Experiment 2

Objective:

1. Double Sideband- Supressed Carrier AM Generation.
2. Double Sideband- Supressed Carrier AM Reception.

Equipment Required:

1. ST2201 and ST2202 with power supply cord
2. CRO with connecting probe
3. Connecting cords

Theory:

Double-sideband suppressed-carrier transmission (DSB-SC) is transmission in which frequencies produced by amplitude modulation (AM) are symmetrically spaced above and below the carrier frequency and the carrier level is reduced to the lowest practical level, ideally being completely suppressed.

In the DSB-SC modulation, unlike in AM, the wave carrier is not transmitted; thus, much of the power is distributed between the sidebands, which implies an increase of the cover in DSB-SC, compared to AM, for the same power used.

DSB-SC is basically an amplitude modulation wave without the carrier, therefore reducing power waste, giving it a 100% efficiency. This is an increase compared to normal AM transmission (DSB), which has a maximum efficiency of 33.333%, since 2/3 of the power is in the carrier which carries no intelligence, and each sideband carries the same information. Single Side Band (SSB) Suppressed Carrier is 100% efficient.

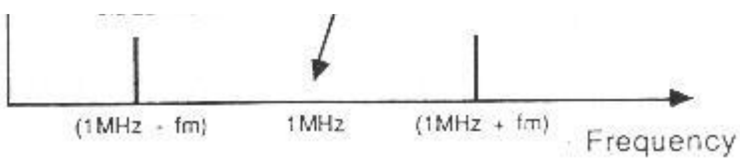

Figure 1

Frequency Spectrum of DSB-SC AM

Generation of DSB-SC AM

DSB-SC is generated by a mixer. This consists of a message signal multiplied by a carrier signal. The mathematical representation of this process is shown below, where theproduct-to-sum trigonometric identity is used.

$$\underbrace{V_m \cos(\omega_m t)}_{\text{Message}} \times \underbrace{V_c \cos(\omega_c t)}_{\text{Carrier}} = \underbrace{\frac{V_m V_c}{2}\left[\cos\left((\omega_m + \omega_c)\, t\right) + \cos\left((\omega_m - \omega_c)\, t\right)\right]}_{\text{Modulated Signal}}$$

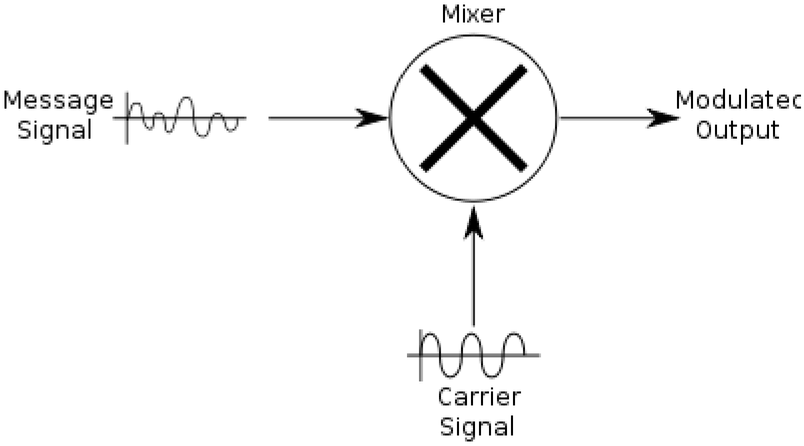

Figure 2

DSB-SC Modulator

A.Setup for Double Sideband- Supressed Carrier AM Generation.

Procedure :

1. Ensure that the following initial conditions exist on the board.

- **a.** Audio input select switch should be in INT position:
- **b.** Mode switch in DSB position.
- **c.** Output amplifier's gain potentiometer in full clockwise position.
- **d.** Speakers switch in OFF position.

2. Turn on power to the **ST2201** board.

3. Turn the audio oscillator block's amplitude pot to its full clockwise (MAX) position, and examine the block's output (TP14) on an oscilloscope.

This is the audio frequency sine wave which will be as our modulating signal. Note that the sine wave’s frequency can be adjusted from about 300 Hz to approximately 3.4 KHz, by adjusting the audio oscillator's frequency potentimeter.

Note also that the amplitude of this audio modulating signal can be reduced to zero, by turning the Audio oscillator's amplitude present to its fully counter-clockwise (MIN) position.

Return the amplitude present to its max position.

4. Turn the balance pot, in the balanced modulator & band pass filter circuit 1 block, to its fully clockwise position. It is this block that we will use to perform *double-side band amplitude modulation.*

5. Monitor, in turn, the two inputs to the balanced modulator & band pass filter circuits 1 block, at TP1 and TP9. Note that:

- **a.** The signal at TP1 is the audio-frequency sinewave from the audio oscillator block. This is the modulating input to our double-sideband modulator.
- **b.** Test Point 9 carries a sine wave of 1MHz frequency and amplitude 120mVpp approx. This is the carrier input to our double-sideband modulator.

6. Now vary the amplitude and frequency of the audio-frequency sinewave, by adjusting the amplitude and frequency present in the audio oscillator block. Note the effect that varying each pot has on the amplitude modulated waveform. The amplitude and frequency amplitudes of the two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero. Do this by turning the amplitude pot to its MIN position, and note that the signal at TP3 becomes an un-modulated sine wave of frequency 1 MHz, indicating that only the carrier component now remains. Return the amplitude pot to its maximum position.

Now turn the balance pot in the balanced modulator & band pass filter circuit 1 block, until the signal at TP3 is as shown in Figure 3

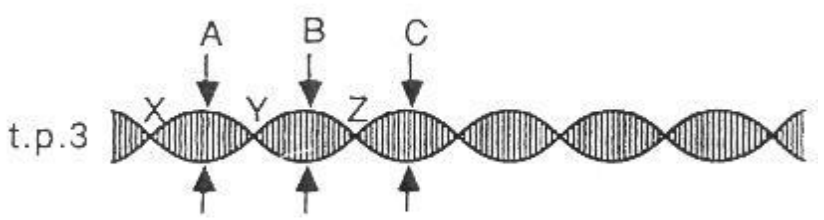

Figure 3

DSB-SC Waveform

The balance pot varies the amount of the 1 MHz carrier component, which is passed from the modulator's output.

By adjusting the pot until the peaks of the waveform (A, B, C and so on) have the same amplitude, we are removing the carrier component altogether.

We say that the carrier has been 'balanced out' (or 'suppressed') to leave only the two sidebands.

Note that once the carrier has been balanced out, the amplitude of TP3's waveform should be zero at minimum points X, Y; Z etc. If this is not the case, it is because one of the two sidebands is being amplified more than the other. To remove this problem, the band pass filter in the balanced modulator & band pass filter circuit 1 block must be adjusted so that it passes both sidebands equally. This is achieved by carefully trimming transformer T1, until the waveform's amplitude is as close to zero as possible at the minimum points.

The waveform at TP3 is known as a double-side suppressed carrier (DSBSC) waveform, and its frequency spectrum is as shown in Figure 4.

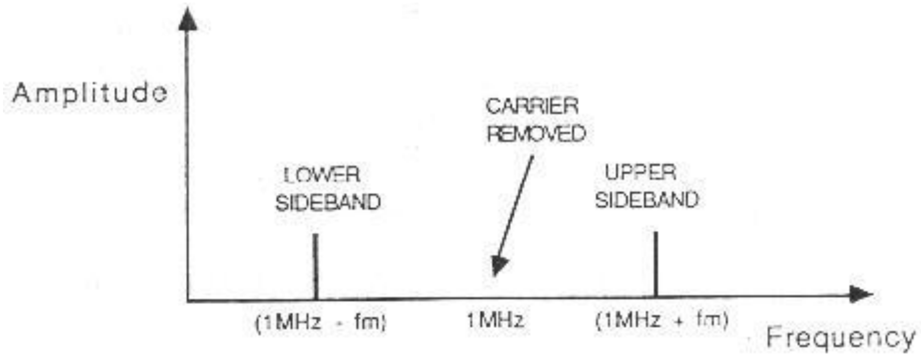

Figure 4

Frequency Spectrum of DSB-SC

Note that now only the two sidebands remain, the carrier component has been removed.

9. Change the amplitude and frequency of the modulating audio signal (by adjusting the audio oscillator block's amplitude and frequency pots), and note the effect that these changes on the DSBSC waveform. The amplitudes of the two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero. Do these by turning the amplitude present to its MIN position, and note that the monitored signal becomes a D C level, indicating that there .are now no frequency components present. Return the amplitude pot to its MAX position.

10. Examine the output from the output amplifier block (TP13), together with the audio modulating signal (at TP1), triggering the scope with the audio modulating signal. Note that the DSBSC waveform appears, amplified slightly at TP13, as we will see later, it is the output amplifier's output signal which will be transmitted to the receiver.

C. Setup for Double Sideband AM Reception

Procedure:

1. Position the **ST2201** & **ST2202** modules, with the **ST2201** board on the left, and a gap of about three inches between them.

2. Ensure that the following initial conditions exist on the **ST2201** board.

- **a.** Audio oscillator's amplitude pot in fully clockwise position.
- **b.** Audio input select switch in INT position.
- **c.** Balance pot in balanced modulator & band pass filter circuit 1 block, in full clockwise position;
- **d.** Mode switch in DSB position.
- **e.** Output amplifier's gain pot in full counter-clockwise position.
- **f.** TX output select switch in ANT position:
- **g.** Audio amplifier’s volume pot in fully counter-clockwise position.
- **h.** Speaker switch in ON position.
- **i.** On-board antenna in vertical position, and fully extended.

3. Ensure that the following initial conditions exist on the **ST2102** board:

- **a.** RX input select switch in ANT position.
- **b.** R.F. amplifier's tuned circuit select switch in INT position.
- **c.** R.E amplifier's gain pot in fully clock-wise position;
- **d.** AGC switch in INT position.
- **e.** Detector switch in diode position.
- **f.** Audio amplifier's volume pot in fully counter-clockwise position.
- **g.** Speaker switch in ON position.
- **h.** Beat frequency oscillator switch in OFF position.
- **i.** On-board antenna in vertical position, and fully extended.

4. Turn on power to the modules.

5. The first stage or 'front end' of the **ST2202** AM receiver is the R.F amplifier stage. This is a wide -bandwidth tuned amplifier stage, which is tuned into the wanted station by means of the tuning dial.

Once it has been tuned into the wanted station, the R.F. amplifier, having little selectivity, will not only amplify, but also those frequencies that are close to the wanted frequency. As we will see later, these nearby frequencies will be removed by subsequent stages of the receiver, to leave only the wanted signal. Examine the envelope of the signal at the R.F. amplifier's output (at TP12), with an a.c. - coupled oscilloscope channel. Note that :

a. The amplifier's output signal is very small in amplitude (a few tens of millivolts at the most). This is because one stage of amplification is not sufficient to bring the signal's amplitude up to a reasonable level.

b. Only a very small amount of amplitude modulation can be detected, if any. This is because there are many unwanted frequencies getting through to the amplifier output, which tend to 'drown out' the wanted AM Signal.

You may notice that the waveform itself drifts up and down on the scope display, indicating that the waveform's average level is changing. This is due to the operation of the AGC circuit.

6. The next stage of the receiver is the mixer stage, which mixes the R.F. amplifier's output with the output of a local oscillator. The Frequency of the local oscillator is also tuned by means of the tuning dial, and is arranged so that its frequency is always 455 KHz above the signal frequency that the R.F. amplifier is tuned to. This fixed frequency difference is always present, irrespective of the position of the tuning dial, and is arranged so that its frequency is always 455 KHz above the signal frequency that the R.F. amplifier is tuned to. This fixed frequency difference is always present, irrespective of the position of the tuning dial, and is known as the intermediate frequency (IF for short). This frequency relationship is shown below, for some arbitrary position of the tuning dial.

Examine the output of the local oscillator block, and check that its frequency varies as the tuning dial is turned. Re-time the receiver to a radio station.

7. The operation of the mixer stage is basically to shift the wanted signal down to the IF frequency, irrespective of the position of the tuning dial. This is achieved in two stages.

a. By mixing the local oscillator's output sine wave with the output from the R.F. amplifier block. This produces three frequency components :

The local oscillator frequency = (f sig + IF)

The sum of the original two frequencies, f sum = (2 f sig + IF)
The difference between the original two frequencies
f diff = (f sig + IF - f sig) = IF

b. By strongly attenuating all components except the difference frequency, IF this is done by putting a narrow-bandwidth band pass filter on the mixer's output.

The end result of this process is that the carrier frequency of the selected AM station is shifted down to 455 KHz (the IF Frequency), and the sidebands of the AM signal are now either side of 455 KHz.

8. Note that, since the mixer's band pass filter is not highly selective, it will not completely remove the local oscillators and sum frequency components from the mixer's output. This is the case particularly with the local oscillator component, which is much larger in amplitude than the sum and difference components.

Examine the output of the mixer block (TP20) with an a.c. coupled oscilloscope channel, and note that the main frequency component present changes as the tuning dial is turned. This is the local oscillator component, which still dominates the mixer's output, in spite of being attenuated by the mixer's band pass filter.

9. Tune in to a strong broadcast station again and note that the monitored signal shows little, if any, sign of modulation. This is because the wanted component, which is now at the IF frequency of 455 KHz, is still very small in component, which is now at the IF frequency of 455 KHz, is still very small in comparison to the local oscillator component.

Examine the output of IF amplifier1 (at. TP24) with an a.c.-coupled oscilloscope channel, and note that:

- The overall amplitude of the signal is much larger than the signal amplitude at the mixer's output, indicating that voltage amplification has occurred.
- The dominant component of the signal is now at 455 KHz, irrespective of any particular station you have tuned into. This implies that the wanted signal, at the IF frequency, has been amplified to a level where it dominates over the unwanted components.
- The envelope of the signal is modulated in amplitude, according to the sound information being transmitted by the station you have tuned into.

10. Examine the output of IF amplifier 2 (TP28) with an a.c.-coupled oscilloscope channel, noting that the amplitude of the signal has been further amplified by this second IF amplitude of the signal has been further amplified by this second IF amplifier stage.

IF amplifier 2 has once again preferentially amplified signals around the IF frequency (455 KHz), so that:

- The unwanted local oscillator and sum components from the mixer are now so small in comparison, that they can be ignored totally,
- Frequencies close to the I F frequency, which are due to stations close to the wanted station, are also strongly attenuated.

The resulting signal at the output of IF amplifier 2 (TP28) is therefore composed almost entirely of a 455 KHz carrier, and the A.M. sidebands either side of it carrying the wanted audio information.

11. The next step is extract this audio information from the amplitude variations of the signal at the output of IF amplifier 2. This operation is performed by the diode detector block, whose output follows the changes in the amplitude of the signal at its input.

To see how this works, examine the output of the diode detector block (TP31), together with the output from. IF amplifier 2 (at tp28). Note that the signal at the diode detector's output:

- Follows the amplitude variations of the incoming signal as required.
- Contains some ripple at the IF frequency of 455 KHz, and
- The signal has a positive DC offset, equal to half the average peak to peak amplitude of the incoming signal. We will see how we make use of this offset later on, when we look at automatic gain control (AGC).

Frequently Asked Questions

Que 1. Define DSB-SC.

Ans. After modulation, the process of transmitting the sidebands (USB, LSB) alone and suppressing the carrier is called as Double Side Band-Suppressed Carrier.

Que 2. What are the disadvantages of DSB-FC?

Ans.(i) Power wastage takes place in DSB-FC

(ii) DSB-FC is bandwidth inefficient system.

Que 3.Define Coherent Detection.

Ans. During Demodulation carrier is exactly coherent or synchronized in both the frequency and phase, with the original carrier wave used to generate the DSB-SC wave. This method of detection is called as coherent detection or synchronous detection.

Que.4.How will you generating DSBSC-AM?

Ans.There are two ways of generating DSBSC-AM such as

a).Balanced modulator

b).Ring modulators