Dept. Of Electronics & Communication Engineering

Analog Communications Lab Manual (S/W) (III ECE – I Sem)

Balaji Institute of Technology & Science

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List of Experiments

- **1. Amplitude Modulation & Demodulation**
- 2. DSB-SC Modulation & Demodulation
- 3. SSB-SC Modulation & Demodulation
- 4. Frequency Division Multiplexing & Demultiplexing
- 5. Frequency Modulation & Demodulation
- 6. PLL as FM Demodulator
- 7. Spectral Characteristics of AM &FM
- 8. Verification of Sampling Theorem
- 9. Pulse Amplitude Modulation & Demodulation
- **10.Time Division Multiplexing & Demultiplexing**
- **11.Pulse Width Modulation**
- **12.Pulse Position Modulation & Demodulation**

<u>1.Amplitude Modulation & Demodulation</u>

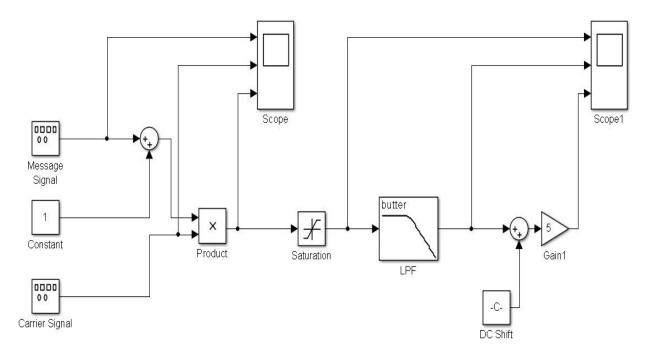
Aim

To study the function of Amplitude Modul**a**tion & Demodulation (Under modulation, Perfect modulation & Over modulation) using Matlab Simulink.

Apparatus Required

- a) Ha**r**dware Tools: Computer system
- b) Software Tool: MATLAB 7.0 and above version

Simulink model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon 🖣. Go to file and select new and then select model. You will get a new window.

- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

Parameters

Under Modulation

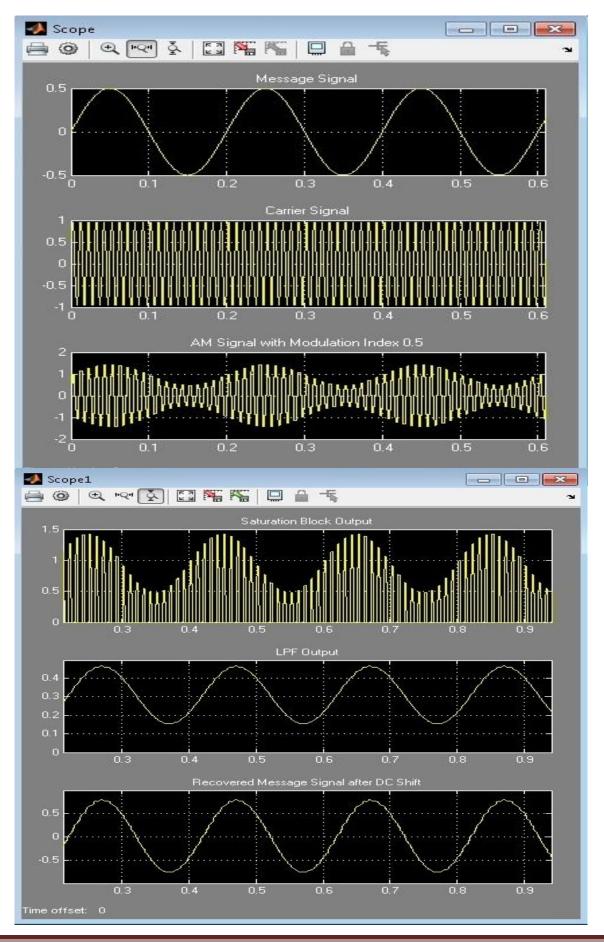
🚡 Source Block Parameters: Message Signal 🛛 💦	🔁 Source Block Parameters: Carrier Signal 🛛 💽
Signal Generator	Signal Generator
Output various wave forms: Y(t) = Amp*Waveform(Freq, t)	Output various wave forms: Y(t) = Amp*Waveform(Freq, t)
Parameters	Parameters
Wave form: sine	Wave form: sine
Time (t): Use simulation time	Time (t): Use simulation time 🔹
Amplitude:	Amplitude:
0.5	1
Frequency:	Frequency:
5	100
Units: Hertz 🔹	Units: Hertz
☑ Interpret vector parameters as 1-D	🗹 Interpret vector parameters as 1-D
OK Cancel Help Apply	OK Cancel Help Apply

🚡 Function Block Parameters: Sum 🛛	x	Source Block Parameters: Constant
Sum	-	Constant
Add or subtract inputs. Specify one of the following: a) string containing + or - for each input port, $ $ for spacer between ports (e.g. ++ - ++) b) scalar, >= 1, specifies the number of input ports to be summed. When there is only one input port, add or subtract elements over all dimensions or one specified dimension		Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value.
	E	Main Signal Attributes
Main Signal Attributes		Constant value:
Icon shape: round	-	
List of signs:		☑ Interpret vector parameters as 1-D
1++		Sampling mode: Sample based
Sample time (-1 for inherited):		
0.001	-	Sample time:
Conservations.	-	inf
۲	F	
OK Cancel Help Appl	y	OK Cancel Help Apply

🔁 Function Block Parameters: Product	
Product	Function Block Parameters: Saturation
Multiply or divide inputs. Choose element-wise or matrix product and	Saturation
specify one of the following: a) * or / for each input port. For example, **/* performs the	Limit input signal to the upper and lower saturation values.
operation 'u1*u2/u3*u4'. b) scalar specifies the number of input ports to be multiplied.	
If there is only one input port and the Multiplication parameter is set to	Main Signal Attributes
Element-wise(.*), a single * or / collapses the input signal using the specified operation. However, if the Multiplication parameter is set to	Upper limit:
Matrix(*), a single * causes the block to output the matrix unchanged, and a single / causes the block to output the matrix inverse.	3
Main Signal Attributes	Lower limit:
Number of inputs:	0
2	
Multiplication: Element-wise(.*)	Treat as gain when linearizing
Sample time (-1 for inherited):	Enable zero-crossing detection
0.001	Sample time (-1 for inherited):
	0.001
🚡 Function Block Parameters: LPF 🛛 💦	🔁 Source Block Parameters: DC Shift 📃
Analog Filter Design (mask) (link)	Constant
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- space form.	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is
Design one of several standard analog filters, implemented in state- space form.	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix
Design one of several standard analog filters, implemented in state-	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is
Design one of several standard analog filters, implemented in state- space form.	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix
Design one of several standard analog filters, implemented in state- space form.	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value.
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass Filter order:	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes Constant value:
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass Filter order: 3	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes Constant value: -0.3055 Interpret vector parameters as 1-D
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass Filter order:	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes Constant value: -0.3055 Interpret vector parameters as 1-D Sampling mode: Sample based *
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass Filter order: 3	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes Constant value: -0.3055 Interpret vector parameters as 1-D
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass Filter order: 3 Passband edge frequency (rad/s):	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes Constant value: -0.3055 Interpret vector parameters as 1-D Sampling mode: Sample based *
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass Filter order: 3 Passband edge frequency (rad/s):	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes Constant value: -0.3055 Interpret vector parameters as 1-D Sampling mode: Sample based Sample time:
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass Filter order: 3 Passband edge frequency (rad/s):	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes Constant value: -0.3055 Interpret vector parameters as 1-D Sampling mode: Sample based * Sample time: D.001
Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Lowpass Filter order: 3 Passband edge frequency (rad/s): 100	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value. Main Signal Attributes Constant value: -0.3055 Interpret vector parameters as 1-D Sampling mode: Sample based * Sample time: 0.001

vlain ain:	Signal Attributes	Parameter Attributes
5 5		
6 18 1	ation: Element-wise	
).001		

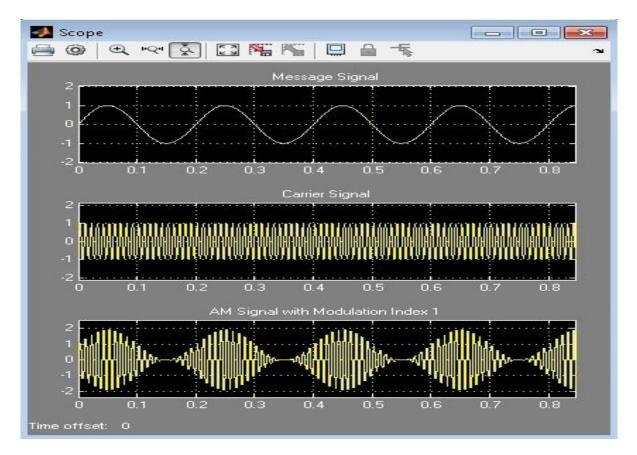
Output

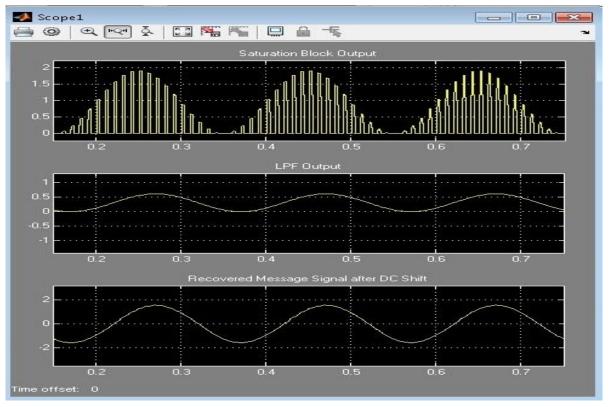


100% Modulation

Change Message signal amplitude to 1Volt.

Change DC Shift to -0.31

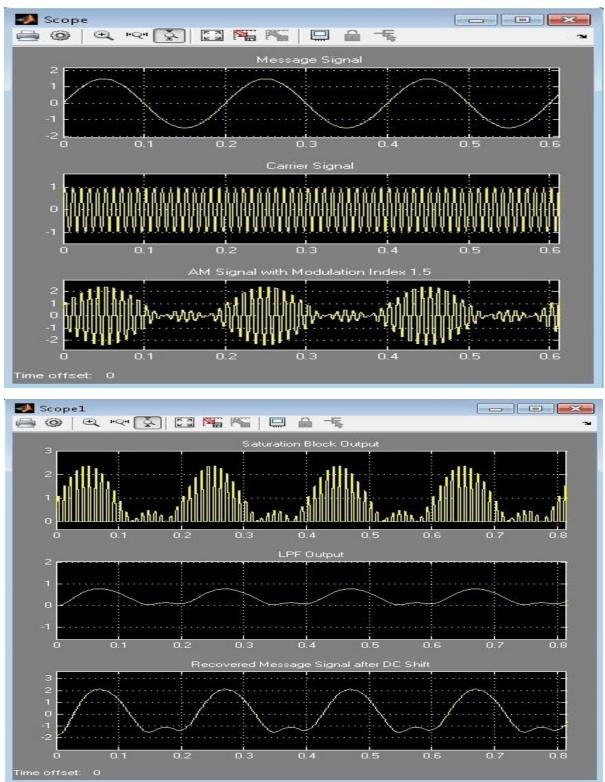




Over Modulation:

Change Message signal amplitude to 1Volt.

Change DC Shift to -0.35



Result

Viva Questions

- 1. Define AM and draw its spectrum?
- 2. Draw the phasor representation of an amplitude modulated wave?
- 3. Give the significance of modulation index?
- 4. What are the different degrees of modulation?
- 5. What are the limitations of square law modulator?
- 6. Compare linear and nonlinear modulators?
- 7. Compare base modulation and emitter modulation?
- 8. Explain how AM wave is detected?
- 9. Define detection process?

10. What are the different types of distortions that occur in an envelop detector? How can they be eliminated?

- 11. What is the condition of for over modulation?
- 12. Define modulation & demodulation?
- 13. What are the different types of linear modulation techniques?
- 14. Explain the working of carrier wave generator.
- 15. Explain the working of modulator circuit.

Applications

- 1. Radio Broadcasting.
- 2. Picture transmission in a TV system.
- 3. AM is used in computer modems, VHF aircraft radio, and in portable two-way radio
- 4. Used to carry message signals in early telephone lines.
- 5. Used to transmit Morse code using radio and other communication systems.
- 6. Used in Navy and Aviation for communications as AM signals can travel longer distances.

Advantages of Amplitude Modulation:

- Because of amplitude modulation wavelength, AM signals can propagate longer distances.
- For amplitude modulation, we use simple and low cost circuit; we don't need any special equipment and complex circuits that are used in frequency modulation.
- The Amplitude modulation receiver will be wider when compared to the FM receiver. Because, atmospheric propagation is good for amplitude modulated signals.
- Bandwidths limit is also big advantage for Amplitude modulation, which doesn't have in frequency modulation.
- Transmitter and receiver are simple in Amplitude modulation. When we take a demodulation unit of AM receiver, it consists of RC filter and a diode which will demodulate the message signal or modulating signal from modulated AM signal, which is unlike in Frequency modulation.

• Zero crossing in Amplitude modulation is equidistant.

Disadvantages of Amplitude Modulation:

- Adding of noise for amplitude modulated signal will be more when compared to frequency modulated signals. Data loss is also more in amplitude modulation due to noise addition. Demodulators cannot reproduce the exact message signal or modulating signal due to noise.
- More power is required during modulation because Amplitude modulated signal frequency should be double than modulating signal or message signal frequency. Due to this reason more power is required for amplitude modulation.
- Sidebands are also transmitted during the transmission of carrier signal. More chances of getting different signal interfaces and adding of noise is more when compared to frequency modulation. Noise addition and signal interferences are less for frequency modulation. That is why Amplitude modulation is not used for broadcasting songs or music.

2. AM-DSB-SC Modulation & Demodulation

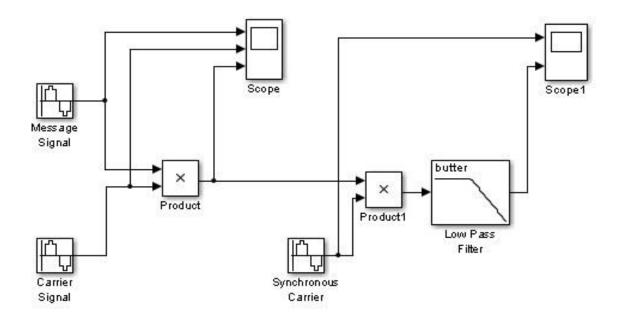
Aim

To perform the AM DSB-SC signal Generation and Detection using Matlab Simulink.

Apparatus Required

- a) Hardware Tools: Computer system
- b) Software Tool: MATLAB 7 or Upgraded version

Simulink Model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

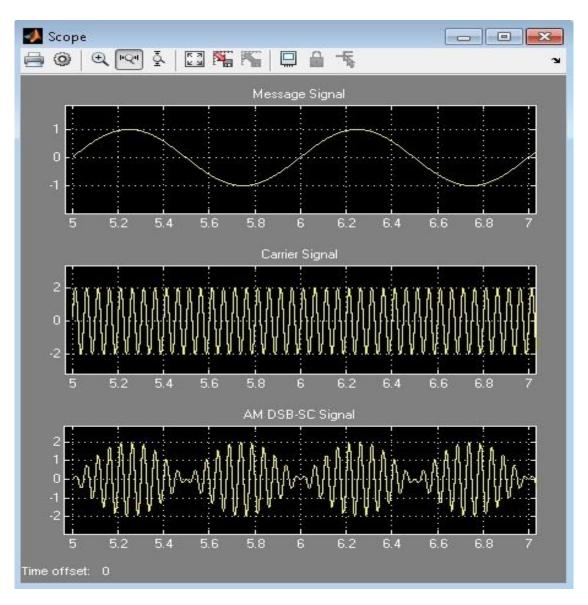
Parameters

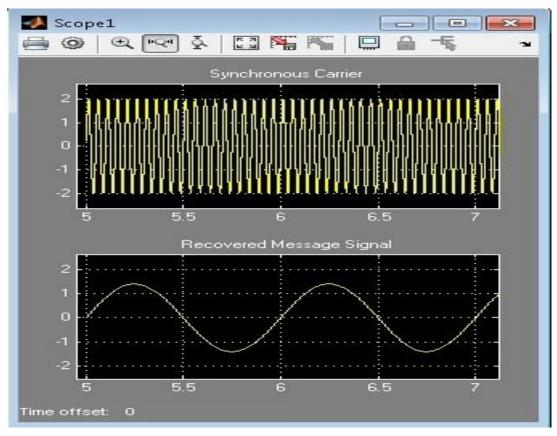
🚡 Source Block Parameters: Message Signal 💽	🔁 Source Block Parameters: Carrier Signal 🛛 🕰
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.	Use the sample-based sine type if numerical problems due to runnin <u>c</u> for large times (e.g. overflow in absolute time) occur.
Parameters	Parameters
Sine type: Time based	Sine type: Time based 🔹
Time (t): Use simulation time 🔹	Time (t): Use simulation time 🔹
Amplitude:	Amplitude:
1	2
Bias:	Bias:
0	0
Frequency (rad/sec):	Frequency (rad/sec):
2*pi*1 =	2*pi*20 =
Phase (rad):	Phase (rad):
0	0
Sample time:	Sample time:
0.001	0.001
☑ Interpret vector parameters as 1-D	☑ Interpret vector parameters as 1-D
* •	* *
OK Cancel Help Apply	OK Cancel Help Apply

🚹 Function Block Parameters: Product	📔 Source Block Parameters: Synchronous Carrier 📃 🔀
Product Multiply or divide inputs. Choose element-wise or matrix product and	Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
specify one of the following: a) * or / for each input port. For example, **/* performs the operation 'u1*u2/u3*u4'. b) scalar specifies the number of input ports to be multiplied. If there is only one input port and the Multiplication parameter is set to Element-wise(.*), a single * or / collapses the input signal using the	Parameters Sine type: Time based Time (t): Use simulation time
specified operation. However, if the Multiplication parameter is set to	Amplitude:
Matrix(*), a single * causes the block to output the matrix unchanged, and a single / causes the block to output the matrix inverse.	2
	Bias:
Main Signal Attributes	0
Number of inputs:	Frequency (rad/sec):
2	2*pi*20 ≡
Multiplication: Element-wise(,*)	Phase (rad):
Sample time (-1 for inherited):	0
0.001	Sample time:
	0.001
	☑ Interpret vector parameters as 1-D
	4 <u> </u>
OK Cancel Help Apply	OK Cancel Help Apply

Function Block Parameters: Low Pass Filter Analog Filter Design (mask) (link)
Design one of several standard analog filters, implemented in state- space form.
Parameters
Design method: Butterworth
Filter type: Lowpass
Filter order:
8
Passband edge frequency (rad/s):
2*pi*1
OK Cancel Help Apply

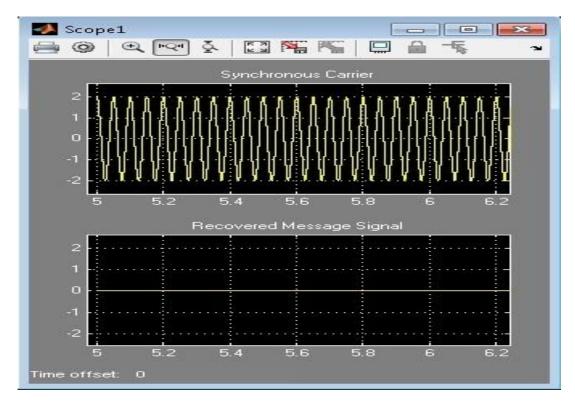
Output





Quadrature Null Effect

• Change Synchronous carrier Signal Phase(rad) to pi/2



Result

Viva Questions

- 1. What are the two ways of generating DSB_SC?
- 2. What are the applications of balanced modulator?
- 3. What are the advantages of suppressing the carrier?
- 4. What are the advantages of balanced modulator?
- 5. What are the advantages of Ring modulator?
- 6. Write the expression for the output voltage of a balanced modulator?
- 7. Explain the working of balanced modulator and Ring Modulator using diodes.

Applications

- 1. Analog TV systems to transmit colour information.
- 2. Used in Point-to-Point communication.
- 3. Used in applications where power requirements are low.
- 4. Used in Linear Radio Communication

3. AM-SSB-SC Modulation & Demodulation

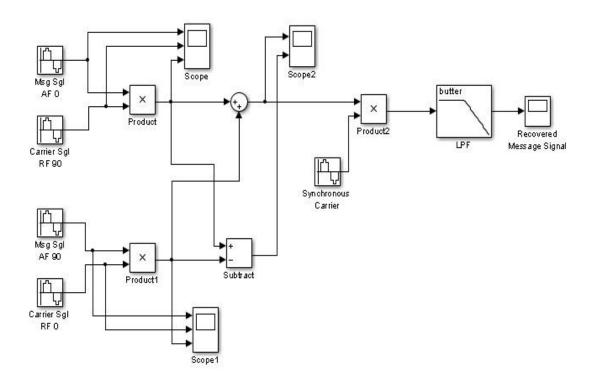
Aim

To perform the AM SSB-SC signal Generation and Detection using Matlab Simulink.

Apparatus Required

- a) Hardware Tools: Computer system
- b) Software Tool: MATLAB 7.0 and above version

Simulink Model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon **i**. Go to file and select new and then select model. You will get a new window.

- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

Parameters

🜇 Source Block Parameters: Msg Sgl AF 0 🛛 👔	x	🔊 Source Block Parameters: Carrier Sgl RF 90
Use the sample-based sine type if numerical problems due to runni for large times (e.g. overflow in absolute time) occur.	nç	Sine Wave Output a sine wave:
Parameters		O(t) = Amp*Sin(Freq*t+Phase) + Bias
Sine type: Time based		Sine type determines the computational technique used. The parameters in the two types are related through:
Time (t): Use simulation time		Samples per period = 2*pi / (Frequency * Sample time)
Amplitude:		Number of offset samples = Phase * Samples per period / (2*pi)
Bias:	_	Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
0 Frequency (rad/sec):		Parameters
2*pi*2	=	Sine type: Time based
Phase (rad):		Time (t): Use simulation time
0		Amplitude:
Sample time:		1
0.001		Bias:
☑ Interpret vector parameters as 1-D		0
۲ (m)		Frenilency (rad/sec).
OK Cancel Help Apply		OK Cancel Help Apply

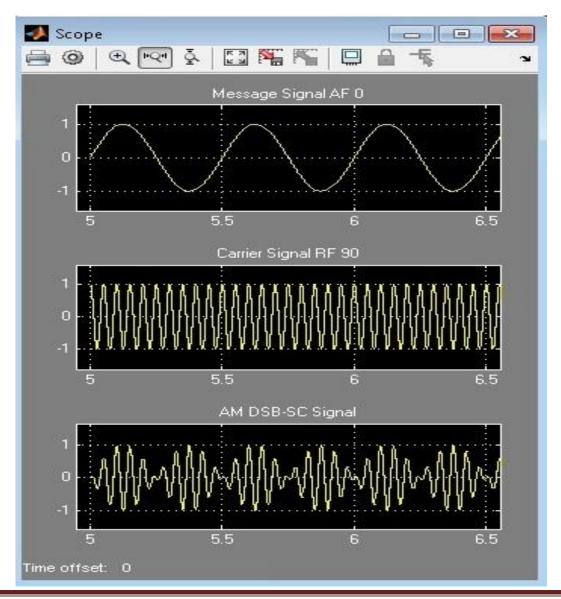
🜇 Source Block Parameters: Msg Sgl AF 90	X	🚹 Source Block Parameters: Carrier Sgl RF 0 🛛 🔜
Use the sample-based sine type if numerical problems due to runni for large times (e.g. overflow in absolute time) occur.	nç	Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
Parameters		Parameters
Sine type: Time based	-	Sine type: Time based
Time (t): Use simulation time	-	Time (t): Use simulation time 🔹
Amplitude:		Amplitude:
1		1
Bias:		Bias:
0		0
Frequency (rad/sec):		Frequency (rad/sec):
2*pi*2	ш	2*pi*20 ≡
Phase (rad):		Phase (rad):
pi/2		0
Sample time:		Sample time:
0.001		0.001
☑ Interpret vector parameters as 1-D		☑ Interpret vector parameters as 1-D
۲ []	P.	۰
OK Cancel Help Appl		OK Cancel Help Apply

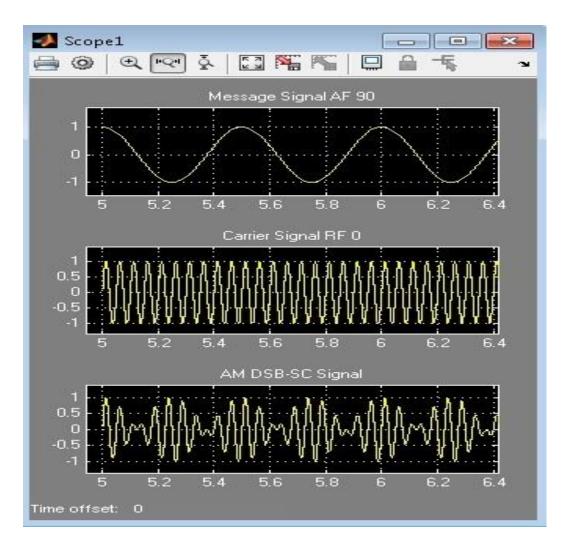
Function Block Parameters: Product	📲 Function Block Parameters: Sum
Product	Sum
Multiply or divide inputs. Choose element-wise or matrix product and specify one of the following: a) * or / for each input port. For example, **/* performs the operation 'u1*u2/u3*u4'. b) scalar specifies the number of input ports to be multiplied. If there is only one input port and the Multiplication parameter is set to Element-wise(.*), a single * or / collapses the input signal using the specified operation. However, if the Multiplication parameter is set to Matrix(*), a single * causes the block to output the matrix unchanged, and a single / causes the block to output the matrix inverse.	Add or subtract inputs. Specify one of the following: a) string containing + or - for each input port, for spacer between ports (e.g. ++ - ++) b) scalar, >= 1, specifies the number of input ports to be summed. When there is only one input port, add or subtract elements over all dimensions or one specified dimension
Main Signal Attributes	Main Signal Attributes
Number of inputs:	Icon shape: round
2	List of signs:
Multiplication: Element-wise(.*)	1++
Sample time (-1 for inherited):	Sample time (-1 for inherited):
0.001	0.001
	1 G38/20/6487
	۲
OK Cancel Help Apply	OK Cancel Help Apply

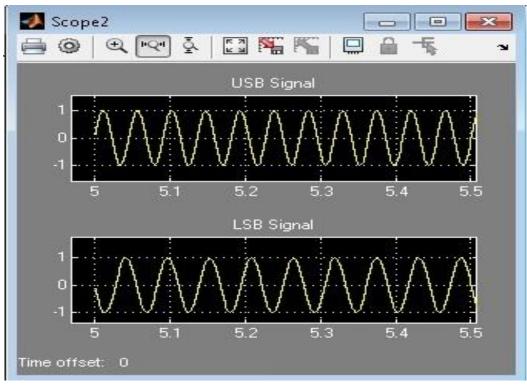
🔁 Function Block Parameters: Subtract	Source Block Parameters: Synchronous Carrier
Sum Add or subtract inputs. Specify one of the following:	Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
 a) string containing + or - for each input port, for spacer between ports (e.g. ++ - ++) b) scalar, >= 1, specifies the number of input ports to be summed. 	Parameters Sine type: Time based
When there is only one input port, add or subtract elements over all dimensions or one specified dimension	Time (t): Use simulation time
	Amplitude:
Main Signal Attributes	1
Icon shape: rectangular 🔹	Bias:
List of signs:	0
+-	Frequency (rad/sec):
Constanting (1 for intervited).	2*pi*20 =
Sample time (-1 for inherited):	Phase (rad):
0.001	0
	Sample time:
	0.001
	☑ Interpret vector parameters as 1-D
	* •
OK Cancel Help Apply	OK Cancel Help Apply

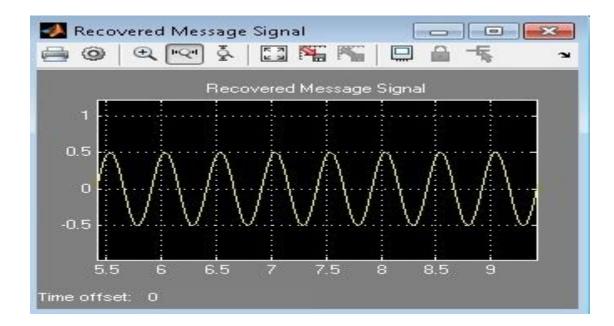
🔁 Function Block Parameters: LPF 🛛 🔀
Analog Filter Design (mask) (link)
Design one of several standard analog filters, implemented in state- space form.
Parameters
Design method: Butterworth
Filter type: Lowpass
Filter order:
8
Passband edge frequency (rad/s):
2*pi*20
OK Cancel Help Apply

Output









Result

Viva Questions

- 1. What are the different methods to generate SSB-SC signal?
- 2. What is the advantage of SSB-SC over DSB-SC?
- 3. Explain Phase Shift method for SSB generation.
- 4. Why SSB is not used for broadcasting?
- 5. Give the circuit for synchronous detector?
- 6. What are the uses of synchronous or coherent detector?
- 7. Give the block diagram of synchronous detector?
- 8. Why the name synchronous detector?

Applications

- 1. SSB transmission is used in applications where the Power saving is required in Mobile systems.
- 2. SSB is also used in applications in which bandwidth requirements are low
 - Ex: Point-to-Point Communication Land, Air and Maritime Mobile communications TV, Telemetry and Military communications Radio Navigation & Amateur Radio

4. Frequency Division Multiplexing & DeMultiplexing (with DSB-SC)

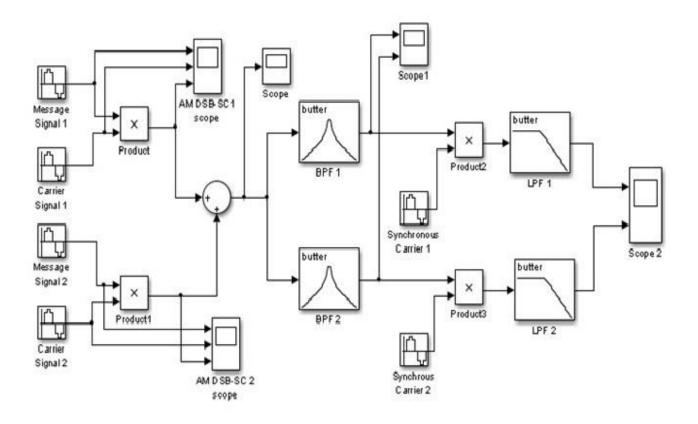
Aim

To perform the Frequency Division Multiplexing with AM DSB-SC sign**a**ls using Matlab Simulink.

Apparatus Required

- a) Hardware Tools: Computer system
- b) Software Tool: MATLAB 7.0 and above version

Simulink Model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon 👰. Go to file and select new and then select model. You will get a new window.

- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

Parameters

🚹 Source Block Parameters: Message Signal 1	×	🔁 Source Block Parameters: Carrier Signal 1
Use the sample-based sine type if numerical problems due to run for large times (e.g. overflow in absolute time) occur.	ning	Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
Parameters		Parameters
Sine type: Time based	•	Sine type: Time based 🔹
Time (t): Use simulation time	•	Time (t): Use simulation time 🔹
Amplitude:		Amplitude:
1		1
Bias:		Bias:
0		0
Frequency (rad/sec):		Frequency (rad/sec):
2*pi*5	н	2*pi*50 E
Phase (rad):		Phase (rad):
0		0
Sample time:		Sample time:
0.001		0.001
☑ Interpret vector parameters as 1-D	Ļ	☑ Interpret vector parameters as 1-D
< []	Þ	۲
OK Cancel Help App	oly	OK Cancel Help Apply

🚡 Source Block Parameters: Message Signal 2	📄 📴 Source Block Parameters: Carrier Signal 2
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.	Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
Parameters	Parameters
Sine type: Time based 🔹	Sine type: Time based 🔹
Time (t): Use simulation time 🔹	Time (t): Use simulation time 💌
Amplitude:	Amplitude:
1	1
Bias:	Bias:
0	0
Frequency (rad/sec):	Frequency (rad/sec):
2*pi*10	■ 2*pi*100 ■
Phase (rad):	Phase (rad):
0	0
Sample time:	Sample time:
0.001	0.001
☑ Interpret vector parameters as 1-D	☑ Interpret vector parameters as 1-D
* m	· · · · · · · · · · · · · · · · · · ·
OK Cancel Help Apply	OK Cancel Help Apply

1

	🛅 Function Block Parameters: Sum
Product	Sum
Multiply or divide inputs. Choose element-wise or matrix product and specify one of the following: a) * or / for each input port. For example, **/* performs the	Add or subtract inputs. Specify one of the following: a) string containing + or - for each input port, for spacer between
operation 'u1*u2/u3*u4'.	ports (e.g. ++ - ++)
 b) scalar specifies the number of input ports to be multiplied. If there is only one input port and the Multiplication parameter is set to 	b) scalar, >= 1, specifies the number of input ports to be summed.
Element-wise(.*), a single * or / collapses the input signal using the	When there is only one input port, add or subtract elements over all dimensions or one specified dimension
specified operation. However, if the Multiplication parameter is set to Matrix(*), a single * causes the block to output the matrix unchanged,	amensions of one specified amension
and a single / causes the block to output the matrix inverse,	
Main Signal Attributes	Main Signal Attributes
Number of inputs:	Icon shape: round
2	List of signs:
Multiplication: Element-wise(.*)	1++
Sample time (-1 for inherited):	
0.001	Sample time (-1 for inherited):
	0.001
	· [
OK Cancel Help Apply	OK Cancel Help App
Function Block Parameters: BPF 1	Function Block Parameters: BPF 2
Function Block Parameters: BPF 1	Function Block Parameters: BPF 2 Analog Filter Design (mask) (link)
Analog Filter Design (mask) (link)	Analog Filter Design (mask) (link)
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- space form.	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state space form.
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- apace form.	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in stat
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- pace form. Parameters	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in stat space form.
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- pace form. Design method: Butterworth	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in stat space form. Parameters
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- epace form. Parameters Design method: Butterworth • Filter type: Bandpass •	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in stat space form. Parameters Design method: Butterworth
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- epace form. Parameters Design method: Butterworth • Filter type: Bandpass •	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in stat space form. Parameters Design method: Butterworth Filter type: Bandpass
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 5	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in statispace form. Parameters Design method: Butterworth Filter type: Bandpass Filter order:
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- pace form. Parameters Design method: Butterworth • Filter type: Bandpass • Filter order: 5 Lower passband edge frequency (rad/s):	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in stat space form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 8
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth =ilter type: Bandpass =ilter order: 5 .ower passband edge frequency (rad/s): 2*pi*45	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in statispace form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 8 Lower passband edge frequency (rad/s): 2*pi*90
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 5 Lower passband edge frequency (rad/s): 2*pi*45 Upper passband edge frequency (rad/s):	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state space form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 8 Lower passband edge frequency (rad/s): 2*pi*90 Upper passband edge frequency (rad/s): 1
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 5 Lower passband edge frequency (rad/s):	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state space form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 8 Lower passband edge frequency (rad/s): 2*pi*90
Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- space form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 5 Lower passband edge frequency (rad/s): 2*pi*45 Upper passband edge frequency (rad/s):	Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state space form. Parameters Design method: Butterworth Filter type: Bandpass Filter order: 8 Lower passband edge frequency (rad/s): 2*pi*90 Upper passband edge frequency (rad/s): 1

🞦 Source Block Parameters: Synchronous Carrier 1 🖉	x
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.	×
Parameters	
Sine type: Time based	
Time (t): Use simulation time	
Amplitude:	
1	
Bias:	
0	
Frequency (rad/sec):	
2*pi*50	
Phase (rad):	
0	
Sample time:	
0.001	
☑ Interpret vector parameters as 1-D	4
۲ [] ۲	
OK Cancel Help Apply	

Parameters		
Sine type: [Time based	•
Time (t): 🛛	se simulation time	Ŧ
Amplitude:		
1		
Bias:		
0		
Frequency (rad/sec):	
2*pi*100		
Phase (rad)		
0		
Sample time	1	
0.001		

Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.	Analog Filter Design (mask) (lir
Parameters Sine type: Time based	Design one of several standard space form.
Time (t): Use simulation time Amplitude:	Parameters
1	Design method: Butterworth
Bias:	Filter type: Lowpass
Frequency (rad/sec):	Filter order:
2*pi*100 E Phase (rad):	5
0	Passband edge frequency (rad
Sample time: 0.001	2*pi*10
☑ Interpret vector parameters as 1-D	
< TIT OK Cancel Help Apply	OK

Function Block Parameters: LPF 2

Analog Filter Design (mask) (link)

Design one of several standard analog filters, implemented in state-space form.

Parameters

Design method:

Butterworth

Filter type:

Lowpass

Filter order:

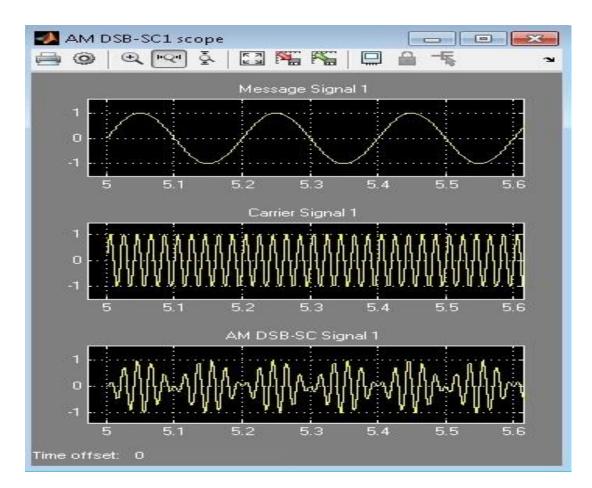
5

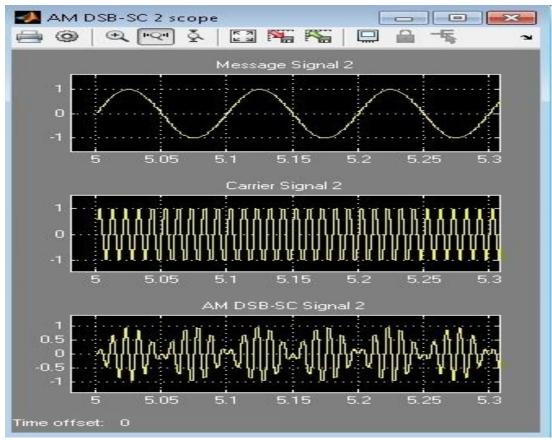
Passband edge frequency (rad/s):

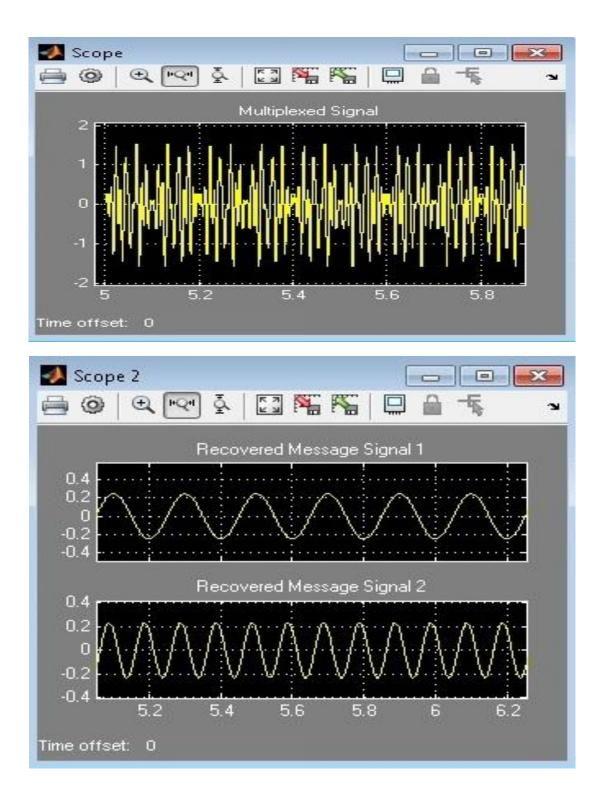
2*pi*10

OK Cancel Help Apply

Output







Result

Viva Questions

- 1. What is Multiplexing?
- 2. What are the different types of Multiplexing techniques?
- 3. What is the difference between FDM and TDM?
- 4. What are the advantages and disadvantages of FDM?
- 5. What is the difficult part in FDM?
- 6. What is the overall bandwidth if N number of signals are multiplexed?
- 7. Why AM SSB-SC is preferred for FDM?
- 8. What is Demultiplexing?

Applications

- 1. FDM is used for FM & AM radio broadcasting. AM broadcasting uses a bandwidth of 550-1650 KHz, where as FM broadcasting used a bandwidth of 88-108 MHz
- 2. FDM is used in Television broadcasting.
- 3. First generation Cellular telephone also uses FDM.
- 4. Used in Stereo FM transmissions.
- 5. Twentieth century telephone companies used FDM for long-distance connections to multiplex thousands of voice signals through co-axial cable systems.
- 6. Telemetry
 - a. Used to send feedback from multiple sensors over a single channel
- 7. Telephone Systems
 - a. Had been used for decades to send multiple telephone conversations over a minimum number of cables
 - b. The multiplexing process is used at multiple levels to send 10,800 phone calls over a single channel
- 8. Cable TV
 - a. Multiple TV signals are multiplexed on a common coaxial cable

5. Frequency Modulation & Demodulation

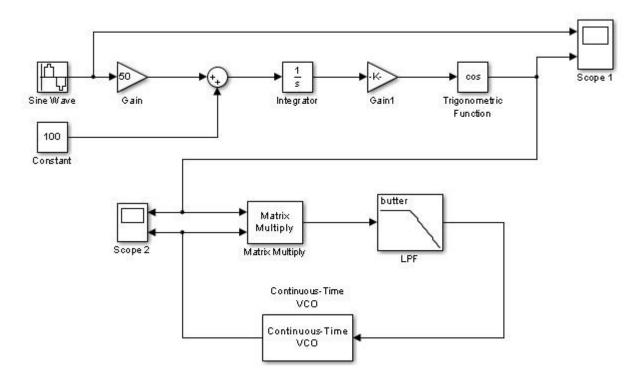
Aim

To perform the Frequency Modul \mathbf{a} tion \mathbf{s} ignal Gene \mathbf{r} ation and Detection using Matlab Simulink.

Apparatus Required

- a) Hardware Tools: Computer system
- b) Software Tool: MATLAB 7.0 and above version.

Simulink Model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink icon . Go to file and select new and then select model. You will get a new window.
- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

Parameters

Source Block Parameters: Sine Wave	🞦 Source Block Parameters: Constant
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.	Constant
Parameters Sine type: Time based Time (t): Use simulation time	Output the constant specified by the 'Constant value' parameter. If 'Constant value' is a vector and 'Interpret vector parameters as 1-D' is on, treat the constant value as a 1-D array. Otherwise, output a matrix with the same dimensions as the constant value.
Amplitude:	Main Signal Attributes
Bias:	Constant value:
Frequency (rad/sec):	IOO ☑ Interpret vector parameters as 1-D
Phase (rad):	Sampling mode: Sample based
Sample time:	Sample time:
Interpret vector parameters as 1-D	inf
<pre> III OK Cancel Help Apply </pre>	OK Cancel Help Apply

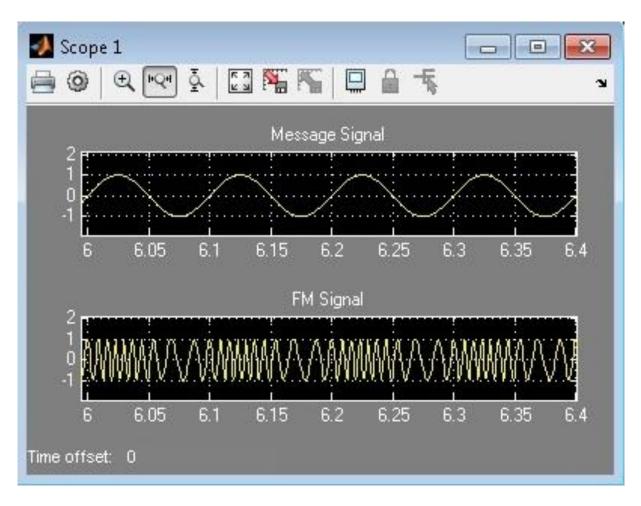
🚹 Function Block Parameters: Gain 🛛 💦 💽	🚡 Function Block Parameters: Sum
Gain Element-wise gain (y = K.*u) or matrix gain (y = K*u or y = u*K). Main Signal Attributes Parameter Attributes Gain:	Sum Add or subtract inputs. Specify one of the following: a) string containing + or - for each input port, for spacer between ports (e.g. ++ - ++) b) scalar, >= 1, specifies the number of input ports to be summed. When there is only one input port, add or subtract elements over all dimensions or one specified dimension
50 Multiplication: Element-wise(K.*u) Sample time (-1 for inherited):	Main Signal Attributes Icon shape: round List of signs:
-1 OK Cancel Help Apply	Sample time (-1 for inherited): 0.001 (

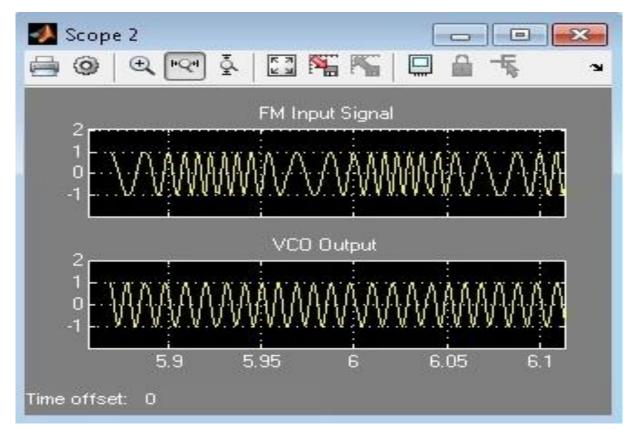
🛅 Function Block Parameters: Integrator	×	Pa Function Block Parameters: Matrix Multiply
Integrator	-	Product
Continuous-time integration of the input signal.		Multiply or divide inputs. Choose element-wise or matrix product and
Parameters		specify one of the following:
External reset: none		a) * or / for each input port. For example, **/* performs the
Initial condition source: internal		operation 'u1*u2/u3*u4'. b) scalar specifies the number of input ports to be multiplied.
		If there is only one input port and the Multiplication parameter is set to
Initial condition:		Element-wise(.*), a single * or / collapses the input signal using the
0		specified operation. However, if the Multiplication parameter is set to Matrix(*), a single * causes the block to output the matrix unchanged,
🗖 Limit output		and a single / causes the block to output the matrix inverse.
Upper saturation limit:		
inf	ш	Main Signal Attributes
Lower saturation limit:		Number of inputs:
-inf		2
Show saturation port		
Show state port		Multiplication: Matrix(*)
Absolute tolerance:		Sample time (-1 for inherited):
auto		-1
Ignore limit and reset when linearizing		
Enable zero-crossing detection		
State Name: (e.g., 'position')		
0		
	T	
OK Cancel Help Apply		OK Cancel Help Apply

🚹 Function Block Parameters: Trigonometric Function	Pa Function Block Parameters: Gain1
Trigonometric Function Trigonometric and hyperbolic functions. When the function has more than one argument, the first argument corresponds to the top (or left) input port. For sin, cos, sincos, cos +j sin, atan2 functions, CORDIC approximation can also be used to compute the output. Parameters	Gain Element-wise gain (y = K.*u) or matrix gain (y = K*u or y = u*K). Main Signal Attributes Parameter Attributes Gain:
Function: cos	2*pi
Approximation method: None Output signal type: auto Sample time (-1 for inherited):	Multiplication: Element-wise(K.*u) Sample time (-1 for inherited):
-1	-1
OK Cancel Help Apply	OK Cancel Help Apply

Function Block Parameters: LPF	🚡 Function Block Parameters: Continuous-Time VCO
	Continuous-Time VCO (mask) (link)
Design one of several standard analog filters, implemented in state- space form.	Generate a continuous-time output signal whose frequency changes in response to the amplitude variations of the input signal. The input signal must be a sample-based scalar.
Parameters	Parameters Output amplitude (V):
Design method: Butterworth	
Filter type: Lowpass	Quiescent frequency (Hz):
Filter order:	Input sensitivity (Hz/V):
5	50
Passband edge frequency (rad/s):	Initial phase (rad):
2*pi*10	0
OK Cancel Help Apply	OK Cancel Help Apply

Output





Result

Viva Questions

- 1. Define FM & PM.
- 2. What are the advantages of Angle modulation over amplitude modulation?
- 3. What is the relationship between PM and FM?
- 4. With a neat block diagram explain how PM is generated using FM.

Applications

- 1. Used for high quality music transmission
- 2. Entertainment broadcasting.
- 3. Used in two way Radio Communication links
- 4. Mobile Radio Communications
- 5. Used for broadcasting music and speech, magnetic tape recording systems, two way radio systems and video transmission systems.
- 6. When noise occurs naturally in radio systems, frequency modulation with sufficient bandwidth provides an advantage in cancelling the noise.
- 7. Frequency modulation is used in audio frequencies to synthesize sound.
- 8. For recording the video signals by VCR systems, frequency modulation is used for intermediate frequencies.

Advantages of Frequency Modulation:

- Frequency modulation has more noise resistivity when compared to other modulation techniques. That's why they are mainly used in broadcasting and radio communications. And we are all well aware that radio communication use mainly frequency modulation for transmission. We know that noise will occur mainly to the amplitude of the signal. In frequency modulation, amplitude is made constant and only frequency is varied, so we can easily find out the noise in the amplitude by using a limiter.
- The frequency modulation is having greater resistance to rapid signal strength variation, which we will use in FM radios even while we are travelling and frequency modulation is also mainly used in mobile communication purposes.
- For transmitting messages in frequency modulation, it does not require special equipments like linear amplifiers or repeaters and transmission levels or higher when compared to other modulation techniques. It does not require any class C or B amplifiers for increasing the efficiency.
- Transmission rate is good for frequency modulation when compared to other modulation that is frequency modulation can transmit around 1200 to 2400 bits per second.
- Frequency modulation has a special effect called capture effect in which high frequency signal will capture the channel and discard the low frequency or weak signals from interference.

Disadvantages of Frequency Modulation:

- In the transmission section, we don't need any special equipment but in the reception, we need more complicated demodulators for demodulating the carrier signal from message or modulating signal.
- Frequency modulation cannot be used to find out the speed and velocity of a moving object. Static interferences are more when compared to phase modulation. Outside interference is one of the biggest disadvantages in the frequency modulation. There may be mixing because of nearby radio stations, pagers, construction walkie-talkies etc.
- To limit the bandwidth in the frequency modulation, we use some filter which will again introduce some distortions in the signal.
- Transmitters and receiver should be in same channel and one free channel must be there between the systems.
- Spectrum space is limit for the frequency modulation and careful controlling the deviation ration.

6. PLL as FM Demodulator

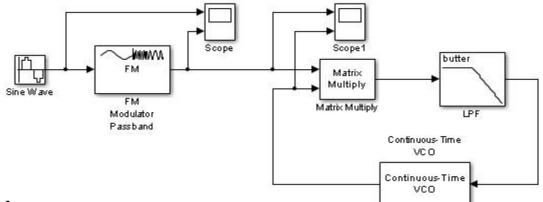
Aim

To perform the operation of FM Demodulation with PLL using Matlab Simulink.

Apparatus Required:

- a) Hardware Tools: Computer system
- b) Software Tool: MATLAB 7.0 or Upgraded version.

Simulink Model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon *icon*. Go to file and select new and then select model. You will get a new window.

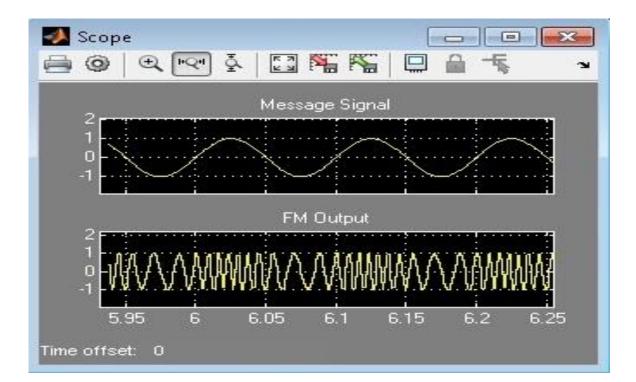
- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

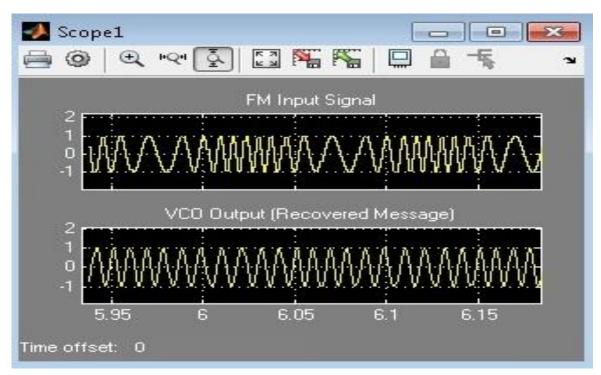
Parameters

🚡 Function Block Parameters: FM Modulator Passband 📃 🔀
FM Modulator Passband (mask) (link)
Modulate the input signal using the frequency modulation method.
The input signal must be a scalar.
Parameters
Carrier frequency (Hz):
100
Initial phase (rad):
0
Frequency deviation (Hz):
50
OK Cancel Help Apply

🔁 Source Block Parameters: Sine Wave	🛐 Function Block Parameters: Matrix Multiply
Source Block Parameters: Sine Wave Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur. Parameters Sine type: Time based Time (t): Use simulation time Amplitude: 1	Product Multiply or divide inputs. Choose element-wise or matrix product and specify one of the following: a) * or / for each input port. For example, **/* performs the operation 'u1*u2/u3*u4'. b) scalar specifies the number of input ports to be multiplied. If there is only one input port and the Multiplication parameter is set to Element-wise(.*), a single * or / collapses the input signal using the specified operation. However, if the Multiplication parameter is set to Matrix(*), a single * causes the block to output the matrix unchanged, and a single / causes the block to output the matrix inverse.
Bias: 0 Frequency (rad/sec): 2*pi*10 Phase (rad): 0 Sample time: 1/1000 ✓ Interpret vector parameters as 1-D	Main Signal Attributes Number of inputs: Image: Strate S
< OK Cancel Help Apply	OK Cancel Help Apply
Function Block Parameters: LPF Analog Filter Design (mask) (link) Design one of several standard analog filters, implemented in state- space form.	Function Block Parameters: Continuous-Time VCO Continuous-Time VCO (mask) (link) Generate a continuous-time output signal whose frequency changes in response to the amplitude variations of the input signal. The input signal must be a sample-based scalar.
Parameters	Parameters

🔁 Function Block Parameters: LPF	Pa Function Block Parameters: Continuous-Time VCO
	Continuous-Time VCO (mask) (link)
Design one of several standard analog filters, implemented in state- space form.	Generate a continuous-time output signal whose frequency changes in response to the amplitude variations of the input signal. The input signal must be a sample-based scalar.
Parameters	Parameters
	Output amplitude (V):
Design method: Butterworth	
Filter type: Lowpass	Quiescent frequency (Hz):
- the sector	100
Filter order:	Input sensitivity (Hz/V):
5	50
Passband edge frequency (rad/s):	Initial phase (rad):
2*pi*10	0
OK Cancel Help Apply	OK Cancel Help Apply





Result

Viva Questions

- 1. What is PLL?
- 2. What are the applications of PLL?
- 3. What is the free running frequency?
- 4. What is the function of VCO?
- 5. What is the function of Low Pass Filter in PLL?
- 6. What is the function of Phase Detector?

7. Spectral Characteristics of AM & FM

Aim

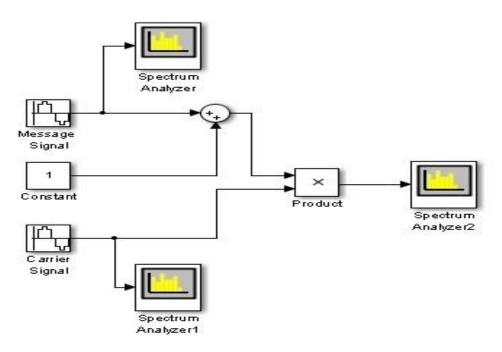
To verify the Spectral Components of AM and FM using Matlab Simulink.

Apparatus Required

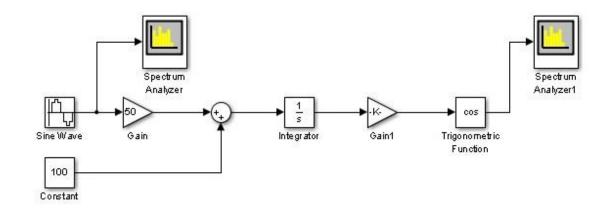
- a) Hardware Tools: Computer system
- b) Software Tool: MATLAB 7.0 or Upgraded version.

Simulink Model

Amplitude Modulation Setup



Frequency Modulation Setup



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon **•**. Go to file and select new and then select model. You will get a new window.

- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

Parameters

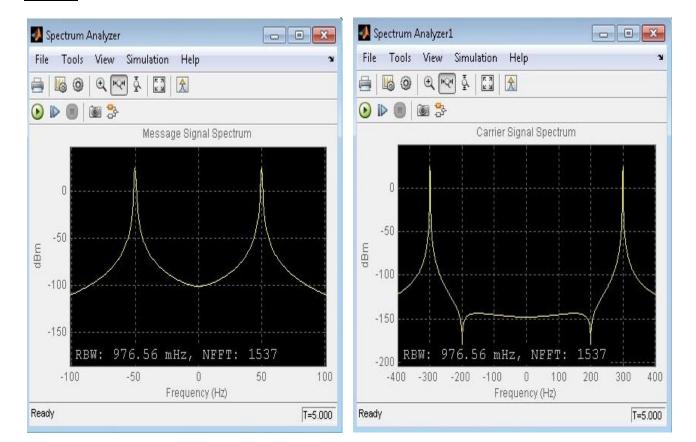
For Amplitude Modulation:

Set the Message signal amplitude = 1V and frequency = 50 Hz Set the Carrier signal amplitude = 1V and frequency = 300 Hz

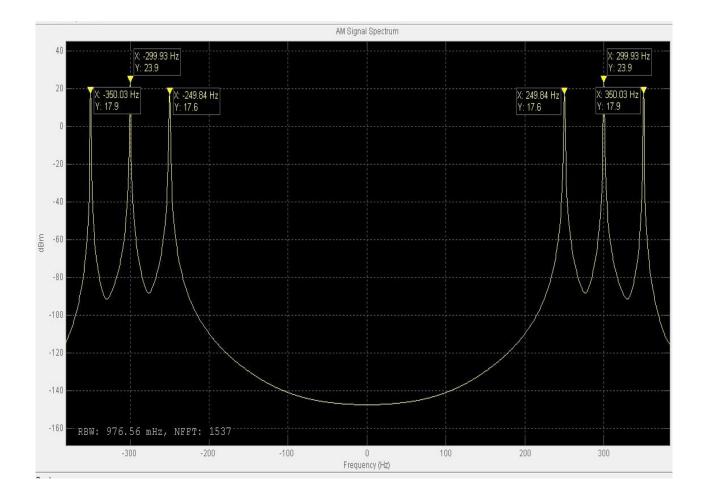
For Amplitude Modulation:

As per the setting of Frequency Modulation setup of Experiment No. 5

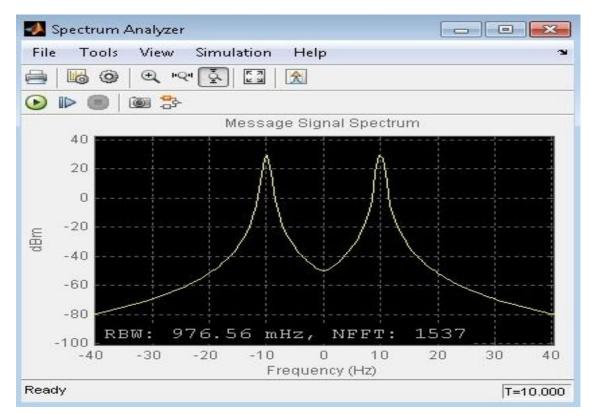
Output

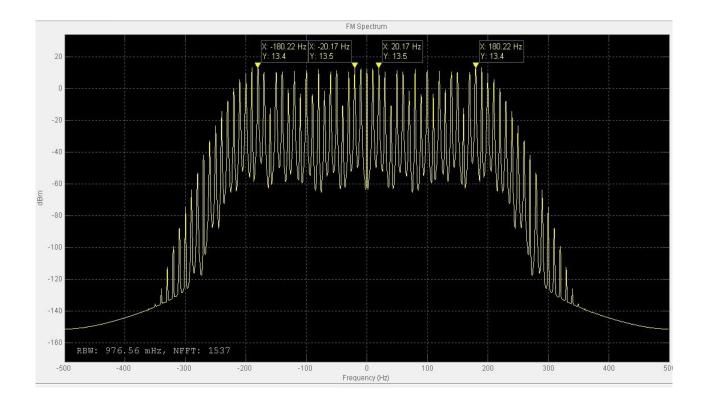


For AM



For FM





Viva Questions

- 1. What is the Spectrum?
- 2. Why we consider negative frequencies while dealing with Specturm?
- 3. What is the bandwidth of AM?
- 4. What is the bandwidth of FM?
- 5. What is the theoretical bandwidth of FM signal?
- 6. Draw the spectrum of the message signal with frequency of 1000 Hz.
- 7. What is the difference between Spectrums of AM and FM?
- 8. Draw the spectrum of AM DSB-SC?
- 9. Draw the spectrum of AM SSB-SC when USB is extracted from the transmitter?
- 10. Draw the spectrum of AM SSB-SC when LSB is extracted from the transmitter?

Applications

1. Frequency domain analysis of AM & FM signals.

8. Verification of Sampling Theorem

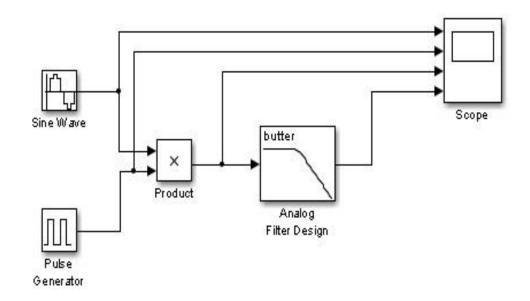
Aim

To verify Sampling theorem for different Sampling frequencies using Matlab Simulink.

Apparatus Required

- a) Ha**r**dware Tools: Computer system
- b) Software Tool: MATLAB 7.0 or Upgraded version.

Simulink Model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon 🚵. Go to file and select new and then select model. You will get a new window.

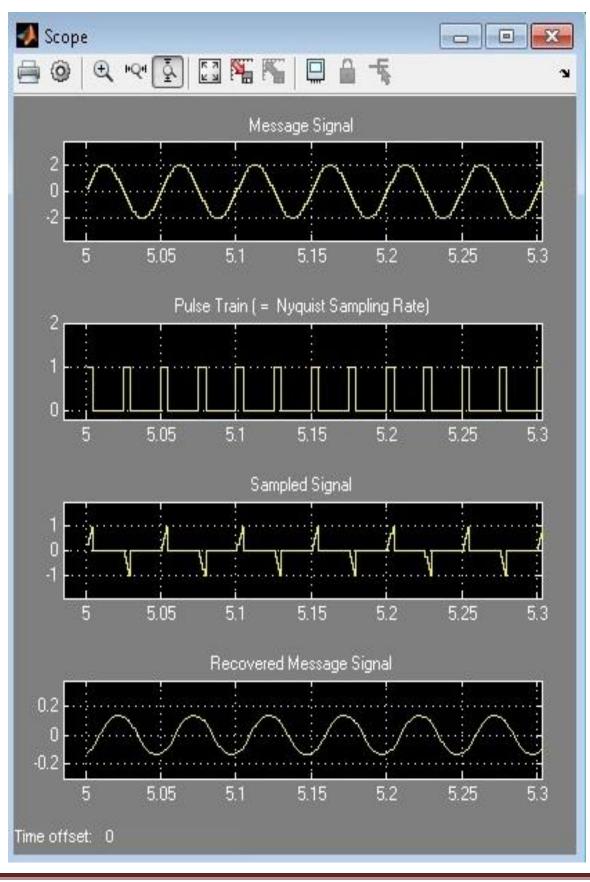
- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

Parameters

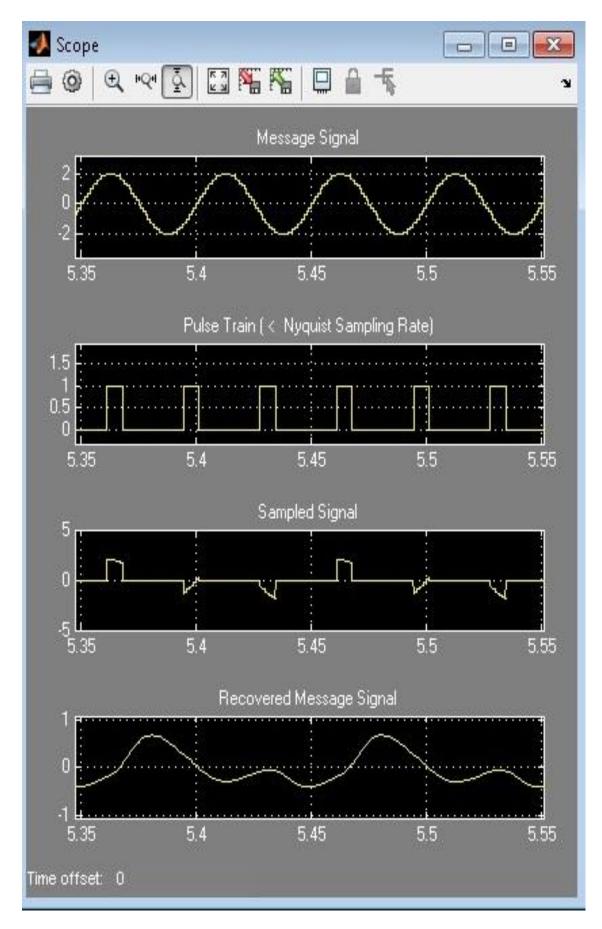
🚡 Source Block Parameters: Sine Wave	🛅 Source Block Parameters: Pulse Generator
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.	Pulse Generator Output pulses:
Parameters Sine type: Time based Time (t): Use simulation time Amplitude:	if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end
Bias:	Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, while Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver.
Frequency (rad/sec): 2*pi*20	Parameters Pulse type: Time based
Phase (rad):	Time (t): Use simulation time Amplitude:
Sample time: 0.001	1 Period (secs):
Interpret vector parameters as 1-D	0.025 Pulse Width (% of period)
OK Cancel Help Apply	OK Cancel Help Apply

🔁 Function Block Parameters: Analog Filter Design 🛛 🛛 💽		
Analog Filter Design (mask) (link)		
Design one of several standard analog filters, implemented in state- space form.		
Parameters		
Design method: Butterworth		
Filter type: Lowpass		
Filter order:		
3		
Passband edge frequency (rad/s):		
2*pi*20		
OK Cancel Help Apply		

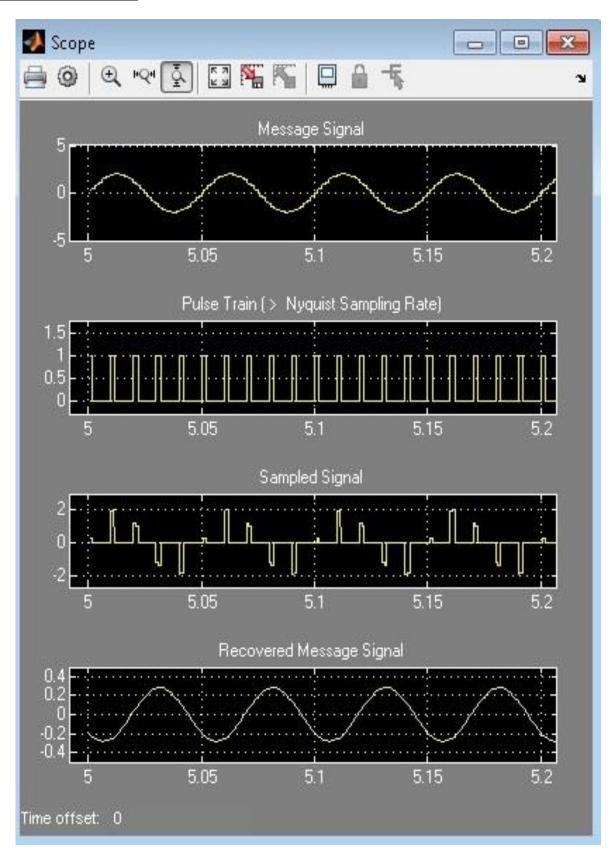
Nyquist Rate



Under Sampling Rate



Over Sampling Rate



Result

Viva Questions

- 1. What are the types of sampling?
- 2. State sampling theorem?
- 3. What happens when $f_s < 2 f_m$?
- 4. How will be the reconstructed signal when $f_s \ge 2f_m$?
- 5. Explain the operation of sampling circuit?
- 6. Explain the operation of re-construction circuit?
- 7. Who formalized the sampling theorem?
- 8. What are the applications of the Sampling theorem?
- 9. Is the sampling theorem basis for the modern digital communications?
- 10. Is the voice signal sampling of 8000 Hz, follows sampling theorem in Land line Telephone Exchange.

9. Pulse Amplitude Modulation & Demodulation

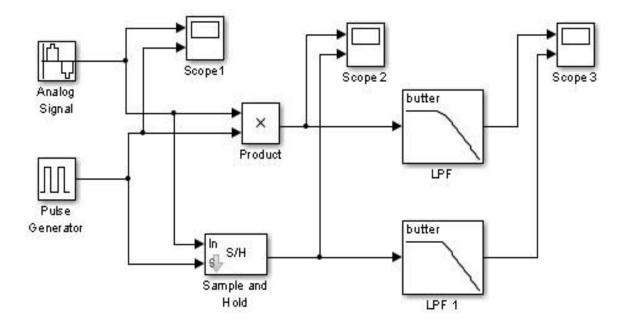
Aim

To perform Pulse Amplitude Modulation and Demodulation using Matlab Simulink.

Apparatus Required

- a) Hardware Tools: Computer system
- b) Software Tool: MATLAB 7.0 or Upgraded version.

Simulink Model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon **i**. Go to file and select new and then select model. You will get a new window.

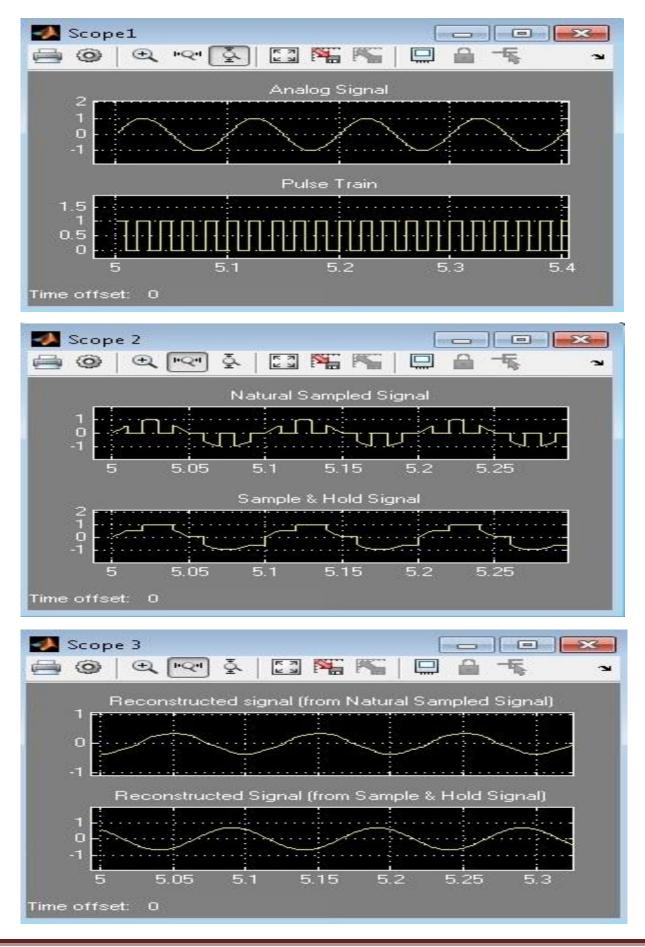
- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

Parameters

🚹 Source Block Parameters: Analog Signal	×	Source Block Parameters: Pulse Generator	×
Use the sample-based sine type if numerical problems due for large times (e.g. overflow in absolute time) occur.	to running	Pulse type determines the computational technique used.	*
Parameters		Time-based is recommended for use with a variable step solver, w Sample-based is recommended for use with a fixed step solver or	
Sine type: Time based		within a discrete portion of a model using a variable step solver.	
		Parameters	-
Time (t): Use simulation time		Pulse type: Time based	-
Amplitude:		Time (t): Use simulation time	_
		··· L	-
Bias:		Amplitude:	
0		1	
Frequency (rad/sec):		Period (secs):	E
2*pi*10	=	0.02	
Phase (rad):		Pulse Width (% of period):	_
		50	
Sample time: 0.001		Phase delay (secs):	_
☑ Interpret vector parameters as 1-D	-	Interpret vector parameters as 1-D	-
< [•	۲. [] ۲. []	F
OK Cancel Help	Apply	OK Cancel Help App	ly
🚡 Function Block Parameters: Sample and Hold	**	🛐 Function Block Parameters: LPF	×
Sample and Hold (mask) (link)		Analog Filter Design (mask) (link)	
Sample and hold input signal.		Design one of several standard analog filters, implemented in stat space form.	e-
The output follows input 1 (In) as long as input 2 (S) is TRU output is held when input 2 becomes FALSE (0).	JE (1). The	Parameters	
Parameters		Design method: Butterworth	•
Initial condition:		Filter type: Lowpass	•]
0		Filter order:	
Sample time:	14	2	
0.001		Passband edge frequency (rad/s):	
		2*pi*10	
OK Cancel Help	Apply	OK Cancel Help App	ly
P. Function Dis als Davis	DE 1		

space form.			
Parameters			
Design method	: Butterworth	1	
Filter type: Lo	wpass		
5			
Passband edge	frequency (ra	ad/s):	
2*pi*10			

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Result

Viva Questions

- 1. TDM is possible for sampled signals. What kind of multiplexing can be used in continuous modulation systems?
- 2. What is the minimum rate at which a speech signal can be sampled for the purpose of PAM?
- 3. What is cross talk in the context of time division multiplexing?
- 4. Which is better, natural sampling or flat topped sampling and why?
- 5. Why a dc offset has been added to the modulating signal in this board? Was it essential for the working of the modulator? Explain?
- 6. If the emitter follower in the modulator section saturates for some level of input signal, then what effect it will have on the output?
- 7. Derive the mathematical expression for frequency spectrum of PAM signal.
- 8. Explain the modulation circuit operation?
- 9. Explain the demodulation circuit operation?
- 10. Is PAM & Demodulation is sensitive to Noise?

Applications

- 1. It is mainly used in Ethernet which is type of computer network communication, we know that we can use Ethernet for connecting two systems and transfer data between the systems. Pulse amplitude modulation is used for Ethernet communications.
- 2. It is also used for photo biology which is a study of photosynthesis.
- 3. Used as electronic driver for LED lighting.
- 4. Used in many micro controllers for generating the control signals etc.

10. Time Division Multiplexing & DeMultiplexing

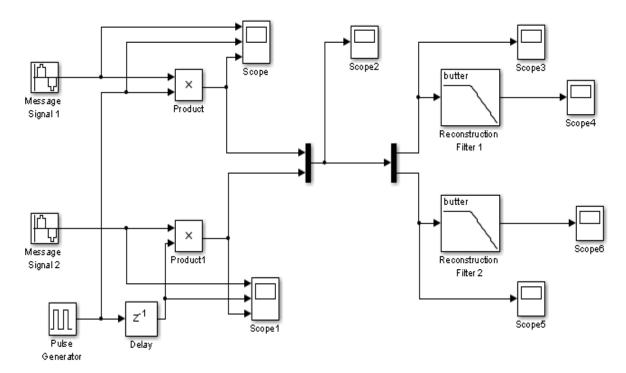
Aim

To perform the Time Division Multiplexing using Matlab Simulink.

Apparatus Required

- a) Hardware Tools: Computer system
- b) Software Tool: MATLAB 7.0 or Upgraded Version

Simulink Model



Procedure

- 1. Switch on the computer and click on the MATLAB icon.
- 2. Go to start at the bottom of the command window, then select "Simulink" then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon **i**. Go to file and select new and then select model. You will get a new window.

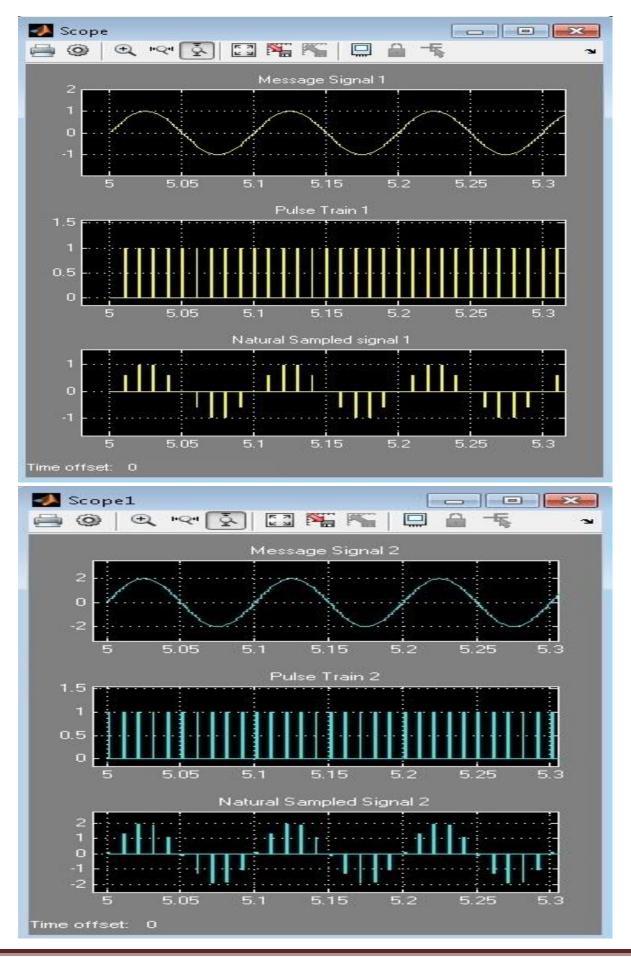
- 3. Arrange the functional blocks as shown in Simulink model.
- 4. Assign required parameters to each functional block.
- 5. Observe the outputs on scope.

Parameters

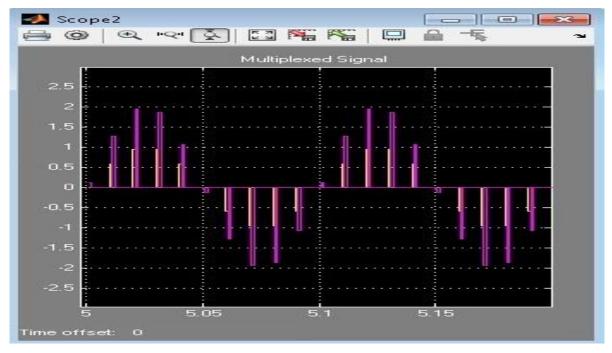
🚹 Source Block Parameters: Message Signal 1 📃 📈	📄 👘 Source Block Parameters: Message Signal 1
Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.	Use the sample-based sine type if numerical problems due to running for large times (e.g. overflow in absolute time) occur.
Parameters	Parameters
Sine type: Time based	Sine type: Time based 🔹
Time (t): Use simulation time	Time (t): Use simulation time
Amplitude:	Amplitude:
1	1
Bias:	Bias:
0	0
Frequency (rad/sec):	Frequency (rad/sec):
2*pi*10	≡ 2*pi*10 E
Phase (rad):	Phase (rad):
0	0
Sample time:	Sample time:
0.001	0.001
☑ Interpret vector parameters as 1-D	☑ Interpret vector parameters as 1-D
۰ m ۲	• · · · · · · · · · · · · · · · · · · ·
OK Cancel Help Apply	OK Cancel Help Apply
Source Block Parameters: Pulse Generator	Eurotion Block Parameters: Delay
Source Block Parameters: Pulse Generator Image: Constant State	
	Delay
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude	
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage.
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else	Delay Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0	Delay Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi	Delay Delay Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Upper Limit
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used.	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Upper Limit Delay length: Dialog
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi Sample-based is recommended for use with a fixed step solver or	Delay Delay Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Upper Limit
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver.	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Upper Limit Delay length: Dialog
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver. Parameters Pulse type: Time based	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Delay length: Dialog 1 Initial condition: Dialog 0
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver. Parameters Pulse type: Time based Time (t): Use simulation time	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Upper Limit Delay length: Dialog • 1 Initial condition: Dialog • 0 Algorithm External reset:
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver. Parameters Pulse type: Time based	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Upper Limit Delay length: Dialog • Initial condition: Dialog • Algorithm External reset: Input processing: Elements as channels (sample based) •
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whith Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver. Parameters Pulse type: Time based Time (t): Use simulation time Amplitude: •	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Upper Limit Delay length: Dialog • 1 Initial condition: Dialog • 0 Algorithm External reset:
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver. Parameters Pulse type: Time based Time (t): Use simulation time Amplitude: 1	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Upper Limit Delay length: Dialog • Initial condition: Dialog • Algorithm External reset: Input processing: Elements as channels (sample based) •
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver. Parameters Pulse type: Time based Time (t): Use simulation time Amplitude: 1 Period (secs): 0.01 Pulse Width (% of period):	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Delay length: Dialog 1 Initial condition: Dialog 0 Algorithm External reset: None Input processing: Elements as channels (sample based) • Use circular buffer for state Use circular buffer for state
Pulse Generator Output pulses: if (t >= PhaseDelay) && Pulse is on Y(t) = Amplitude else Y(t) = 0 end Pulse type determines the computational technique used. Time-based is recommended for use with a variable step solver, whi Sample-based is recommended for use with a fixed step solver or within a discrete portion of a model using a variable step solver. Parameters Pulse type: Time based Time (t): Use simulation time Amplitude: 1 Period (secs): 0.01	Delay Delay input by a fixed or variable number of samples. Based on an external signal, the block can reset its state to the specified initial condition (from dialog or input port). The block supports both circular and array buffer for state storage. Main State Attributes Data Source Value Delay length: Dialog 1 Initial condition: Dialog 0 Algorithm External reset: None Input processing: Elements as channels (sample based) • Use circular buffer for state Use circular buffer for state

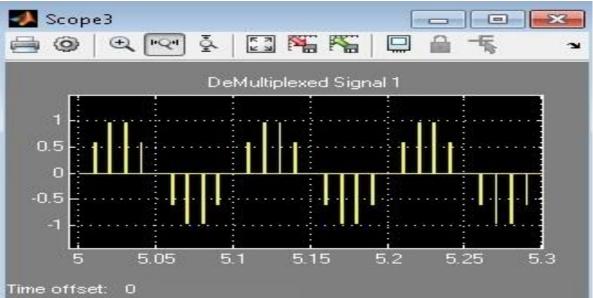
🚡 Function Block Parameters: Mux 🛛 🔀	🚡 Function Block Parameters: Demux
Mux Multiplex scalar or vector signals.	Demux Split vector signals into scalars or smaller vectors. Check 'Bus Selection Mode' to split bus signals.
Parameters Number of inputs:	Parameters Number of outputs: 2
2 Display option: bar	Display option: bar
OK Cancel Help Apply	OK Cancel Help Apply

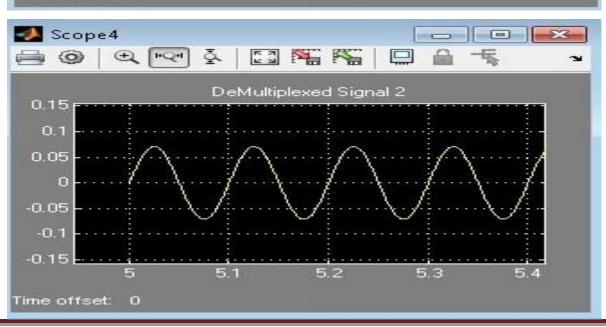
Tunction Block Parameters: Reconstruction Filter 1	🚹 Function Block Parameters: Reconstruction Filter 2
Analog Filter Design (mask) (link)	Analog Filter Design (mask) (link)
Design one of several standard analog filters, implemented in state- space form.	Design one of several standard analog filters, implemented in state- space form.
Parameters	Parameters
Design method: Butterworth	Design method: Butterworth
Filter type: Lowpass	Filter type: Lowpass
Filter order:	Filter order:
8	8
Passband edge frequency (rad/s):	Passband edge frequency (rad/s):
2*pi*10	2*pi*20
OK Cancel Help Apply	OK Cancel Help Apply



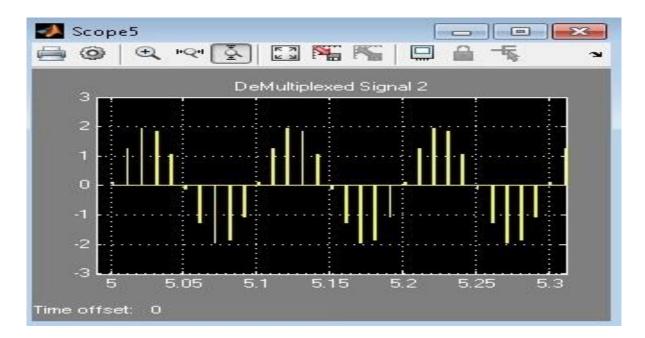
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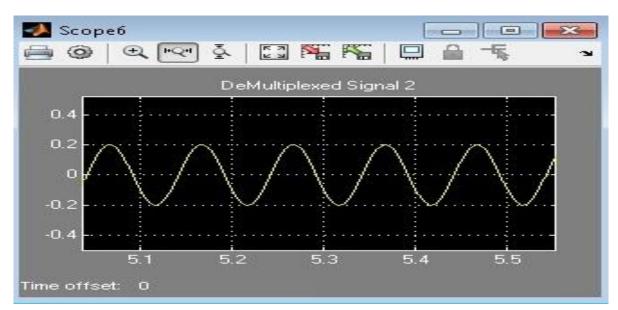






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Result

Viva Questions

- 1. Draw the TDM signal with 2 signals being multiplexed over the channel?
- 2. Define guard time & frame time?
- 3. Explain block schematic of TDM?
- 4. How TDM differ from FDM?
- 5. What type of filter is used at receiver end in TDM system?
- 6. What are the applications of TDM?

7. If 2 signal band limited to 3 kHz, 5 KHz & are to be time division multiplexed. What is the maximum permissible interval between 2 successive samples.?

8. Is the bandwidth requirement for TDM & FDM will be same?

9. Is TDM system is relatively immune to interference with in channels (inter channel cross talk) as compared to FDM?

10. Is the FDM susceptible to harmonic distortion compared to TDM?

11. In what aspects, TDM is superior to FDM?

Applications

- 1. Telephone based applications uses Pulse coded modulation signals which further uses time division multiplexing for the efficient use of channel bandwidth.
- 2. Used in wire line telephone systems and some cellular telephone systems.

11.PWM Modulation & Demodulation

Aim

To perform PWM modulation and Demodulation using MATLAB.

Apparatus Required

Ha**r**dware Tools: Computer system Software Tool: MATLAB 7.0 or Upgraded Version

Program

```
%PWM wave generation
```

```
t=0:0.001:1;
```

```
s=sawtooth(2*pi*10*t+pi);
```

```
m=0.75*sin(2*pi*1*t);
```

```
n=length(s);
```

```
for i=1:n
```

```
if (m(i)>=s(i))

pwm(i)=1;

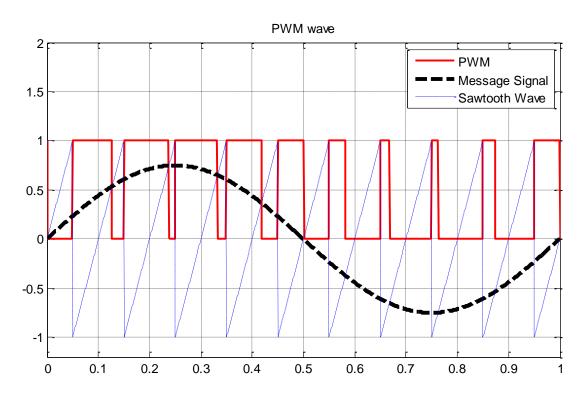
elseif (m(i)<=s(i))

pwm(i)=0;
```

end

end

plot(t,pwm,'-r',t,m,'--k',t,s,'--b');grid; title('PWM wave');axis([0 1 -1.5 1.5]);



Result

Viva Questions

1. An audio signal consists of frequencies in the range of 100Hz to 5.5KHz.What is the minimum frequency at which it should be sampled in order to transmit it through pulse modulation?

2. Draw a TDM signal which is handling three different signals using PWM?

3. What do you infer from the frequency spectrum of a PWM signal?

4. Clock frequency in a PWM system is 2.5 kHz and modulating signal frequency is 500Hzhowmany pulses per cycle of signal occur in PWM output? Draw the PWM signal?

5. Why should the curve for pulse width Vs modulating voltage be linear?

- 6. What is the other name for PWM?
- 7. What is the disadvantage of PWM?
- 8. Will PWM work if the synchronization between Tx and Rx fails?
- 9. Why integrator is required in demodulation of PWM?
- 10. What kind of conversion is done in PWM generation?

Applications

- 1. PWM is used in telecommunication systems.
- 2. PWM can be used to control the amount of power delivered to a load without incurring the losses. So, this can be used in power delivering systems.
- 3. Audio effects and amplifications purposes also used.
- 4. PWM signals are used to control the speed of the robot by controlling the motors.
- 5. PWM is also used in robotics.
- 6. Embedded applications.
- 7. Analog and digital applications etc.

12.Pulse Position Modulation & Demodulation

Aim

To simulate PPM modulation and demodulation using MATLAB.

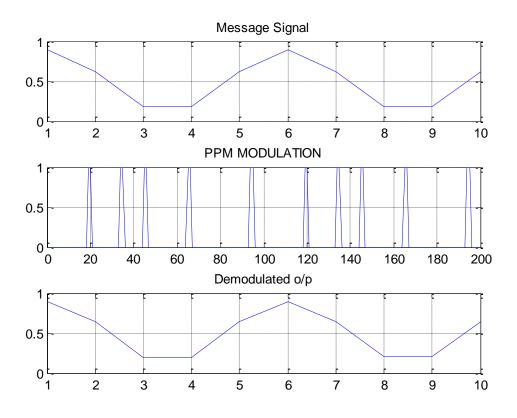
Apparatus Required

a) Hardware Tools: Computer system

b) Software Tool: MATLAB 7.0 or Upgraded Version

Code:

clc clear all; close all; fc=50; fs=1000; f1=200;f2=300; t=0:1/fs:((2/f1)-(1/fs)); x1=0.4*cos(2*pi*f1*t)+0.5; subplot(311);plot(x1); title('Message Signal'); grid; subplot(312); y=modulate(x1,fc,fs,'ppm'); plot(y); title('PPM MODULATION'); grid; z=demod(y,fc,fs,'ppm'); subplot(313);plot(z); title('Demodulated o/p'); grid;



Result

Viva Questions

- 1. What is the advantage of PPM over PWM?
- 2. Is the synchronization is must between Tx and Rx
- 3. Shift in the position of each pulse of PPM depends on what?
- 4. Can we generate PWM from PPM?
- 5. Why do we need 555 timers?
- 6. Does PPM contain derivative of modulating signal compared to PWM?
- 7. For above scheme, do we have to use LPF and integrator in that order?
- 8. If we convert PPM to PWM & then detect the message signal, will the o/p has less distortion?
- 9. Is synchronization critical in PPM?
- 10. How robust is the PPM to noise?

Applications

- 1. Used in non coherent detection where a receiver does not need any Phase lock loop for tracking the phase of the carrier.
- 2. Used in radio frequency (RF) communication.
- 3. Also used in contactless smart card, high frequency, RFID (radio frequency ID) tags and etc.

Advantages of Pulse Position Modulation (PPM)

- Pulse position modulation has low noise interference when compared to PAM because amplitude and width of the pulses are made constant during modulation.
- Noise removal and separation is very easy in pulse position modulation.
- Power usage is also very low when compared to other modulations due to constant pulse amplitude and width.

Disadvantages of Pulse Position Modulation (PPM)

- The synchronization between transmitter and receiver is required, which is not possible for every time and we need dedicated channel for it.
- Large bandwidth is required for transmission same as pulse amplitude modulation.
- Special equipments are required in this type of modulations.