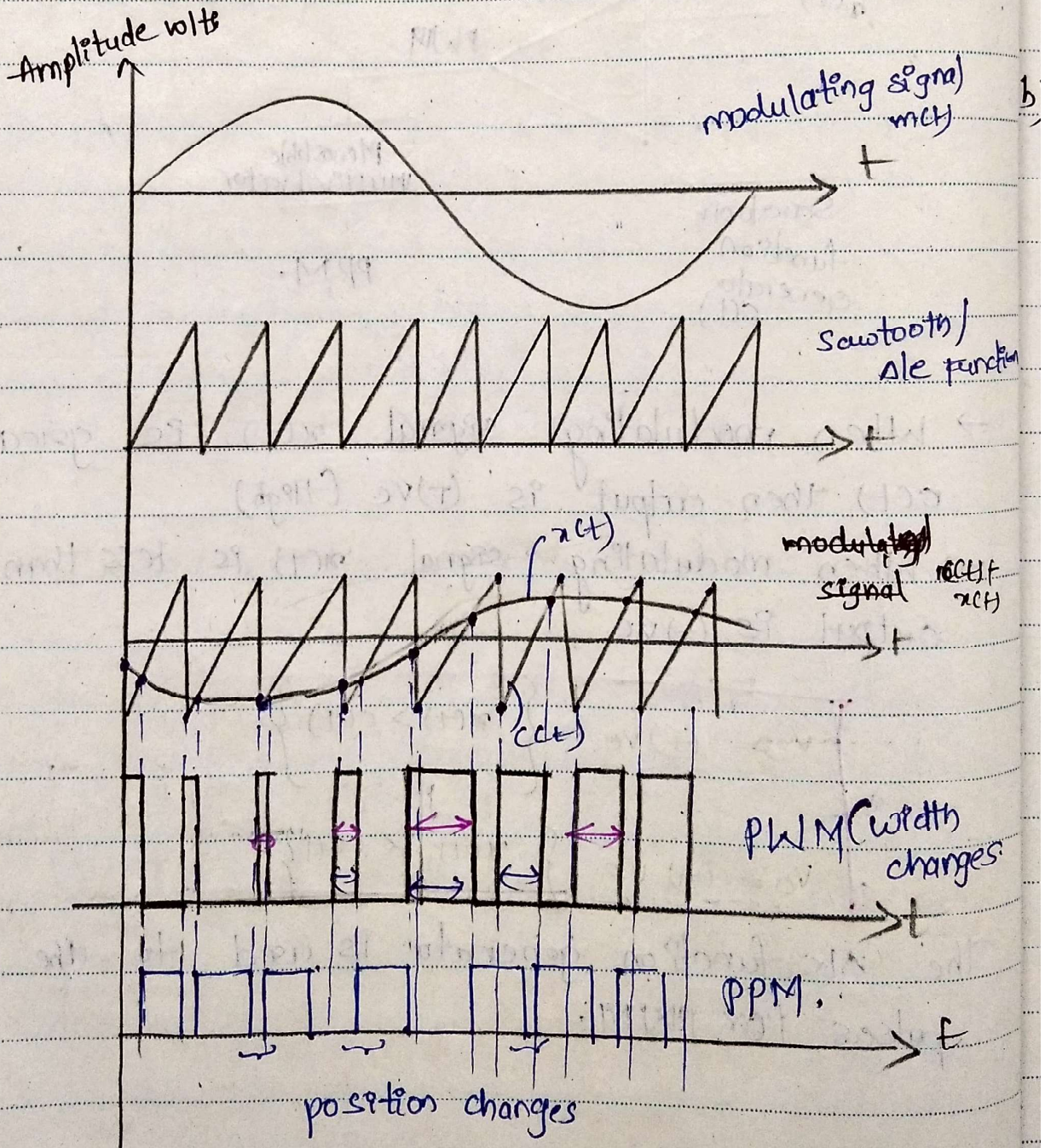
Q.No	Marks

Q.No	Marks

(ESTD-1995)

→ The tailing edge of PWM is starting / leading edge of PPM.

→ In this way PPM is generated.



For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks
3(a)	6

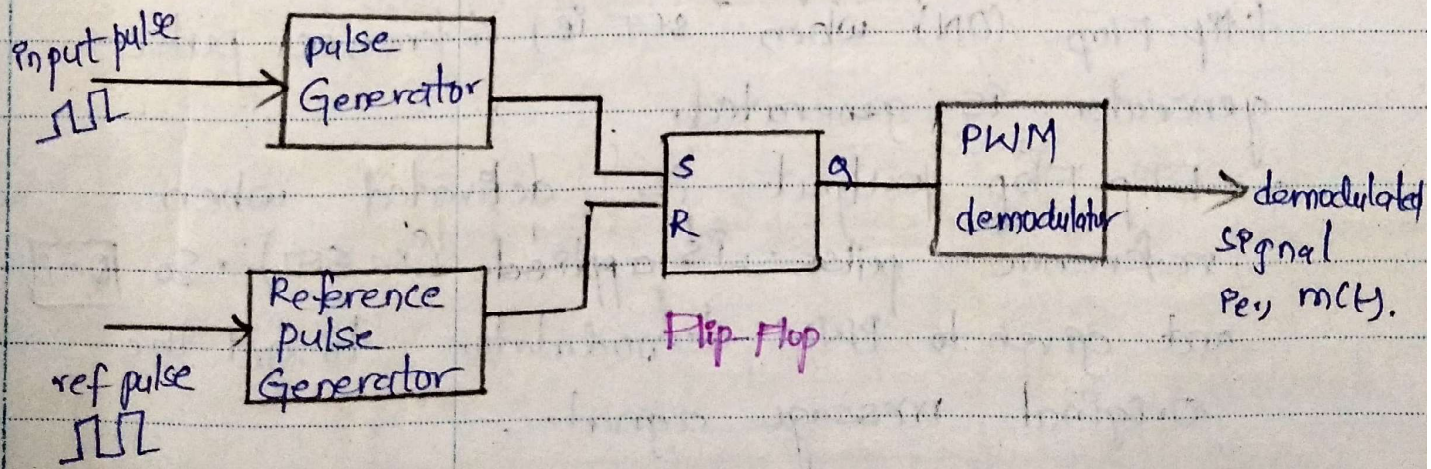
(ESTD-1995)

In PPM, position varies with respect to PWM.
→ When PWM is low, the ppM is high (ie starting point).
→ As width and Amplitude remains constant, so the power required for transmission is constant.

b) Demodulation of PPM:

Demodulation: The process of recreating original base-band signal is called demodulation.
→ In demodulation of PPM block diagram consists of pulse generator, Reference pulse generator, SR Flip-Flop and PWM demodulator and LPF.

Block diagram of Demodulation of PPM:



For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks

(ESTD-1995)

Pulse generator: It is used to generate pulses which contains RESET.

Reference pulse generator: It is also used to generate pulse but used to SET the signal.

SR- Flip-Flop:

S	R	Q	$\overline{Q}$	
0	0			previous state
0	1	0	1	
1	0	1	0	
1	1	X	X	Undetermined state.

Flip-Flop "OFF" when RESET signal is generated because (00) No change state.

Flip-Flop (ON) when SET ie, reference pulse generator is generated.

→ Flip-Flop output is activated when reference pulse is applied (ie SET). So Q=1 and given to PWM demodulator to get the Orginal message signal.

For the use of Evaluator Only

Q.No	Marks

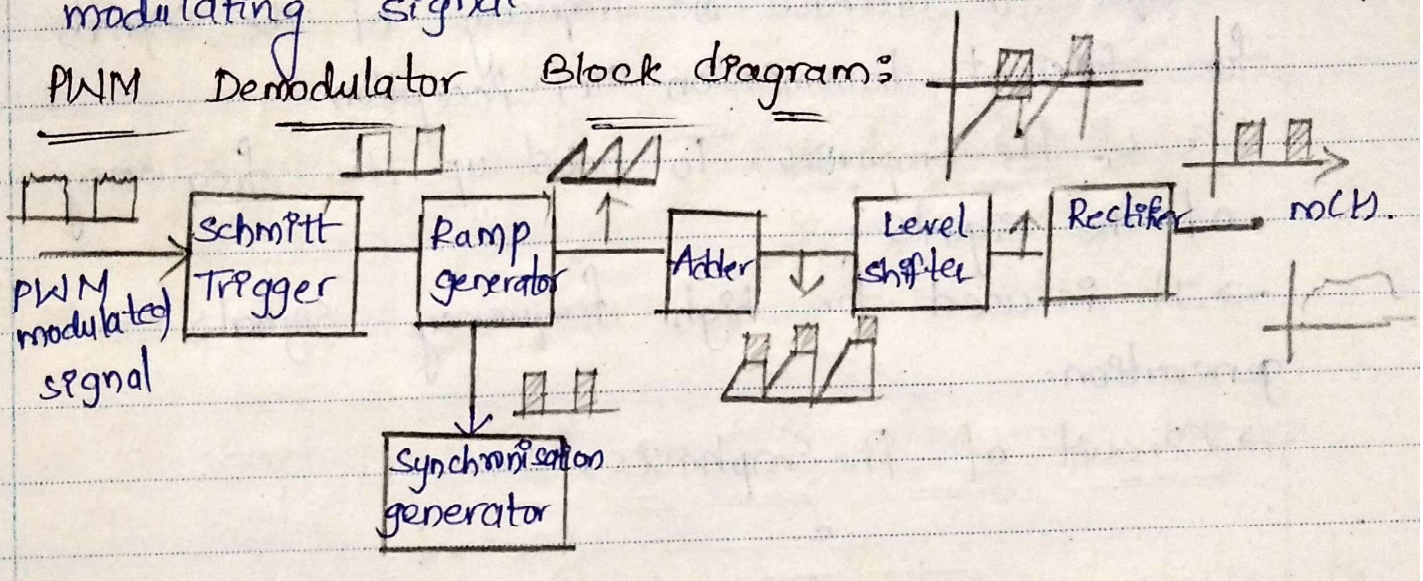
Q.No	Marks

Q.No	Mark

(ESTD-1995)

PWM Demodulator: It is used to get the original modulating signal.

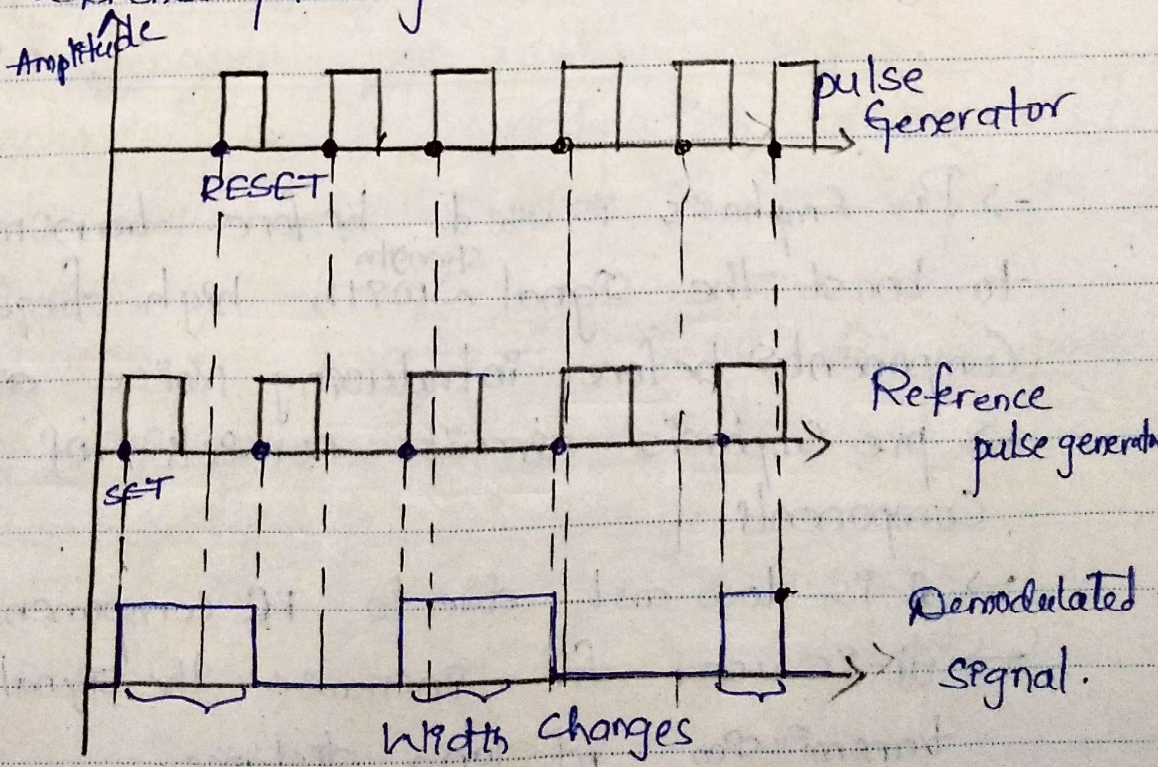
PWM Demodulator Block diagram:



PPM demodulation:

During pulse generator demodulated set to zero

During reference pulse generator it set to (1).



For the use of Evaluator Only

Q.No	Marks
3(b)	7

Q.No	Marks

Q.No	Marks

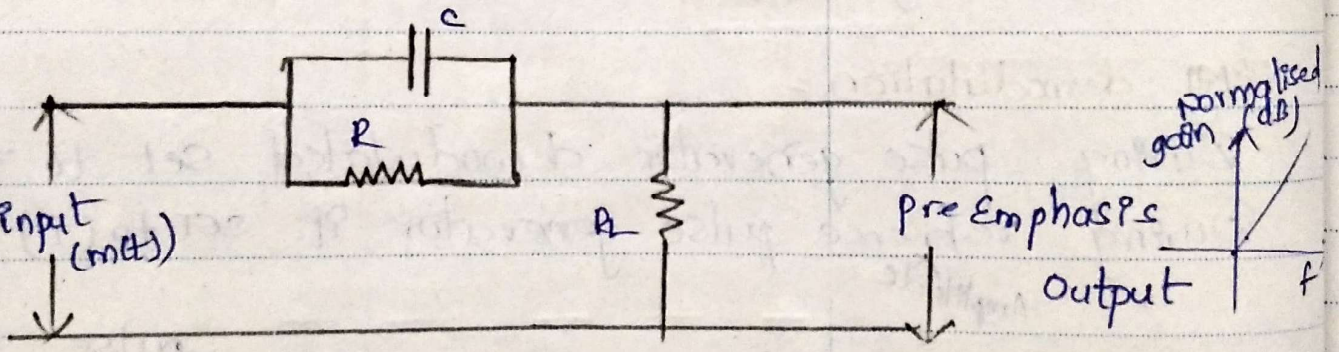
(ESTD-1995)

4a) **Pre-Emphasis**: Pre-emphasis a modulation technique to increase the frequency of the signals for efficient transmission and Reception.

Use of Pre-Emphasis: To boost up the frequency of the signals.

→ It is used for high frequency signals generation.

→ **Circuit of Pre Emphasis**:



→ Pre-Emphasis is used before transmitter to boost the signal strength with high frequency components before introducing Noise at Receiver.

→ pre-Emphasis circuit consists of R & C components

→ It is low cost due to RC components.

→ It is used for increase the signal for transmission of long distance.

For the use of Evaluator Only

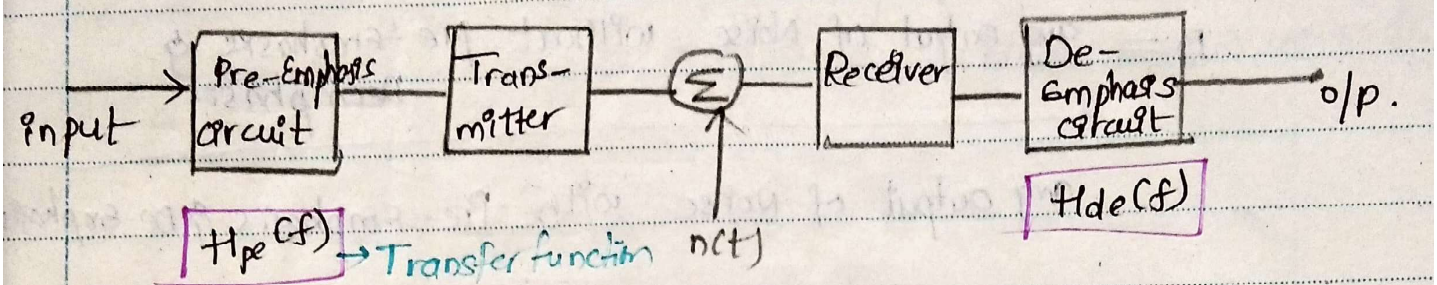
Q.No	Marks

Q.No	Marks

Q.No	Marks

(ESTD-1995)

Block diagram of Pre-Emphasis & De-Emphasis:



→ Pre-Emphasis is for boost the frequency before Transmitter.

→ De-Emphasis : De-Emphasis to unemphasing the output signal frequency.

→ De-Emphasis should made due to reduce the boost frequency at pre-Emphasis stage.

→ De-Emphasis is for demodulation, to recreate/recover the original input signal of Pre-Emphasis circuit.

→ (SNR) can be improved using pre-Emphasis and DE-Emphasis.

→ Hpe(f) → Transfer function of Pre-Emphasis
Hde(f) → Tranfer function of De-Emphasis.

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks

(ESTD-1995)

→ De-emphasis Transfer function should be inverse of pre-Emphasis Transfer function.

$D = \frac{\text{avg output of Noise without pre-emphasis \& Deemphasis}}{\text{avg output of Noise with Pre-emphasis \& De-emphasis}}$

$$\left(\frac{S}{N}\right)_0 = \frac{K}{2\pi A_c} \int_0^t H_{de}(f)\, df = \frac{K}{2\pi A_c} \int_0^t \cdots N_0 \quad j$$

$$= \frac{K (2\pi f)^2 N_0 \cdot A_c}{(2\pi)^2 A_c}$$

$$= \frac{K f^2 N_0}{A_c^2}$$

$$S_{Nd}(f) = \frac{K f_0^2}{A_c^2}$$

$$\frac{1}{A_c} \int_0^{\omega} f^2 \, df = \frac{f^3}{3}$$

$$D = \frac{K 2\omega^3}{3 A_c^2} \quad \left\{ \text{without Preemphasis} \right. \qquad \frac{1}{3A_c^2}\left[2\omega^3\right]$$

$$H_{pe}(f) = \frac{1}{1 + j(f/f_0)}$$

$$D = \frac{(\omega/\omega_0)^3}{\frac{N_0}{A_c}\left[\left(\frac{\omega}{f_0}\right) - \tan^{-1}\left(\frac{\omega}{\omega_0}\right)\right]}$$

$$\int \frac{1}{1+x^2} = \tan^{-1}$$

$$a\frac{x}{a} - \tan$$

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks
4(a)	8

(ESTD-1995)

$w = 15.9K$;

b

D = 13 dB, using pre-Emphasis and De-Emphasis which is improved with FM with 40-50db

4b) $m(t) = 5\cos(2\pi 15\times10^3)t$;

$m_c = A\cos\omega_c t$

(i) Modulation Index for FM =

$\beta = K_f \cdot f_m$

$\beta = \frac{\Delta f}{f_m}$

$\Delta f = K_f \cdot A(m)$

for $f_m = 15KHz$; $A_m = 5$; $A_c = A$

$\beta = \frac{\Delta f}{f_m}$

$\Delta f = f_c + f_m \,/\, f_c - f_m)$

$\Delta f = 15KHz$

$\therefore \max(\beta) =$

$\omega_c = 2\pi f_c t$

$2\pi f_c t$.

If $K_f = 1$

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks

(ESTD-1995)

$\Delta f = K_f \cdot A_m$

$\therefore \beta = \frac{\Delta f}{f_m} = \frac{\overbrace{K_f \cdot A_m}^{①}}{f_m} = \frac{(1)5}{15K} = 0.000333 < 1$

BW of FM is $(BW) = 2f_m$ $\quad \because m_a < 1$ NBFM

$= 2(15K)$

Bandwidth 30 KHz

(ii) Ma for $f_m = 5KHz$.

$m_a = \frac{K_f \cdot A_m}{5KHz} = 0.001 \quad \underbrace{< 1}_{\text{NBFM}}$

Bandwidth of NBFM $= 2f_m$

$= 2(5KHz)$

$= 10KHz$

As Frequency of modulating signal reduces, the Bandwidth also reduces from 30KHz to 10KHz.

ie) K_f : Frequency Sensitity.

For the use of Evaluator Only

Q.No	Marks
4(b)	2

Q.No	Marks

Q.No	Marks

(ESTD-1995)

5a) RF Bands values and their applications:

Frequency Band	Frequency Range	~~Bandwidth~~ Wavelength	Applications.
Very low Frequency (VLF)	300HZ-30KHZ	10^5-10^4 mts	Broadcasting, Radio wave Communication & Television.
Low Frequency (LF)	30KHZ-300KHZ	10^4-10^3 mts	Marine applications, Radiowave Broadcasting
Medium Frequency (MF)	300KHZ-3M	10^3-10^2 mts	FM Broadcasting, TV Broadcasting, Radio Broadcasting.
High Frequency (HF)	3M-30MHZ	10^2-10^1	Radio Communication, Radars, satellite Navigation.
Very High Frequency	30M-300MHZ	10^1-10^0	FM Broad casting, TV communication
Ultrahigh Frequency	300MHZ-3GHZ	-	satellite, Radars, Radio
Super High Frequency	3GHZ-30GHZ	-	Radio waves / Radio Broadcasting

For the use of Evaluator Only

Q.No	Marks
5(a)	8

Q.No	Marks

Q.No	Marks

(ESTD-1995)

Frequency Band	Frequency Range	~~Bandwidth~~ Wavelength	Applications
Extremely High Frequency	>300GHz	-	Media applications Radars, Radio, Satellite.
Extremely Low Frequency	30Hz - 300Hz	-	Broadcasting.

⇒ Frequency Range / Band of Radio Frequency bands is from 30Hz to 300GHz

→ Which is widely used higher frequency in RF bands is 300GHz for Media applications.

5b) Communication process:

Communication is the process of conveying / transfing information from one place to another.

Communication type based on Transmission Media

1) Line Communication
2) Radio Communication.

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks

(ESTD-1995)

1) Wire (channel) Communication: It uses the pair of Conductors for communication. At a time only one electromagnetic wave is transmitted.

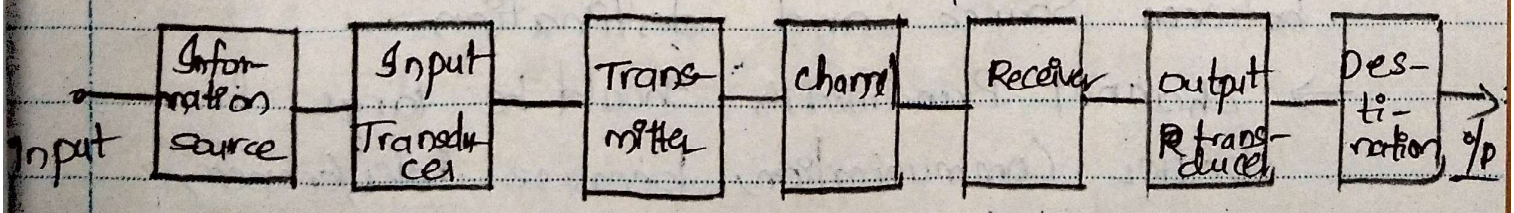
→ Limited transmission is possible

2) Radio Communication: The travel of EM waves in a open space is called Radio wave communication

→ While transfering the information, due to open space communication, they will interfere with other signals.

→ Due to this, to eliminate Interference, RF spectum is used and it acts as a seperate RF spectum for each signal, which is possible through Modulation.

Communication System:



For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks

(ESTD-1995)

Information source: It is used to get the information ie. input signal for Receiver.

Input Transducer: It is used to convert non-electical to electrical signal (ie Audio; msg) signal for Transmitter. It contains the signal, which is good for propagation through Transmitter.

Transmitter: Transmitter is not only used for Modulation but also raises the amplitude level of input signal

→ It generates a signal which is required for better propagation.

Transmitter output depends on Modulation type

Modulation type:
- AM } continous wave Modulation
- FM } PTM.
- PM }

Channel: It is space between transmitter and Receiver. channel is used to get the relation between Source and destination.

→ channel Bandwidth should be large for better Communication, because interference, noise will be added.

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

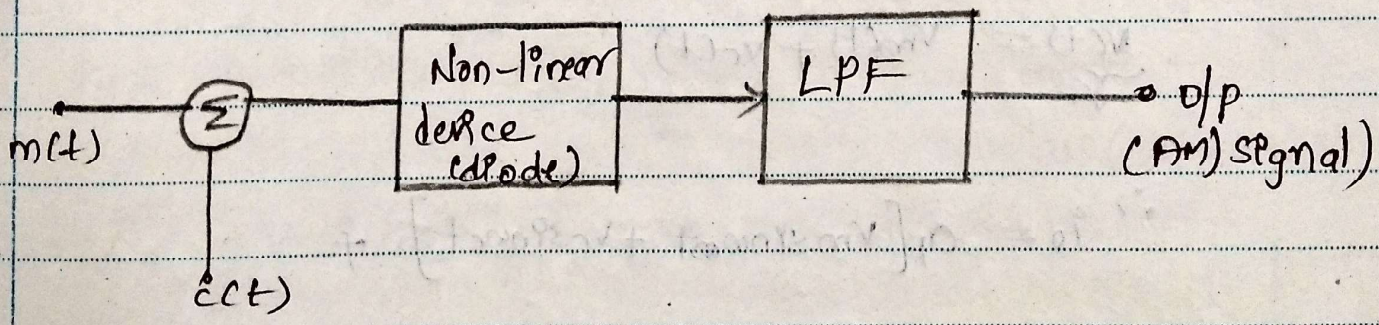
Q.No	Marks

(ESTD-1995)

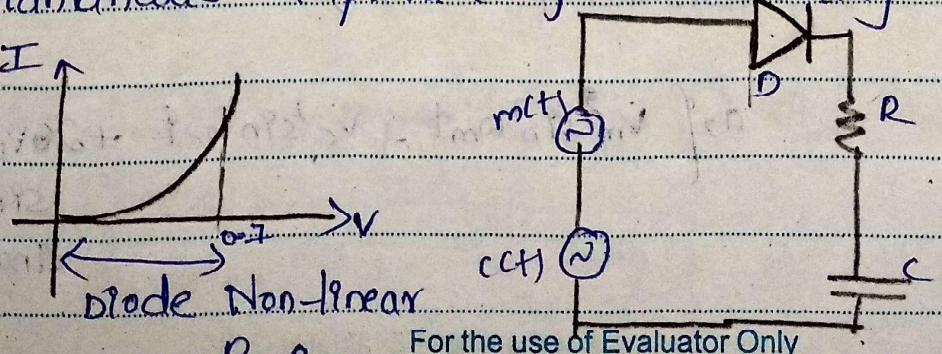
Receiver: Demodulation is done at Receiver section the process of recreating original information source.

Output transducer: It is used to convert electrical to non-electrical which is original information source.

7a) Generation of AM using Square law Modulator:



AM (Amplitude Modulation): The amplitude of the carrier signal various inaccordance with the instantaneous amplitude of modulating signal m(t).



For the use of Evaluator Only

Q.No	Marks
5(b)	4

Q.No	Marks

Q.No	Marks

(ESTD-1995)

The Non-linearity of (VI) characteristics of diode can be represented by using the equation as

$$i_0 = a_1 V + a_2 V^2$$

where a_1, a_2 are taylor series coefficients / Constants.

$$V_m(t) = A_m \sin\omega_m t$$

$$v_c(t) = V_c \sin\omega_c t$$

$$v(t) = v_m(t) + v_c(t)$$

$$\therefore i_0 = a_1\left[V_m \sin\omega_m t + V_c \sin\omega_c t\right] + a_2\left[V_m \sin\omega_m t + V_c \sin\omega_c t\right]^2$$

$$i_0 = a_1\left[V_m \sin\omega_m t + V_c \sin\omega_c t\right] + a_2\left[V_m^2 \sin^2\omega_m t + V_c^2 \sin^2\omega_c t + 2 V_m V_c \sin\omega_m t \sin\omega_c t\right]$$

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks

(ESTD-1995)

$$v_o = a_1 V_m \sin\omega_m t + a_1 V_c \sin\omega_c t + a_2 V_m^2 \sin^2\omega_m t + V_2 a_2 \sin^2\omega_c t + 2a_2 V_m V_c \sin\omega_m t \cdot \sin\omega_c t$$

When (v_o) is passed through lowpass filter all high frequency components are eliminated except carrier and low frequency terms.

$$v_o = a_1 V_c \sin\omega_c t + 2a_2 V_m V_c \sin\omega_m t \cdot \sin\omega_c t$$

$$v_o = a_1 V_c \sin\omega_c t + a_2 V_m V_c \left[\cos(\omega_c - \omega_m)t - \cos(\omega_c + \omega_m)t\right]$$

$$v_o = a_1 V_c \sin\omega_c t + a_2 V_m V_c \cos(\omega_c - \omega_m)t - a_2 V_m V_c \cos(\omega_c + \omega_m)t$$

Modulation Index is

$$v(t) = a_1 V_m \sin\omega_m t + 2a_1 a_2 V_m V_c \sin\omega_m t \cdot \sin\omega_c t$$

$$= a_1 V_m \left[a_1 + \frac{2a_2}{a_1} \sin\omega_m t\right] \cdot \sin\omega_c t$$

$$\therefore \boxed{m_a = \frac{2a_2}{a_1} V_c}$$

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

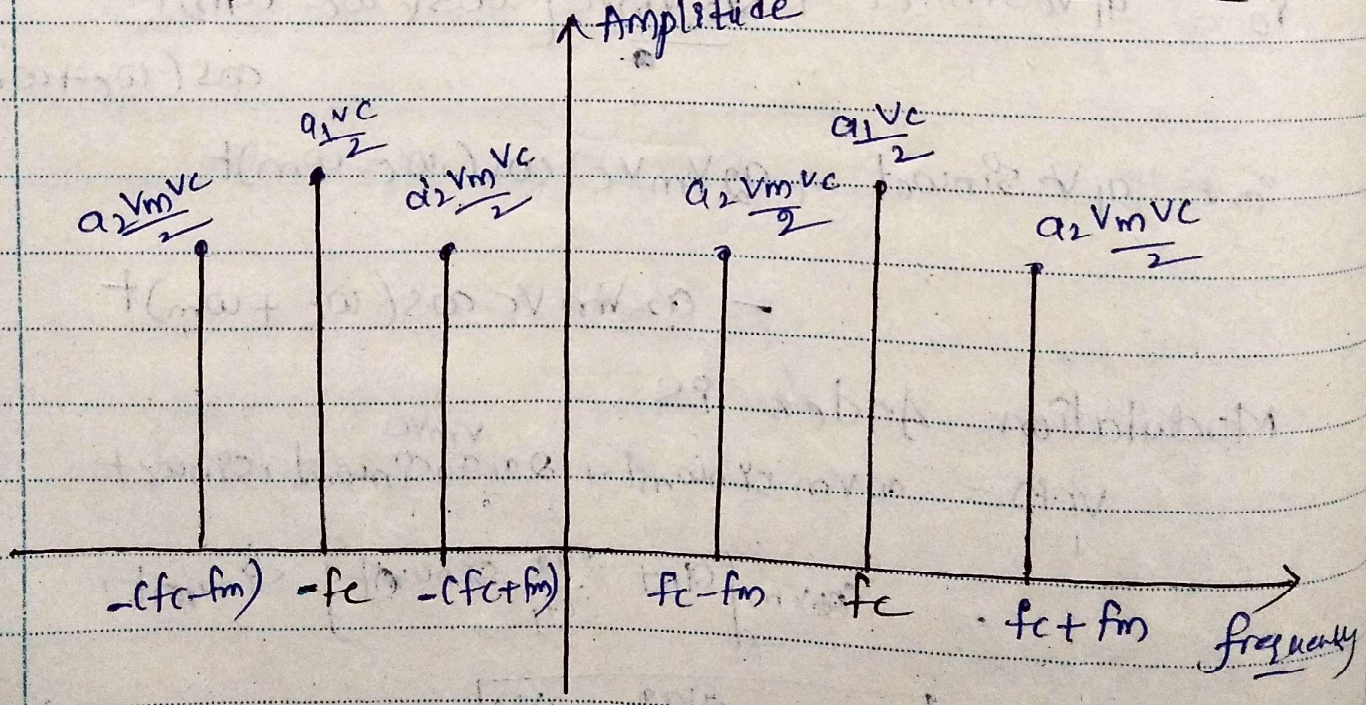
Q.No	Marks

(ESTD-1995)

$$i_o = a_1 V_c \sin\omega_c t + a_2 V_m V_c \cos(\omega_c - \omega_m)t - a_2 V_m V_c \cos(\omega_c + \omega_m)t$$

$$P_o(f) = \frac{a_1 V_c}{2}\left[\delta(f-f_c) - \delta(f+f_c)\right] + \frac{a_2 V_m V_c}{2}\left[\delta(f-(f_c-f_m)) + \delta(f+(f_c-f_m))\right] - \frac{a_2 V_m V_c}{2}\left[\delta(f-(f_c+f_m)) + \delta(f+(f_c+f_m))\right]$$

Frequency Spectrum of Square Law Modulator:



For the use of Evaluator Only

Q.No	Marks
7(a)	8

Q.No	Marks

Q.No	Marks

(ESTD-1995)

7b) Power Relations in AM techniques:

P_{total} is sum of Carrier & Side Band powers.

$P_T = V_{rms}^2 R$; $V_{rms} = \left(\frac{V_{max}}{\sqrt{2}}\right)^2$

$$P_C = \left(\frac{V_c}{\sqrt{2}}\right)^2 = \left(\frac{m_a E_c}{\sqrt{2}}\right)^2 = \frac{m_a E_c^2}{2}$$

$$P_{SideBands} = \left(\frac{m_a E_c}{2\sqrt{2}}\right)^2 = \frac{m_a^2 E_c^2}{8}$$

$$P_{USB} = P_{LSB} = \frac{m_a^2 E_c^2}{8}$$

$$\therefore P_{Total} = P_c + P_{USB} + P_{LSB}$$

m

4×2

$$P_T = \frac{m_a^2 E_c^2}{2} + \frac{m_a^2 E_c^2}{8} + \frac{m_a^2 E_c^2}{8}$$

$$P_T = \frac{m_a^2 E_c^2}{2}\left[1 + 1/8 + 1/8\right]$$

$1 + \frac{1}{4}$

$1 + 1/4$

$\frac{5}{4}$

$$P_T = m_a^2 P_c\left[1 + \frac{1}{2}\right]$$

$$P_T = P_c\left[1 + \frac{m_a^2}{2}\right] \quad \because AM ; \text{ let } m_a = 1.$$

For the use of Evaluator Only

Q.No	Marks

Q No	Marks

Q.No	Marks

(ESTD-1995)

$$P_T = P_c\left[1+\tfrac{1}{2}\right]$$

$$\boxed{P_T = P_c[1.5]_{max}}$$

$$P_c = 0.5 P_{max}$$

∴ Usable ~~frequency~~ power for AM is 0.5 only and remaining is wasted.

if, $$\eta = \frac{\text{Usable power}}{\text{Total power}} \times 100$$

$$\eta = \frac{P\frac{m_a^2 E_c^2}{4}}{P_c\left[1+\frac{m_a^2}{2}\right]} \times 100$$

$\frac{m_a^2 E_c^2}{4}$

$\frac{m_a^2}{2}[P_c]$

$$\eta = \frac{P_c\left[\frac{m_a^2}{2}\right]}{P_c\left[1+\frac{m_a^2}{2}\right]} \times 100$$

$$\eta = \frac{m_a^2}{2+m_a^2} \times 100$$

∴ Let max

$$\eta = \frac{1}{3} \times 100$$

$$\boxed{\eta = 33.333\%}$$

For the use of Evaluator Only

Q.No	Marks
7(b)	4

Q.No	Marks

Q.No	Marks

(ESTD-1995)

6a) Product of Transfer functions of Pre-Emphasis & De-Emphasis is constant;

Transfer function of Pre-Emphasis is

$$H_{pe}(f) = \frac{1}{1+j(f/f_0)}$$

∴ Transfer function of De-Emphasis is Inverse of Pre-Emphasis ie,

$$H_{de}(f) = \frac{1}{H_{pe}(f)}$$

$$\therefore \quad H_{de}(f) = \frac{1}{\frac{1}{1+j(f/f_0)}}$$

$$|H_{de}(f)| = \frac{1}{\frac{1}{\sqrt{1+(f/f_0)^2}}}$$

$$|H_{pe}(f)| = \frac{1}{\sqrt{1+(f/f_0)^2}}$$

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks

(ESTD-1995)

$\therefore$ product of $H_{pe}(f)$ & $H_{de}(f)$ is

$$H_{pe}(f) \cdot H_{de}(f) = \frac{1}{\sqrt{1+(f/f_0)^2}} \cdot \sqrt{1+(f/f_0)^2}$$

$$\boxed{H_{pe}(f) \cdot H_{de}(f) = 1}$$

$\therefore$ Hence, proved the product of transfer function of pre-Emphasis and De-Emphasis is constant.

6b) Capture- Effect: In FM, reception, the receiver selects the signals which is stronger of two signals based on magnitude level.

Explanation:

During transmission & reception of signals, if original signal is interfere with no desired FM signal then receiver receives the interefering signal.

For the use of Evaluator Only

Q.No	Marks
6(a)	4

Q.No	Marks

Q.No	Marks

(ESTD-1995)

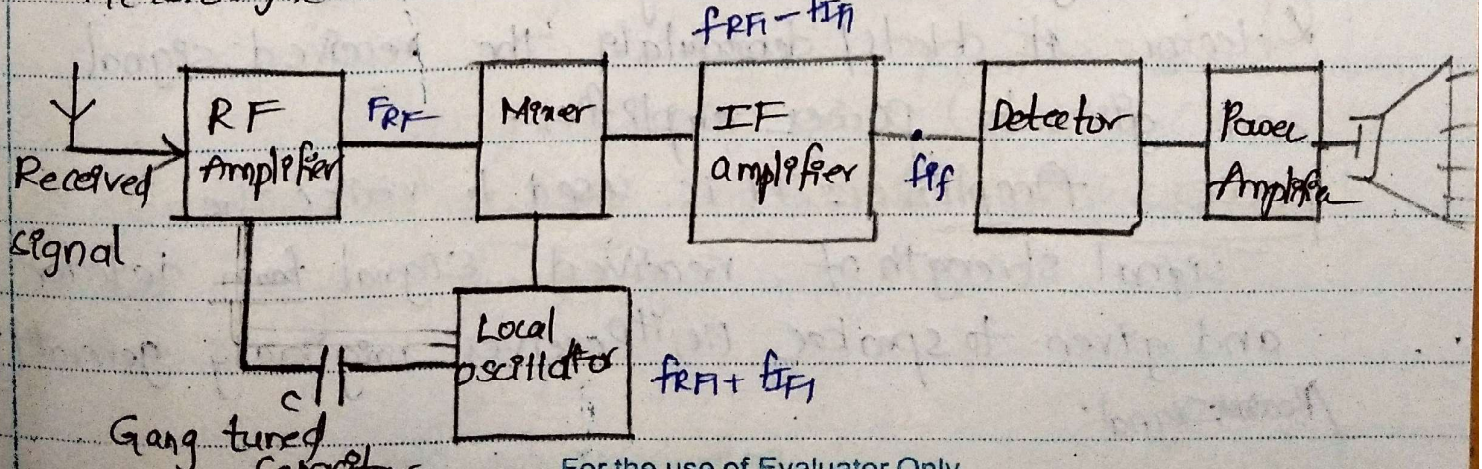
When interfering signal frequency is equal to the frequency of desired FM signal, then receiver back and forth between the signals and receives.

→ If interfering signal frequency is stronger than the desired FM signal then receiver holds the both signals for certain duration.

i.e) Receiver holds interfering signal for some time and desired FM signal for some time.

→ Due to this receiver unable to select the desired FM signal. So, this is Capture Effect.

2a) Process of Frequency change in Super-Heterodyne Receivers:



For the use of Evaluator Only

Q.No	Marks
8(b)	2

Q.No	Marks

Q.No	Marks

RF Amplifier: It is used to raise the frequency of desired extracted signal by rejecting all other frequencies.

→ It is given to Mixer

Local oscillator: It is used to generate the carrier signal and given to Mixer

→ Mixer is used to combine the RF amplifier frequency and local oscillator frequency.

Generally, it is a multiplier.

IF Amplifier: It is the intermediate frequency amplifier which amplifies single constant frequency throughout receiver (ie 455KHz)

→ It is a constant frequency generating amplifier for better quality factor which gives more sensitivity and selectivity of a receiver.

Detector: It detects / demodulates the received signal and given to power amplifier.

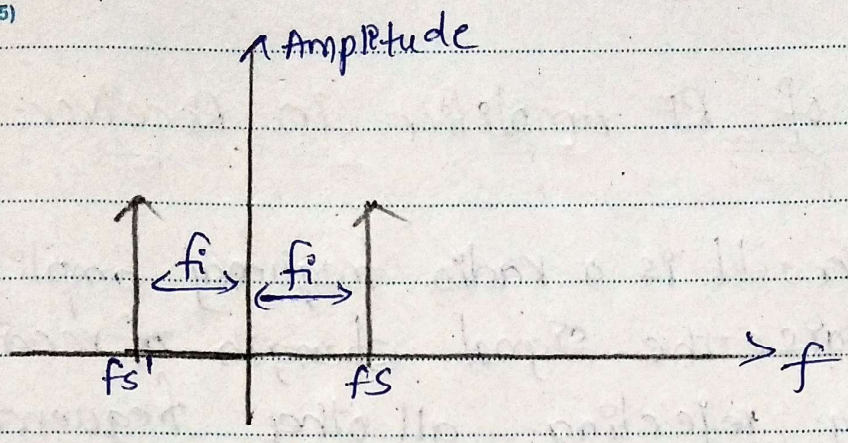
power Amplifier: It is used to raise the signal strength of received signal from detector and given to speaker (ie it Receives originally generated Received signal.

For the use of Evaluator Only

Q.No	Marks

Q.No	Marks

Q.No	Marks



$fs = f_L - fi$

$fs' = f_L + f_R$

$fs' = fs + fi + fi$

$$\boxed{fs' = fs + 2fi}$$

2fi is the intermediate frequency, if it is more than modulating signal, it also received by receiver which leads to distortion.

→ IF Amplifier generate Image frequency.

(i.e.) The frequency of the above (desired signal fs ^ local oscillator frequency) same as below the local oscillator frequency.

Image Rejection Ratio $\rho = \sqrt{1+Q^2\rho^2}$

$$\rho = \frac{fs'}{fs} - \frac{fs}{fs'}$$

For the use of Evaluator Only

Q.No	Marks
2(a)	8

Q.No	Marks

Q.No	Marks

(ESTD-1995)

2b) Functions of RF Amplifier in Receiver:

RF Amplifier: It is a radio frequency amplifier used to raise the signal strength of received signal by rejecting all other frequencies.

→ It receives and amplify the voltage levels/re powes for better transmission to load.

→ It provides high gain.

→ It provides (SNR)

→ It provides gain adjacent to local oscillator frequency.

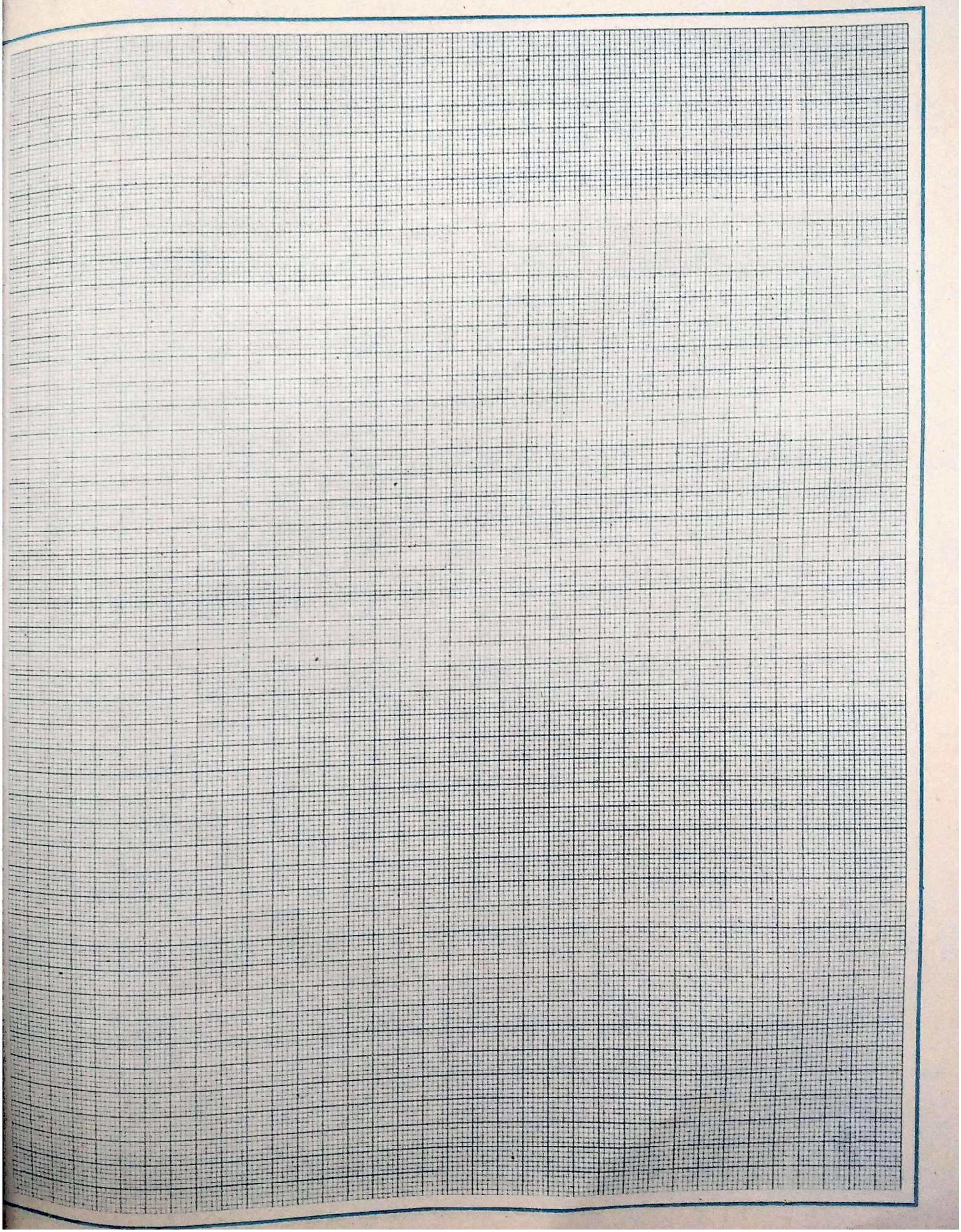
→ It provides high Image Rejection ratio for the received signal.

For the use of Evaluator Only

Q.No	Marks
2b)	1

Q.No	Marks

Q.No	Marks



CO-PO Attainment Process

Academic Year: 2022-23 Regulations: R20

Year & Sem: III-I Batch: 20

Branch: ECE Subject: AC

Name of the Faculty: D.Rajesh Setty

External Question Paper Marks --> Cos							
Q.No.	CO 1	CO 2	CO 3	CO 4	CO 5	CO 6	Total
1 a)	2						2
b)	2						2
c)		2					2
d)	2						2
e)				2			2
f)		2					2
g)	2						2
2 a)				9			9
b)				5			5
c)							0
3 a)	7						7
b)	7						7
c)							0
4 a)		9					9
b)			5				5
c)							0
5 a)	10						10
b)	4						4
c)							0
6 a)		10					10
b)		4					4
c)							0
7 a)	9						9
b)	5						5
c)							0
Total	50	27	5	16	0	0	98

Mid I Marks --> Cos							
Q.No.	CO 1	CO 2	CO 3	CO 4	CO 5	CO 6	Total
1 a)		1					1
b)			1				1
c)		1					1
d)	1						1
e)			1				1
2 a)	3						3
b)				2			2
3 a)	3						3
b)		2					2
4 a)				3			3
b)		2					2
5 a)			3				3
b)		2					2
Total	7	8	5	5	0	0	25

Mid II Marks --> Cos							
Q.No.	CO 1	CO 2	CO 3	CO 4	CO 5	CO 6	Total
1 a)	1						1
b)		1					1
c)			1				1
d)				1			1
e)					1		1
2 a)	2						2
b)			3				3
3 a)		3					3
b)					2		2
4 a)			3				3
b)		2					2
5 a)					3		3
b)				2			2
Total	3	6	7	3	6	0	25

Total	10	14	12	8	6	0	50

Assignment 1 Marks --> Cos							
	CO 1	CO2	CO3	CO4	CO5	CO6	Total
	1	1	1	1	1		5
Total	1	1	1	1	1	0	5

Assignment 2 Marks --> Cos							
	CO1	CO2	CO3	CO4	CO5	CO6	Total
	1	1	1	1	1		5
Total	1	1	1	1	1	0	5

Total	2	2	2	2	2	0	10

S.No	Reg.No.	Internal marks	Assignment marks	Final Internal marks	Total Final Marks	External Marks	% of IM	% of AM	% of EM	N CO 1	N CO 2	N CO 3	N CO 4	N CO 5	N CO 6
1	20091A0402	23	5	28	82	54	115	50	77.1	79.47	83.01	89.43	81.16	85.45	0
2	20091A0404	23	5	28	77	49	115	50	70	73.36	77.89	86.97	76.25	85.45	0
4	20091A0408	23	5	28	82	54	115	50	77.1	79.47	83.01	89.43	81.16	85.45	0
5	20091A0419	24	5	29	78	49	120	50	70	73.84	78.93	89.28	77.22	88.18	0
6	20091A0435	24	5	29	82	53	120	50	75.7	78.73	83.03	91.25	81.14	88.18	0
7	20091A0436	24	5	29	82	53	120	50	75.7	78.73	83.03	91.25	81.14	88.18	0
8	20091A0437	21	5	26	68	42	105	50	60	63.84	68.63	78.9	67.46	80	0
9	20091A0439	24	5	29	78	49	120	50	70	73.84	78.93	89.28	77.22	88.18	0
10	20091A0450	23	5	28	76	48	115	50	68.6	72.13	76.86	86.48	75.27	85.45	0
11	20091A0466	24	5	29	78	49	120	50	70	73.84	78.93	89.28	77.22	88.18	0
12	20091A0469	24	5	29	86	57	120	50	81.4	83.62	87.12	93.22	85.07	88.18	0
13	20091A0470	24	5	29	73	44	120	50	62.9	67.72	73.8	86.82	72.31	88.18	0
14	20091A0471	24	5	29	82	53	120	50	75.7	78.73	83.03	91.25	81.14	88.18	0
15	20091A0472	24	5	29	72	43	120	50	61.4	66.5	72.78	86.33	71.33	88.18	0
16	20091A0475	24	5	29	89	60	120	50	85.7	87.29	90.2	94.7	88.02	88.18	0
17	20091A0478	24	5	29	85	56	120	50	80	82.4	86.1	92.73	84.09	88.18	0
18	20091A0479	23	5	28	74	46	115	50	65.7	69.69	74.81	85.49	73.31	85.45	0
19	20091A0480	22	5	27	84	57	110	50	81.4	82.66	85.04	88.59	83.15	82.73	0
20	20091A0488	24	5	29	75	46	120	50	65.7	70.17	75.85	87.81	74.27	88.18	0
21	20091A0490	22	5	27	65	38	110	50	54.3	59.42	65.57	79.24	64.49	82.73	0
22	20091A0497	21	5	26	79	53	105	50	75.7	77.29	79.9	84.31	78.26	80	0
23	20091A04A3	24	5	29	78	49	120	50	70	73.84	78.93	89.28	77.22	88.18	0
24	20091A04C1	24	5	29	81	52	120	50	74.3	77.5	82	90.76	80.16	88.18	0
25	20091A04C4	24	5	29	80	51	120	50	72.9	76.28	80.98	90.27	79.18	88.18	0
26	20091A04C9	24	5	29	79	50	120	50	71.4	75.06	79.95	89.78	78.2	88.18	0
27	20091A04D3	21	5	26	72	46	105	50	65.7	68.73	72.73	80.87	71.38	80	0
28	20091A04D7	22	5	27	77	50	110	50	71.4	74.1	77.87	85.15	76.27	82.73	0
29	20091A04D9	24	5	29	73	44	120	50	62.9	67.72	73.8	86.82	72.31	88.18	0
30	20091A04E0	24	5	29	74	45	120	50	64.3	68.94	74.83	87.32	73.29	88.18	0
31	20091A04E1	23	5	28	77	49	115	50	70	73.36	77.89	86.97	76.25	85.45	0

32	20091A04E5	18	5	23	66	43	90	50	61.4	63.62	66.53	72.45	65.55	71.82	0
33	20091A04E6	23	5	28	74	46	115	50	65.7	69.69	74.81	85.49	73.31	85.45	0
34	20091A04E7	22	5	27	75	48	110	50	68.6	71.65	75.82	84.16	74.31	82.73	0
35	20091A04E8	22	5	27	77	50	110	50	71.4	74.1	77.87	85.15	76.27	82.73	0
36	20091A04E9	23	5	28	73	45	115	50	64.3	68.46	73.79	85	72.33	85.45	0
37	20091A04F1	22	5	27	69	42	110	50	60	64.32	69.67	81.21	68.42	82.73	0
38	20091A04F8	24	5	29	76	47	120	50	67.1	71.39	76.88	88.3	75.25	88.18	0
39	20091A04F9	23	5	28	74	46	115	50	65.7	69.69	74.81	85.49	73.31	85.45	0
40	20091A04G0	24	5	29	83	54	120	50	77.1	79.95	84.05	91.74	82.13	88.18	0
41	20091A04G1	19	5	24	52	28	95	50	40	45.75	52.2	67.38	51.79	74.55	0
42	20091A04G2	23	5	28	74	46	115	50	65.7	69.69	74.81	85.49	73.31	85.45	0
43	20091A04G3	24	5	29	75	46	120	50	65.7	70.17	75.85	87.81	74.27	88.18	0
44	20091A04G5	23	5	28	69	41	115	50	58.6	63.57	69.69	83.03	68.4	85.45	0
45	20091A04H5	24	5	29	74	45	120	50	64.3	68.94	74.83	87.32	73.29	88.18	0
46	20091A04H7	24	5	29	71	42	120	50	60	65.27	71.75	85.84	70.34	88.18	0
47	20091A04H8	24	5	29	71	42	120	50	60	65.27	71.75	85.84	70.34	88.18	0
48	20091A04H9	24	5	29	70	41	120	50	58.6	64.05	70.73	85.35	69.36	88.18	0
49	20091A04J0	24	5	29	83	54	120	50	77.1	79.95	84.05	91.74	82.13	88.18	0
50	20091A04J3	23	5	28	85	57	115	50	81.4	83.14	86.08	90.91	84.11	85.45	0
51	20091A04J9	21	5	26	74	48	105	50	68.6	71.17	74.78	81.85	73.35	80	0
52	20091A04K1	18	5	23	64	41	90	50	58.6	61.17	64.48	71.46	63.59	71.82	0
53	20091A04K5	22	5	27	70	43	110	50	61.4	65.54	70.7	81.7	69.4	82.73	0
54	20091A04K6	24	5	29	88	59	120	50	84.3	86.07	89.17	94.2	87.03	88.18	0
55	20091A04M3	24	5	29	86	57	120	50	81.4	83.62	87.12	93.22	85.07	88.18	0
56	20091A04M7	22	5	27	78	51	110	50	72.9	75.32	78.89	85.64	77.26	82.73	0
57	20091A04M9	24	5	29	74	45	120	50	64.3	68.94	74.83	87.32	73.29	88.18	0
58	20091A04N6	22	5	27	67	40	110	50	57.1	61.87	67.62	80.23	66.46	82.73	0
59	20091A04N7	24	5	29	78	49	120	50	70	73.84	78.93	89.28	77.22	88.18	0
60	20091A04P0	22	5	27	69	42	110	50	60	64.32	69.67	81.21	68.42	82.73	0
61	20091A04P8	22	5	27	84	57	110	50	81.4	82.66	85.04	88.59	83.15	82.73	0
62	21095A0404	24	5	29	82	53	120	50	75.7	78.73	83.03	91.25	81.14	88.18	0
63	21095A0412	24	5	29	82	53	120	50	75.7	78.73	83.03	91.25	81.14	88.18	0
64	21095A0417	23	5	28	79	51	115	50	72.9	75.8	79.93	87.95	78.22	85.45	0
65	21095A0430	22	5	27	85	58	110	50	82.9	83.88	86.07	89.08	84.13	82.73	0
66	21095A0432	23	5	28	88	60	115	50	85.7	86.81	89.16	92.38	87.05	85.45	0
67	18091A04A3	18	5	23	71	48	90	50	68.6	69.74	71.65	74.91	70.46	71.82	0
68	19091A0404	16	5	21	81	60	80	50	85.7	83.45	81.87	76.18	80.32	66.36	0
69	19091A0410	18	5	23	80	57	90	50	81.4	80.74	80.88	79.33	79.3	71.82	0
70	19091A0417	20	5	25	70	45	100	50	64.3	67.03	70.66	78.06	69.44	77.27	0
71	19091A0419	17	5	22	87	65	85	50	92.9	90.05	88.03	80.96	86.19	69.09	0
72	19091A0424	24	5	29	84	55	120	50	78.6	81.17	85.08	92.24	83.11	88.18	0
73	19091A0426	20	5	25	81	56	100	50	80	80.48	81.93	83.47	80.24	77.27	0
74	19091A0428	21	5	26	71	45	105	50	64.3	67.5	71.7	80.37	70.4	80	0
75	19091A0431	19	5	24	65	41	95	50	58.6	61.65	65.52	73.78	64.55	74.55	0
76	19091A0432	23	5	28	79	51	115	50	72.9	75.8	79.93	87.95	78.22	85.45	0
77	19091A0437	20	5	25	86	61	100	50	87.1	86.59	87.06	85.93	85.15	77.27	0
78	19091A0446	22	5	27	79	52	110	50	74.3	76.55	79.92	86.13	78.24	82.73	0
79	19091A0454	20	5	25	84	59	100	50	84.3	84.15	85.01	84.95	83.19	77.27	0
81	19091A0455	23	5	28	77	49	115	50	70	! ▪	77.89	86.97	76.25	85.45	0
82	19091A0456	18	5	23	71	48	90	50	68.6	69.74	71.65	74.91	70.46	71.82	0
83	19091A0460	23	5	28	93	65	115	50	92.9	92.93	94.28	94.84	91.96	85.45	0
84	19091A0462	20	5	25	63	38	100	50	54.3	58.46	63.49	74.62	62.57	77.27	0

85	19091A0464	21	5	26	64	38	105	50	54.3	58.94	64.53	76.93	63.53	80	0
86	19091A0467	20	5	25	89	64	100	50	91.4	90.26	90.13	87.41	88.1	77.27	0
87	19091A0469	21	5	26	82	56	105	50	80	80.96	82.98	85.79	81.2	80	0
88	19091A0479	22	5	27	78	51	110	50	72.9	75.32	78.89	85.64	77.26	82.73	0
89	19091A0480	19	5	24	65	41	95	50	58.6	61.65	65.52	73.78	64.55	74.55	0
90	19091A0481	22	5	27	61	34	110	50	48.6	54.53	61.47	77.28	60.56	82.73	0
91	19091A0490	20	5	25	76	51	100	50	72.9	74.36	76.81	81.01	75.33	77.27	0
92	19091A0491	20	5	25	61	36	100	50	51.4	56.02	61.44	73.63	60.6	77.27	0
93	19091A0495	19	5	24	60	36	95	50	51.4	55.54	60.4	71.32	59.64	74.55	0
94	19091A0498	22	5	27	84	57	110	50	81.4	82.66	85.04	88.59	83.15	82.73	0
95	19091A04A4	20	5	25	82	57	100	50	81.4	81.7	82.96	83.96	81.22	77.27	0
96	19091A04A6	21	5	26	86	60	105	50	85.7	85.85	87.07	87.75	85.13	80	0
97	19091A04A7	21	5	26	75	49	105	50	70	72.4	75.8	82.34	74.33	80	0
98	19091A04A9	21	5	26	76	50	105	50	71.4	73.62	76.83	82.83	75.31	80	0
99	19091A04B1	22	5	27	86	59	110	50	84.3	85.11	87.09	89.57	85.11	82.73	0
100	19091A04B4	22	5	27	83	56	110	50	80	81.44	84.02	88.1	82.16	82.73	0
101	19091A04B9	19	5	24	73	49	95	50	70	71.44	73.72	77.71	72.41	74.55	0
102	19091A04C2	22	5	27	95	68	110	50	97.1	96.12	96.31	94	93.95	82.73	0
103	19091A04C4	24	5	29	78	49	120	50	70	73.84	78.93	89.28	77.22	88.18	0
104	19091A04C5	19	5	24	85	61	95	50	87.1	86.12	86.02	83.62	84.19	74.55	0
105	19091A04C8	24	5	29	86	57	120	50	81.4	83.62	87.12	93.22	85.07	88.18	0
106	19091A04D2	15	5	20	66	46	75	50	65.7	65.85	66.48	66.98	65.61	63.64	0
107	19091A04D8	18	5	23	75	52	90	50	74.3	74.63	75.75	76.88	74.39	71.82	0
108	19091A04E0	20	5	25	76	51	100	50	72.9	74.36	76.81	81.01	75.33	77.27	0
109	19091A04E7	21	5	26	75	49	105	50	70	72.4	75.8	82.34	74.33	80	0
110	19091A04E9	18	5	23	76	53	90	50	75.7	75.85	76.78	77.37	75.37	71.82	0
111	19091A04F0	21	5	26	82	56	105	50	80	80.96	82.98	85.79	81.2	80	0
112	19091A04F6	22	5	27	68	41	110	50	58.6	63.09	68.65	80.72	67.44	82.73	0
113	19091A04F8	22	5	27	74	47	110	50	67.1	70.43	74.79	83.67	73.33	82.73	0
114	19091A04G0	19	5	24	70	46	95	50	65.7	67.77	70.65	76.24	69.46	74.55	0
115	19091A04G9	18	5	23	74	51	90	50	72.9	73.41	74.73	76.38	73.41	71.82	0
116	19091A04H2	21	5	26	56	30	105	50	42.9	49.16	56.33	72.99	55.68	80	0
117	19091A04H4	19	5	24	75	51	95	50	72.9	73.88	75.77	78.7	74.37	74.55	0
118	19091A04J2	19	5	24	76	52	95	50	74.3	75.11	76.79	79.19	75.35	74.55	0
119	19091A04K2	21	5	26	92	66	105	50	94.3	93.19	93.22	90.7	91.02	80	0
120	19091A04K6	20	5	25	71	46	100	50	65.7	68.25	71.69	78.55	70.42	77.27	0
121	19091A04M1	21	5	26	76	50	105	50	71.4	73.62	76.83	82.83	75.31	80	0
122	19091A04M8	20	5	25	77	52	100	50	74.3	75.59	77.84	81.5	76.31	77.27	0
123	19091A04N3	21	5	26	66	40	105	50	57.1	61.39	66.58	77.91	65.49	80	0
124	19091A04N5	20	5	25	71	46	100	50	65.7	68.25	71.69	78.55	70.42	77.27	0
125	19091A04N7	18	5	23	84	61	90	50	87.1	85.64	84.97	81.3	83.23	71.82	0
126	19091A04P1	21	5	26	77	51	105	50	72.9	74.84	77.85	83.33	76.29	80	0
127	19091A04P8	21	5	26	78	52	105	50	74.3	76.07	78.88	83.82	77.28	80	0
128	19091A04Q2	22	5	27	89	62	110	50	88.6	88.78	90.17	91.05	88.06	82.73	0
129	19091A04Q3	20	5	25	77	52	100	50	74.3	75.59	77.84	81.5	76.31	77.27	0
130	19091A04R2	20	5	25	90	65	100	50	92.9	91.49	91.16	87.9	89.08	77.27	0
131	19091A04R4	22	5	27	90	63	110	50	90	90	91.19	91.54	89.04	82.73	0
132	19091A04S1	21	5	26	78	52	105	50	74.3	76.07	78.88	83.82	77.28	80	0
133	20095A0403	23	5	28	81	53	115	50	75.7	78.25	81.98	88.94	80.18	85.45	0
134	20095A0410	24	5	29	82	53	120	50	75.7	78.73	83.03	91.25	81.14	88.18	0
135	20095A0413	15	5	20	70	50	75	50	71.4	70.74	70.58	68.95	69.54	63.64	0
136	20095A0415	23	5	28	89	61	115	50	87.1	88.03	90.18	92.87	88.04	85.45	0

137	20095A0422	19	5	24	75	51	95	50	72.9	73.88	75.77	78.7	74.37	74.55	0
138	18091A0485	15	5	20	67	47	75	50	67.1	67.07	67.5	67.47	66.59	63.64	0
139	19091A0409	17	5	22	66	44	85	50	62.9	64.36	66.51	70.63	65.57	69.09	0
141	19091A0418	17	5	22	68	46	85	50	65.7	66.81	68.56	71.61	67.54	69.09	0
142	19091A0422	14	5	19	80	61	70	50	87.1	83.72	80.81	72.05	79.38	60.91	0
143	19091A0430	20	5	25	66	41	100	50	58.6	62.13	66.56	76.09	65.51	77.27	0
144	19091A0438	19	5	24	70	46	95	50	65.7	67.77	70.65	76.24	69.46	74.55	0
145	19091A0440	22	5	27	89	62	110	50	88.6	88.78	90.17	91.05	88.06	82.73	0
146	19091A0443	17	5	22	71	49	85	50	70	70.48	71.64	73.09	70.48	69.09	0
147	19091A0448	19	5	24	91	67	95	50	95.7	93.45	92.16	86.57	90.08	74.55	0
148	19091A0457	19	5	24	78	54	95	50	77.1	77.55	78.84	80.17	77.31	74.55	0
149	19091A0468	17	5	22	67	45	85	50	64.3	65.59	67.54	71.12	66.55	69.09	0
150	19091A0470	20	5	25	83	58	100	50	82.9	82.93	83.98	84.45	82.2	77.27	0
151	19091A0472	20	5	25	81	56	100	50	80	80.48	81.93	83.47	80.24	77.27	0
152	19091A0473	14	5	19	52	33	70	50	47.1	49.47	52.12	58.27	51.89	60.91	0
153	19091A0476	15	5	20	52	32	75	50	45.7	48.73	52.13	60.09	51.87	63.64	0
154	19091A0477	20	5	25	67	42	100	50	60	63.36	67.59	76.58	66.49	77.27	0
155	19091A0483	16	5	21	66	45	80	50	64.3	65.11	66.5	68.8	65.59	66.36	0
156	19091A0487	16	5	21	54	33	80	50	47.1	50.43	54.2	62.9	53.81	66.36	0
157	19091A0494	18	5	23	66	43	90	50	61.4	63.62	66.53	72.45	65.55	71.82	0
158	19091A0497	14	5	19	58	39	70	50	55.7	56.81	58.26	61.22	57.78	60.91	0
159	19091A0499	21	5	26	61	35	105	50	50	55.27	61.46	75.45	60.58	80	0
160	19091A04A2	18	5	23	70	47	90	50	67.1	68.51	70.63	74.42	69.48	71.82	0
161	19091A04A8	17	5	22	73	51	85	50	72.9	72.93	73.69	74.07	72.44	69.09	0
162	19091A04B0	18	5	23	69	46	90	50	65.7	67.29	69.6	73.92	68.5	71.82	0
163	19091A04B2	20	5	25	83	58	100	50	82.9	82.93	83.98	84.45	82.2	77.27	0
164	19091A04B8	21	5	26	55	29	105	50	41.4	47.94	55.31	72.5	54.69	80	0
165	19091A04C0	15	5	20	48	28	75	50	40	43.84	48.03	58.13	47.94	63.64	0
166	19091A04D0	18	5	23	78	55	90	50	78.6	78.3	78.83	78.35	77.33	71.82	0
167	19091A04D1	13	5	18	52	34	65	50	48.6	50.22	52.1	56.45	51.9	58.18	0
168	19091A04D7	17	5	22	57	35	85	50	50	53.36	57.29	66.2	56.74	69.09	0
169	19091A04E2	19	5	24	67	43	95	50	61.4	64.1	67.57	74.76	66.51	74.55	0
170	19091A04E6	15	5	20	66	46	75	50	65.7	65.85	66.48	66.98	65.61	63.64	0
171	19091A04F3	20	5	25	83	58	100	50	82.9	82.93	83.98	84.45	82.2	77.27	0
172	19091A04F5	15	5	20	50	30	75	50	42.9	46.28	50.08	59.11	49.9	63.64	0
173	19091A04G3	18	5	23	80	57	90	50	81.4	80.74	80.88	79.33	79.3	71.82	0
174	19091A04G4	17	5	22	66	44	85	50	62.9	64.36	66.51	70.63	65.57	69.09	0
175	19091A04H5	20	5	25	73	48	100	50	68.6	70.69	73.74	79.54	72.39	77.27	0
176	19091A04H7	19	5	24	61	37	95	50	52.9	56.76	61.42	71.81	60.62	74.55	0
177	19091A04H8	16	5	21	61	40	80	50	57.1	58.99	61.37	66.34	60.68	66.36	0
179	19091A04J5	19	5	24	79	55	95	50	78.6	78.78	79.87	80.67	78.3	74.55	0
180	19091A04J6	13	5	18	47	29	65	50	41.4	44.1	46.98	53.99	47	58.18	0
181	19091A04J7	16	5	21	71	50	80	50	71.4	71.22	71.62	71.26	70.5	66.36	0
182	19091A04J8	20	5	25	71	46	100	50	65.7	68.25	71.69	78.55	70.42	77.27	0
183	19091A04K1	19	5	24	83	59	95	50	84.3	83.67	83.97	82.63	82.22	74.55	0
184	19091A04K4	16	5	21	66	45	80	50	64.3	65.11	66.5	68.8	65.59	66.36	0
185	19091A04M0	17	5	22	74	52	85	50	74.3	74.15	74.71	74.56	73.43	69.09	0
186	19091A04M4	18	5	23	78	55	90	50	78.6	78.3	78.83	78.35	77.33	71.82	0
187	19091A04N1	13	5	18	64	46	65	50	65.7	64.89	64.4	62.35	63.69	58.18	0
188	19091A04P2	15	5	20	74	54	75	50	77.1	75.64	74.68	70.92	73.47	63.64	0
189	19091A04P3	17	5	22	62	40	85	50	57.1	59.47	62.41	68.66	61.64	69.09	0
190	19091A04P7	18	5	23	67	44	90	50	62.9	64.84	67.55	72.94	66.53	71.82	0

191	19091A04Q0	16	5	21	69	48	80	50	68.6	68.78	69.57	70.28	68.54	66.36	0
192	19091A04Q4	19	5	24	80	56	95	50	80	80	80.89	81.16	79.28	74.55	0
193	19091A04Q5	18	5	23	74	51	90	50	72.9	73.41	74.73	76.38	73.41	71.82	0
195	19091A04R0	17	5	22	63	41	85	50	58.6	60.69	63.44	69.15	62.63	69.09	0
196	19091A04R5	18	5	23	85	62	90	50	88.6	86.86	86	81.79	84.21	71.82	0
197	19091A04R6	16	5	21	74	53	80	50	75.7	74.89	74.69	72.74	73.45	66.36	0
199	19091A04R7	16	5	21	80	59	80	50	84.3	82.23	80.84	75.69	79.34	66.36	0
200	19091A04S3	20	5	25	72	47	100	50	67.1	69.47	72.71	79.04	71.4	77.27	0
201	19091A04S4	14	5	19	69	50	70	50	71.4	70.26	69.54	66.64	68.58	60.91	0
202	20095A0405	21	5	26	72	46	105	50	65.7	68.73	72.73	80.87	71.38	80	0
203	20095A0406	24	5	29	72	43	120	50	61.4	66.5	72.78	86.33	71.33	88.18	0
204	20095A0408	21	5	26	86	60	105	50	85.7	85.85	87.07	87.75	85.13	80	0
205	20095A0409	18	5	23	80	57	90	50	81.4	80.74	80.88	79.33	79.3	71.82	0
206	20095A0412	18	5	23	56	33	90	50	47.1	51.39	56.28	67.53	55.73	71.82	0
207	20095A0416	20	5	25	74	49	100	50	70	71.92	74.76	80.03	73.37	77.27	0
208	20095A0417	18	5	23	76	53	90	50	75.7	75.85	76.78	77.37	75.37	71.82	0
209	20095A0419	21	5	26	84	58	105	50	82.9	83.41	85.03	86.77	83.17	80	0
210	20095A0425	24	5	29	92	63	120	50	90	90.96	93.27	96.17	90.96	88.18	0
213	19091A0401	10	5	15	61	46	50	50	65.7	63.45	61.27	55.41	60.8	50	0
214	19091A0402	10	5	15	66	51	50	50	72.9	69.57	66.4	57.87	65.71	50	0
215	19091A0403	11	5	16	41	25	55	50	35.7	38.25	40.79	47.39	41.14	52.73	0
216	19091A0420	14	5	19	68	49	70	50	70	69.04	68.51	66.14	67.59	60.91	0
217	19091A0425	9	5	14	68	54	45	50	77.1	72.76	68.43	57.03	67.69	47.27	0
218	19091A0427	17	5	22	76	54	85	50	77.1	76.59	76.76	75.55	75.39	69.09	0
219	19091A0435	13	5	18	65	47	65	50	67.1	66.12	65.42	62.85	64.67	58.18	0
220	19091A0439	17	5	22	52	30	85	50	42.9	47.24	52.17	63.74	51.83	69.09	0
221	19091A0445	17	5	22	90	68	85	50	97.1	93.72	91.11	82.43	89.14	69.09	0
222	19091A0447	15	5	20	75	55	75	50	78.6	76.86	75.7	71.41	74.45	63.64	0
223	19091A0449	20	5	25	75	50	100	50	71.4	73.14	75.79	80.52	74.35	77.27	0
224	19091A0453	11	5	16	56	40	55	50	57.1	56.59	56.17	54.77	55.87	52.73	0
225	19091A0458	16	5	21	72	51	80	50	72.9	72.45	72.64	71.76	71.48	66.36	0
226	19091A0461	14	5	19	51	32	70	50	45.7	48.25	51.09	57.78	50.9	60.91	0
227	19091A0465	17	5	22	73	51	85	50	72.9	72.93	73.69	74.07	72.44	69.09	0
228	19091A0475	17	5	22	80	58	85	50	82.9	81.49	80.86	77.51	79.32	69.09	0
229	19091A0484	16	5	21	56	35	80	50	50	52.88	56.25	63.88	55.77	66.36	0
230	19091A0492	13	5	18	28	10	65	50	14.3	20.86	27.51	44.64	28.34	58.18	0
231	19091A0493	16	5	21	63	42	80	50	60	61.44	63.42	67.33	62.65	66.36	0
232	19091A0496	11	5	16	44	28	55	50	40	41.92	43.87	48.87	44.09	52.73	0
233	19091A04A3	12	5	17	46	29	60	50	41.4	43.62	45.93	51.68	46.03	55.45	0
234	19091A04B7	15	5	20	64	44	75	50	62.9	63.41	64.43	66	63.65	63.64	0
235	19091A04C3	20	5	25	65	40	100	50	57.1	60.91	65.54	75.6	64.53	77.27	0
236	19091A04C7	13	5	18	79	61	65	50	87.1	83.24	79.77	69.73	78.41	58.18	0
237	19091A04D3	21	5	26	62	36	105	50	51.4	56.5	62.48	75.95	61.57	80	0
238	19091A04D4	20	5	25	62	37	100	50	52.9	57.24	62.46	74.12	61.59	77.27	0
239	19091A04D5	22	5	27	70	43	110	50	61.4	65.54	70.7	81.7	69.4	82.73	0
240	19091A04F2	15	5	20	61	41	75	50	58.6	59.74	61.36	64.52	60.7	63.64	0
241	19091A04F4	13	5	18	51	33	65	50	47.1	48.99	51.07	55.96	50.92	58.18	0
242	19091A04F7	22	5	27	95	68	110	50	97.1	96.12	96.31	94	93.95	82.73	0
243	19091A04G5	17	5	22	68	46	85	50	65.7	66.81	68.56	71.61	67.54	69.09	0
244	19091A04G6	16	5	21	66	45	80	50	64.3	65.11	66.5	68.8	65.59	66.36	0
245	19091A04H0	13	5	18	49	31	65	50	44.3	46.55	49.03	54.97	48.96	58.18	0
246	19091A04H1	15	5	20	57	37	75	50	52.9	54.84	57.26	62.55	56.77	63.64	0

247	19091A04H6	12	5	17	60	43	60	50	61.4	60.74	60.28	58.56	59.78	55.45	0
248	19091A04H9	11	5	16	57	41	55	50	58.6	57.82	57.19	55.27	56.85	52.73	0
249	19091A04J0	12	5	17	34	17	60	50	24.3	28.94	33.64	45.77	34.25	55.45	0
250	19091A04J1	13	5	18	65	47	65	50	67.1	66.12	65.42	62.85	64.67	58.18	0
251	19091A04J3	15	5	20	50	30	75	50	42.9	46.28	50.08	59.11	49.9	63.64	0
252	19091A04J9	15	5	20	51	31	75	50	44.3	47.5	51.11	59.6	50.88	63.64	0
253	19091A04K3	20	5	25	66	41	100	50	58.6	62.13	66.56	76.09	65.51	77.27	0
254	19091A04K9	16	5	21	62	41	80	50	58.6	60.22	62.4	66.84	61.66	66.36	0
255	19091A04M6	16	5	21	50	29	80	50	41.4	45.54	50.1	60.93	49.88	66.36	0
256	19091A04M7	15	5	20	74	54	75	50	77.1	75.64	74.68	70.92	73.47	63.64	0
257	19091A04N0	15	5	20	70	50	75	50	71.4	70.74	70.58	68.95	69.54	63.64	0
258	19091A04N8	20	5	25	76	51	100	50	72.9	74.36	76.81	81.01	75.33	77.27	0
259	19091A04P5	13	5	18	59	41	65	50	58.6	58.78	59.27	59.89	58.78	58.18	0
260	19091A04P6	11	5	16	49	33	55	50	47.1	48.03	48.99	51.33	49	52.73	0

R20 Regulation	Weightage of Marks	
Final exam	Internal tests	Ass/Quiz
70	20	10
0.7	0.2	0.1

Weightage of marks for each CO

	CO 1	CO 2	CO 3	CO 4	CO 5	CO 6	Total
EM	50	27	5	16	0	0	98
IM	10	14	12	8	6	0	50
AM	2	2	2	2	2	0	10

% Weightage of each CO

	CO 1	CO 2	CO 3	CO 4	CO 5	CO 6	Total
EM (%)	51.0204	27.551	5.10204	16.3265	0	0	100
IM (%)	20	28	24	16	12	0	100
AM (%)	20	20	20	20	20	0	100
Avg	41.7143	26.8857	10.3714	16.6286	4.4	0	100

	CO 1		CO 2		CO 3		CO 4		CO 5		CO 6	
	No. of students Attained	Weightage Points	No. of students Attained	Weightage Points	No. of students Attained	Weightage Points	No. of students Attained	Weightage Points	No. of students Attained	Weightage Points	No. of students Attained	Weightage Points
>= 50%	231	3	243	3	248	3	240	3	251	3	0	3
40% to 50%	18	2	7	2	4	2	10	2	1	2	0	2
<40%	3	1	2	1	0	1	2	1	0	1	0	1

Total No. of students	252		252		252		252		252		0	
Attainment value		2.90		2.96		2.98		2.94		3.00		0.00
% of Attainment		91.67		96.43		98.41		95.24		99.60		0.00
Attained or not		YES		YES		YES		YES		YES		0.00

III Year Threshold
50

CO	CO Attainment Value	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12	PSO1	PSO2	PSO3
CO 1	2.90	3	0	0	0	0	0	0	0	0	0	0	0	0	2	
CO 2	2.96	2	3	1	0	0	0	0	0	0	0	0	0	1	0	1
CO 3	2.98	3	3	2	0	0	0	0	0	0	0	0	0	0	2	0
CO 4	2.94	0	3	3	2	0	0	0	0	0	0	0	0	2	0	3
CO 5	3.00	2	2	3	2	0	0	0	0	0	0	0	0	0	0	
CO 6	0.00	0	0	0	0	0	0	0	0	0	0	0	0			
AC		2.96	2.97	2.97	2.97	-	-	-	-	-	-	-	-	2.95	2.94	2.95

6-5 SINGLE- AND INDEPENDENT-SIDEBAND RECEIVERS

Single- and independent-sideband receivers are normally used for professional or commercial communications. There are of course also a lot of amateur SSB receivers, but this section will concentrate on the professional applications. Such receivers are almost invariably required to detect signals in difficult conditions and crowded frequency bands. Consequently, they are always multiple-conversion receivers, of a pattern similar to, or developed from, the block shown in Figure 6-17. The special requirements of SSB and ISB receivers are:

1. High reliability (and simple maintenance), since such receivers may be operated continuously
2. Excellent suppression of adjacent signals
3. Ability to demodulate SSB
4. Good blocking performance
5. High signal-to-noise ratio
6. Ability to separate the independent channels (in the case of ISB receivers)

The specialized aspects of SSB and ISB receivers will now be investigated.

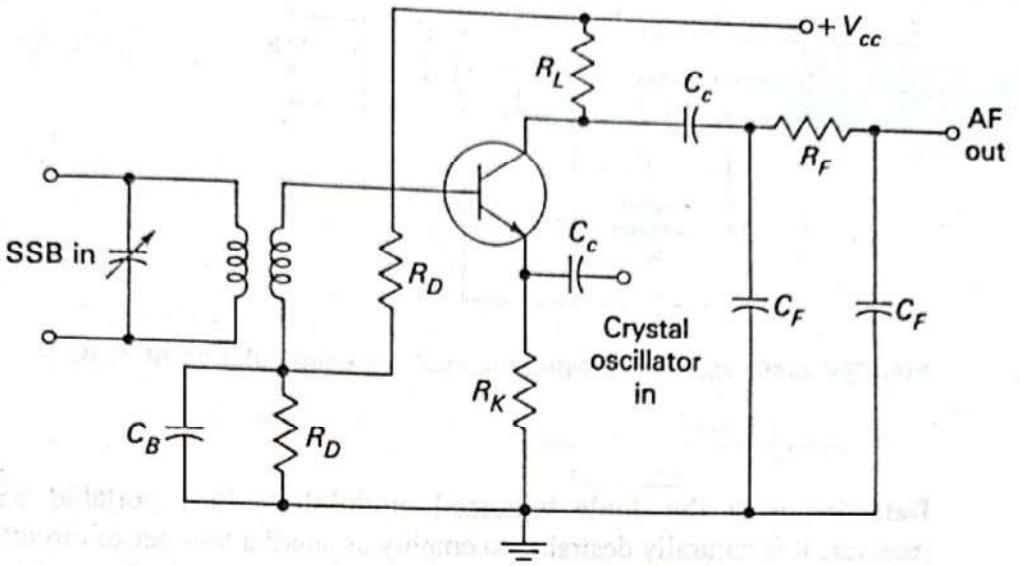


FIGURE 6-44 Product demodulator.

6-5.1 Demodulation of SSB

Demodulation of SSB must obviously be different from ordinary AM detection. The basic SSB demodulation device is the *product detector*, which is rather similar to an ordinary mixer. The balanced modulator is almost always used in transceivers, in which it is important to utilize as many circuits as possible for dual purposes. It is also possible to demodulate SSB with the complete phase-shift network. The complete third-method system can similarly be used for demodulation.

Product demodulator The product demodulator (or detector), as shown in Figure 6-44, is virtually a mixer with audio output. It is popular for SSB, but is equally capable of demodulating all other forms of AM. In the circuit shown, the input SSB signal is fed to the base via a fixed-frequency IF transformer, and the signal from a crystal oscillator is applied to the unbypassed emitter. The frequency of this oscillator is either equal to the nominal carrier frequency or derived from the pilot frequency, as applicable.

If this is a fairly standard double-conversion receiver, like the one shown in Figure 6-47, the IF fed to the product detector will be 455 kHz. If the USB is being received, the signal will cover the frequency band from 455.3 to 458.0 kHz for J3E (in R3E 455 kHz would also be present at this point). This signal is mixed with the output of the crystal oscillator, at 455 kHz. Several frequencies will result in the output, including the difference frequencies (as covered in Section 4-2.1). These range from 300 to 3000 Hz and are the wanted audio frequencies. All other signals present at this point will be blocked by the low-pass filter consisting of capacitors C_F and resistor R_F in Figure 6-44. The circuit has recovered the wanted intelligence from the input signal and is therefore a suitable SSB demodulator.

If the lower sideband is being received, the missing carrier frequency is at 458 kHz, and the sideband stretches from 457.7 to 455 kHz. A new crystal must be switched in for the oscillator, but apart from that, the operation is identical.

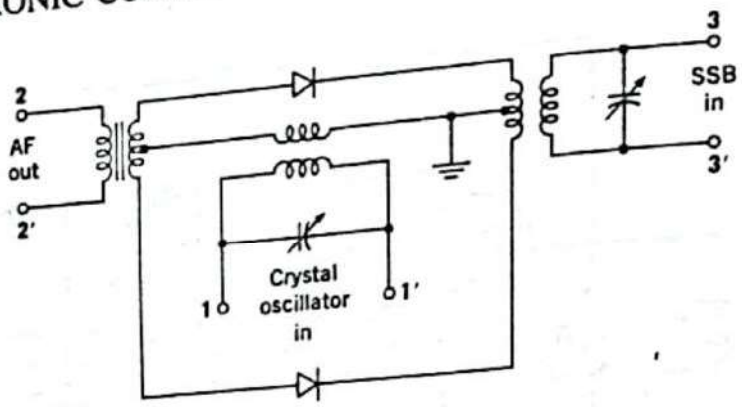


FIGURE 6-45 Balanced modulator used for demodulation of SSB.

Detection with the diode balanced modulator In a portable SSB transmitter-receiver, it is naturally desirable to employ as small a number of circuits as possible to save weight and power consumption. If a particular circuit is capable of performing either function, it is always so used, with the aid of appropriate switching when changing from transmission to reception. Since the diode balanced modulator can demodulate SSB, it is used for that purpose in transceivers, in preference to the product demodulator. A circuit of the balanced modulator is shown in Figure 6-45; it is identical to the one in Figure 4-3*a*, but the emphasis here is on demodulation.

As in carrier suppression, the output of the local crystal oscillator, having the same frequency as in the product detector (200 or 203 kHz, depending on the sideband being demodulated), is fed to the terminals 1–1′. Where the carrier-suppressed signal was taken from the modulator at terminals 3–3′, the SSB signal is now fed in. The balanced modulator now operates as a nonlinear resistance and, as in the product detector, sum and difference frequencies appear at the primary winding of the AF transformer. This transformer will not pass radio frequencies and therefore acts as a low-pass filter, delivering only the audio frequencies to the terminals 2–2′, which have now become the output terminals of the demodulator. It is seen that this circuit recovers the information from the SSB signal, as required, and works very similarly to the product demodulator.

6-5.2 Receiver Types

We shall describe a pilot-carrier receiver and a suppressed-carrier receiver; the suppressed carrier receiver incorporates a frequency synthesizer for extra stability and also is used to show how ISB may be demodulated.

Pilot-carrier receiver As shown in Figure 6-46, in block form, a pilot-carrier receiver is a fairly straightforward communications receiver with trimmings. It uses double conversion, and AFC based on the pilot carrier. AFC is needed to ensure good frequency stability, which must be at least 1 part of 10^7 (long-term) for long-distance telephone and telegraph communications. Note also the use of one local crystal oscillator, with multiplication by 9, rather than two separate oscillators; this also improves stability.

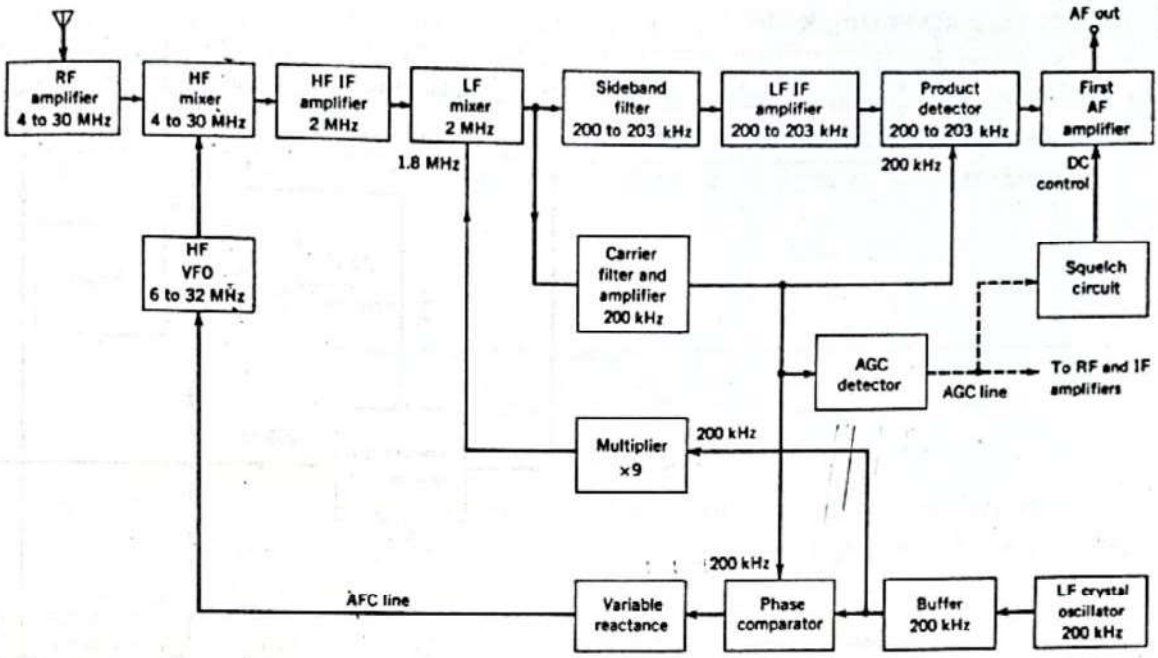


FIGURE 6-46 Block diagram of pilot-carrier single-sideband receiver.

The output of the second mixer contains two components—the wanted sideband and the weak carrier. They are separated by filters, the sideband going to the product detector, and the carrier to AGC and AFC circuits via an extremely narrowband filter and amplifier. The output of the carrier amplifier is fed, together with the buffered output of the crystal oscillator, to a *phase comparator*. This is almost identical to the phase discriminator and works in a similar fashion. The output depends on the *phase difference* between the two applied signals, which is zero or a positive or negative dc voltage, just as in the discriminator. The phase difference between the two inputs to the phase-sensitive circuit can be zero only if the frequency difference is zero. Excellent frequency stability is obtainable. The output of the phase comparator actuates a varactor diode connected across the tank circuit of the VFO and pulls it into frequency as required.

Because a pilot carrier is transmitted, automatic gain control is not much of a problem, although that part of the circuit may look complicated. The output of the carrier filter and amplifier is a carrier whose amplitude varies with the strength of the input signal, so that it may be used for AGC after rectification. Automatic gain control is also applied to the squelch circuit, as explained in Section 6-3.2 It should also be mentioned that receivers of this type often have AGC with two different time constants. This is helpful in telegraphy reception, and in coping to a certain extent with signal-strength variations caused by fading.

Suppressed-carrier receiver A typical block diagram is shown in Figure 6-47. This is actually a much simplified version of the receiver of Figure 6-18, which is capable of receiving all forms of AM but has here been shown in the ISB mode. The receiver has a number of very interesting features, of which the first is the fixed-frequency RF

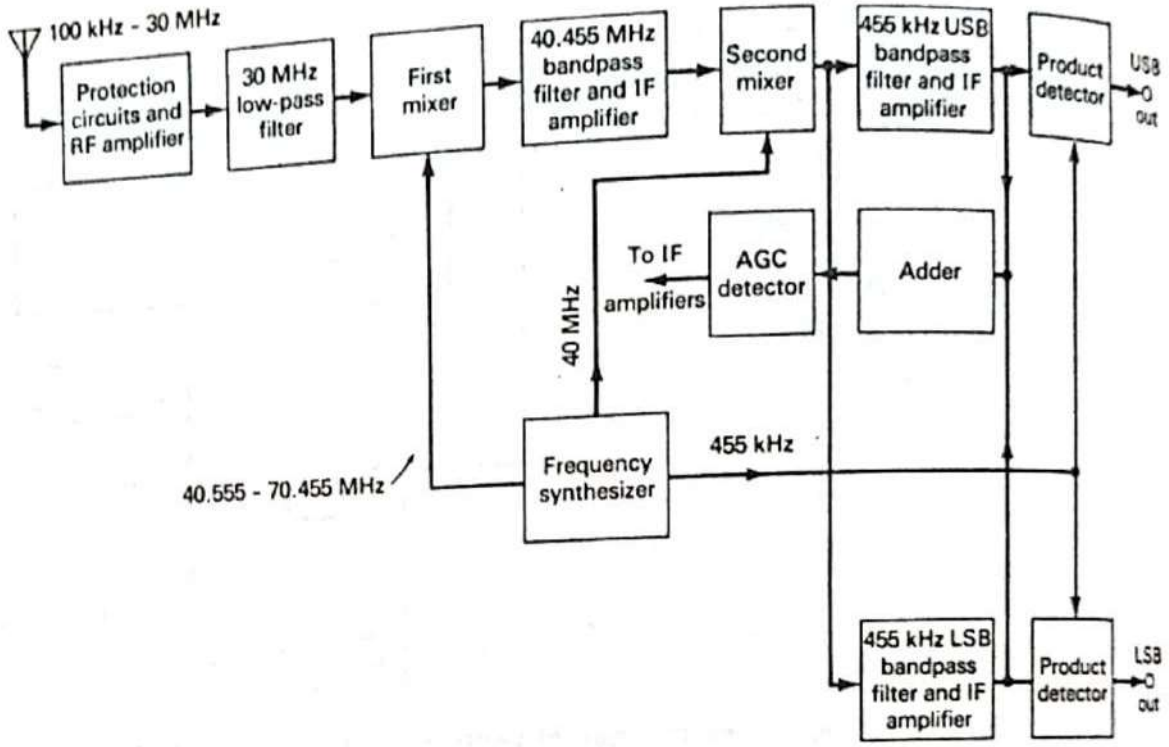


FIGURE 6-47 ISB receiver with frequency synthesizer. ***(This is a simplified block diagram of the RA 1792 receiver in the ISB mode, adapted by permission of Racal Electronics Pty. Ltd.)***

amplifier. This may be wideband, covering the entire 100-kHZ to 30-MHz receiving range; or, optionally, a set of filters may be used, each covering a portion of this range. The second very interesting feature is the very high first intermediate frequency, 40.455 MHz. Such high frequencies have been made possible by the advent of VHF crystal bandpass filters. They are increasingly used by SSB receivers, for a number of reasons. One, clearly, is to provide image frequency rejections much higher than previously available. Another reason is to facilitate receiver tuning. In the RA 1792, which is typical of high-quality professional receivers, a variety of tuning methods are available, such as push-button selection, or even automatic selection of a series of wanted preset channels stored in the microprocessor memory. However, an important method is the orthodox continuous tuning method, which utilizes a tuning knob. Since receivers of this type are capable of remote tuning, the knob actually adjusts the voltage applied to a varactor diode across the VFO in an indirect frequency synthesizer. There is a limit to the tuning range. If the first IF is high, the resulting range (70.455 MHz ÷ 40.555 MHz = 1.74:1) can be covered in a single sweep, with a much lower first IF it cannot be tuned so easily.

It will be seen that this is, nonetheless, a double-conversion superheterodyne receiver, up to the low-frequency IF stages. After this the main differences are due to the presence of the two independent sidebands, which are separated at this point with mechanical filters. If just a single upper and a single lower sideband are transmitted, the USB filter will have a bandpass of 455.25 to 458 kHz, and the LSB filter 452 to 454.75 kHz. Since the carrier is not transmitted, it is necessary to obtain AGC by rectifying part of the combined audio signal. From this a dc voltage proportional to the

average audio level is obtained. This requires an AGC circuit time constant of sufficient length to ensure that AGC is not proportional to the instantaneous audio voltage. Because of the presence of the frequency synthesizer, the frequency stability of such a receiver can be very high. For example, one of the frequency standard options of the RA 1792 will give a long-term frequency stability of 3 parts in 10^9 per day.

Rajeev Gandhi Memorial College of Engineering and Technology, Nandyal
(Autonomous)
Department of Electronics and Communication Engineering

Academic Year: 2022-23

SUBJECT: AC

Class: III-B.Tech; I-Sem

Faculty members taught earlier: N.Fouzia Sulthana, S.V.Ratan Kumar

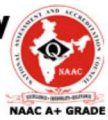
Suggestions:

As per the course structure of the UG (R20), AC is a subject based on signals and systems, communication systems along with numerical problems. Students of II-B.Tech have already studied signals and systems. Hence, following suggestions are made by subject experts.

1. The subject contains theory and Problems. So, it is decided to explicate more revisions and solve number of problems from different text books.
2. To improve the results, it is decided to take care of slow learners, and practice previous question papers as assignments.
3. Regular counseling of the students should be done to follow the lacking in knowledge
4. The environment of the classroom should be friendly, so that they can ask any doubts continuously.

S. V. Ratan Kumar
10/11/20
Signature of the Subject Expert

Rajeev Gandhi Memorial College of Engineering & Technology
(Autonomous Institution | Approved by AICTE New Delhi | Affiliated to JNTU Anantapur)
Address : Nerawada 'X' Roads,Nandyal,Kurnool Dist,Andhra Pradesh,India
Counselling Code : RGIT



D Rajesh Setty
Asst. Prof.
Dept. of ECE
Phone: 9441021792
Email: rsduddela@gmail.com

S.No.	Year (Batch)	Section	Subject	Branch	Registered	Appeared	Failed	Pass(%)	Highest
1	III B.Tech. II Sem. & 2020	A	DC Lab	ECE	65	64	0	100.00	10
2	III B.Tech. II Sem. & 2020	D	DC	ECE	63	57	30	47.37	6
3	III B.Tech. II Sem. & 2020	A	DC	ECE	65	64	1	98.44	10
4	III B.Tech. I Sem. & 2020	C	AC Lab	ECE	61	60	0	100.00	10
5	III B.Tech. I Sem. & 2020	C	AC	ECE	61	60	1	98.33	9
6	III B.Tech. I Sem. & 2020	B	AC	ECE	65	65	0	100.00	9

Copyright © 2021 RGMCET. All rights reserved.

Developed by Suresh Babu J