LJ Technical Systems

ANACOM 1
DSB/SSB AM
Transmitter/Receiver
User Manual

MT106/D

DiGiAC

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Written by: LJ Technical Publications Dept

LJ Technical Systems Ltd.
Francis Way
Bowthorpe Industrial Estate
Norwich. NR5 9JA. England
Telephone: (01603) 748001
Telex: 975504
Fax: (01603) 746340

LJ Technical Systems Inc.
85 Corporate Drive, Holtsville
11742-2007, New York, USA
Telephone: (516) 234 2100
Fax: (516) 234 2656

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Introduction Chapter 1

ANACOM 1 Double/Single Sideband AM Transmitter/Receiver System

Features:

- Separate transmitter and receiver boards (ANACOM 1/1 and ANACOM 1/2 respectively);
- On-board transmitter audio oscillator for use as analog signal source.
- Transmitter generates double-sideband AM (with and without a carrier component), and single-sideband AM waveforms.
- Communication between transmitter and receiver may be via the on-board antennas, or by means of a shielded wire link.
- Receiver is of superheterodyne construction, and receives transmissions throughout the medium-wave AM broadcast band.
- Receiver includes a diode detector for demodulation of double sideband (unsupressed carrier) AM, and a product detector for single-sideband AM demodulation.
- A total of sixteen switched faults affect the operation of the transmitter and receiver, allowing the student to investigate thoroughly all aspects of system operation.

POWER SUPPLY REQUIREMENTS

The power supply requirements of the ANACOM 1/1 module are:

+12 volts at 200mA and -12volts at 50mA

The power supply requirement of the ANACOM 1/2 module is +12 volts at 200mA.

OUTLINE OF THE ANACOM 1/1 TRANSMITTER

ANACOM 1/1 is a radio frequency transmitter capable of generating double-sideband and single-sideband amplitude-modulated waveforms. The layout of the board is shown in Figure 1 below:

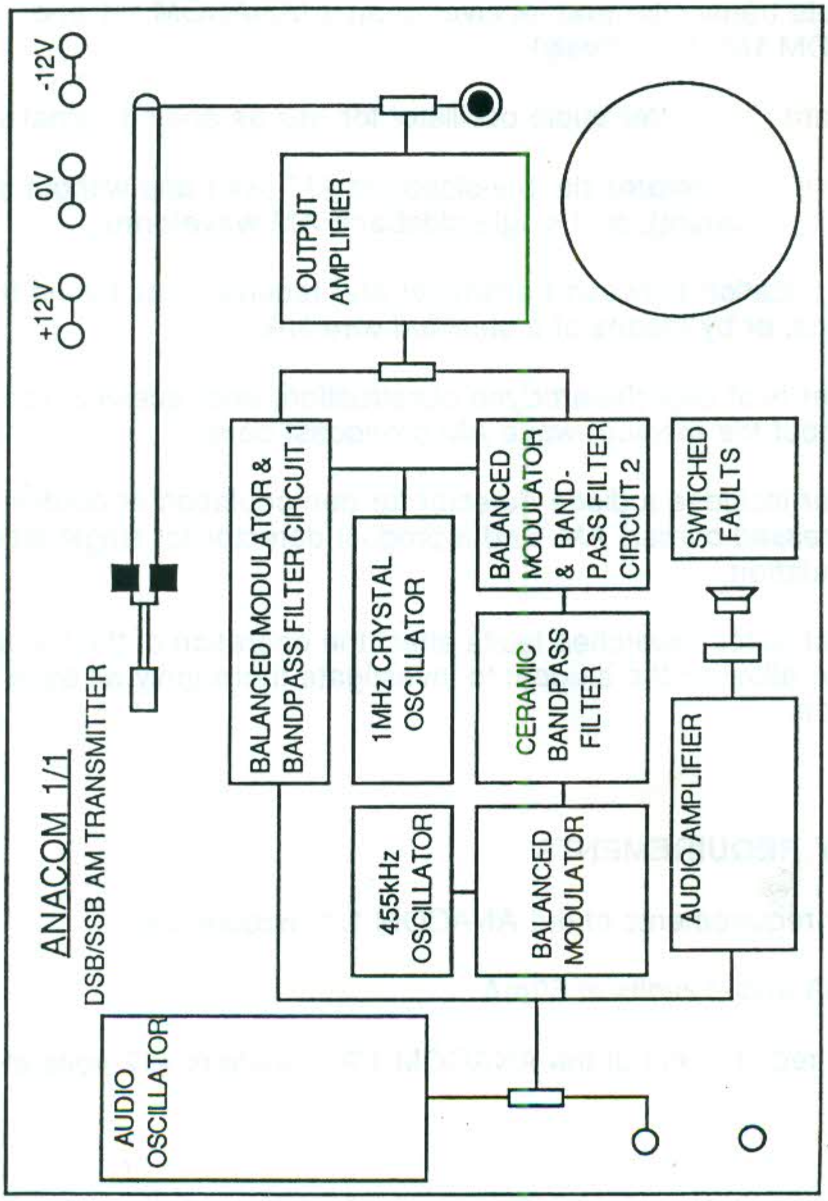


Figure 1

The output from the on-board audio oscillator may be used as a modulating signal. The amplitude of this signal is adjustable, as is the frequency (from 300Hz to 3.4kHz). Alternatively, an external audio-frequency signal may be used as the modulating input signal.

An on-board audio amplifier and loudspeaker allow monitoring of the audio modulating signal. Headphones socket allows the signal to be monitored using a pair of headphones (supplied). Volume of audio output is adjustable by means of the amplifier's volume control.

On-board Double-Sideband Modulator generates a double-sideband AM signal with a carrier frequency of 1MHz. Modulator balance control allows the carrier to be suppressed, to illustrate Double-Sideband Suppressed Carrier (DSBSC) Amplitude Modulation.

On-board Single-Sideband Modulator generates a single-sideband AM signal with a carrier frequency of 1.455MHz. Modulator balance controls allow carrier to be suppressed.

Output amplifier stage can be switched to amplify the output of either the Double-Sideband Modulator or Single-Sideband Modulator. Gain of output amplifier is adjustable.

Final transmitter output may either be linked to the ANACOM 1/2 receiver directly by means of the screened cable supplied, or radiated from the antenna on the board.

The on-board antenna is telescopic, and may be angled and rotated. It folds down into a transit clip mounted on the board. When fully extended, a transmission range of up to 4 feet is obtainable.

All of the transmitter's functional blocks are shown on the PCB mimic, and numbered test points provide access to the main signals on the board. Eight switched faults (4 open-circuit, 4 short-circuit) affect the operation of the major functional blocks within the transmitter.

OUTLINE OF THE ANACOM 1/2 RECEIVER

ANACOM 1/2 is a radio-frequency receiver capable of receiving and demodulating the double-sideband and single-sideband AM transmissions from the ANACOM 1/1 transmitter. In addition, ANACOM 1/2 is capable of receiving double-sideband AM broadcast signals in the AM waveband.

The layout of the board is shown in Figure 2:

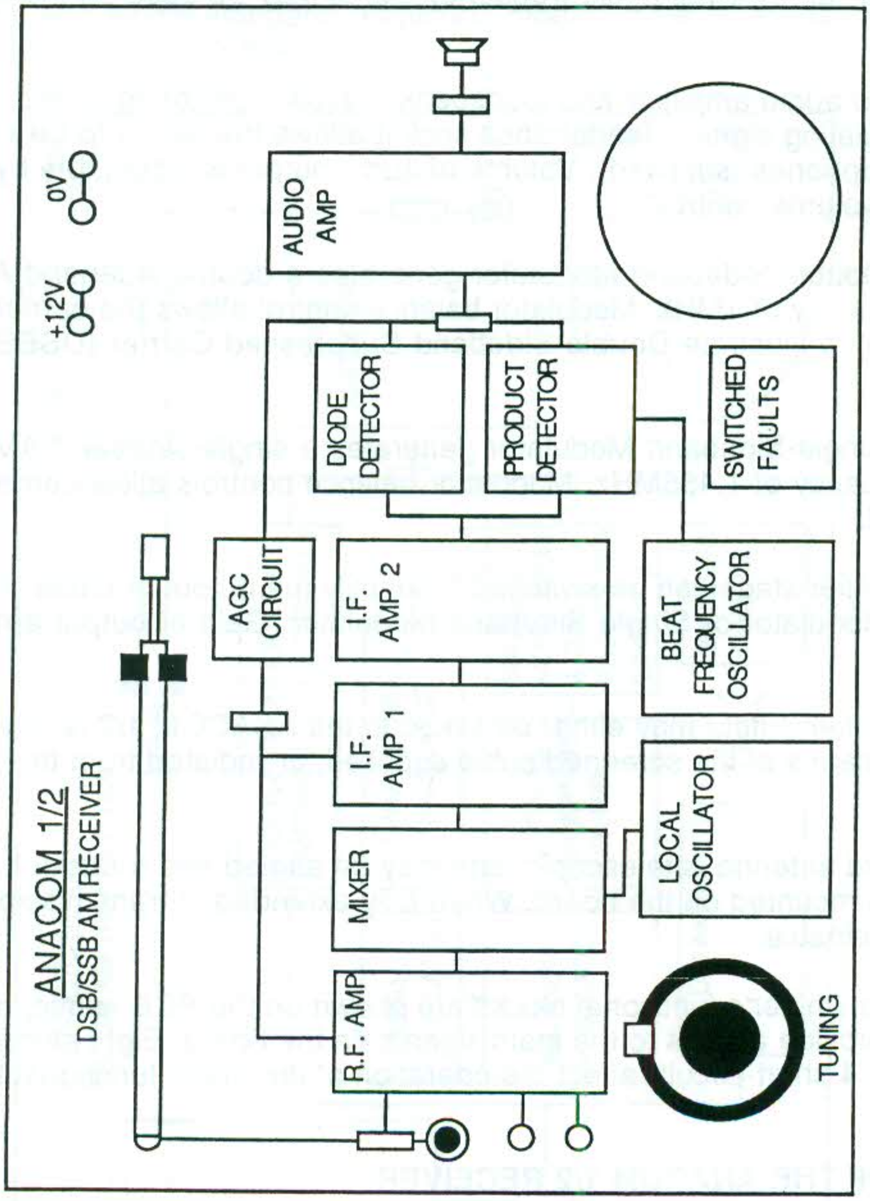


Figure 2

The ANACOM 1/2 receiver has an on-board antenna for the reception of transmissions from the ANACOM 1/1 transmitter, and the reception of AM broadcast signals. The ANACOM 1/1 transmitter and ANACOM 1/2 receiver may also be linked by means of the screened cable supplied.

ANACOM 1/2 is a superheterodyne receiver with two on-board detectors: a diode detector for demodulation of double-sideband (unsuppressed carrier) AM signals, and a product detector for single-sideband AM signals.

Automatic Gain Control (AGC) may be switched in or out to compare the operation of the receiver with, and without, AGC.

The RF amplifier stage's tuned circuit may be disconnected, allowing the ouput from the L.J. Medium Wave Tuned Circuit Module to be fed to the input of the RF amplifier. This allows comparisons to be drawn between the performance of ANACOM 1/2's telescopic antenna, and that of the Tuned Circuit Module's ferrite rod antenna.

The gain of ANACOM 1/2's RF stage is adjustable, and a vernier tuning dial is provided for accurate receiver tuning.

ANACOM 1/2's on-board antenna is telescopic, and may be angled and rotated. It folds down into the transit clip mounted on the board. When fully extended, transmissions from ANACOM 1/1 may be detected with the transmitter and receiver up to 4 feet apart.

The output from either the diode detector or the product detector may be used to drive the on-board audio amplifier and loudspeaker. A headphone socket allows monitoring of the detector output by means of a pair of headphones (supplied). Volume of the audio output is adjustable by means of the amplifier's volume control.

All radio-frequency transformers and trimmer capacitors are accessible, allowing the procedure for 'setting up' the radio receiver's tuned circuits to be thoroughly investigated.

All of the receiver's functional blocks are shown on the PCB mimic, with numbered test points for easy examination of the main receiver signals. Eight switched faults (4 open, 4 closed), are provided on the board, affecting the operation of the major functional blocks within the receiver.

GENERATION OF DOUBLE SIDEBAND AM WAVEFORMS

This experiment investigates the generation of Double Sideband amplitude modulated (AM) waveforms, using the ANACOM 1/1 module. By removing the carrier from such an AM waveform, the generation of Double Sideband Suppressed Carrier (DSBSC) AM is also investigated.

To avoid unnecessary loading of monitored signals, X10 oscilloscope probes should be used throughout this experiment.

EXPERIMENTATION

1. Connect the ANACOM 1/1 module to the power supply as shown below:

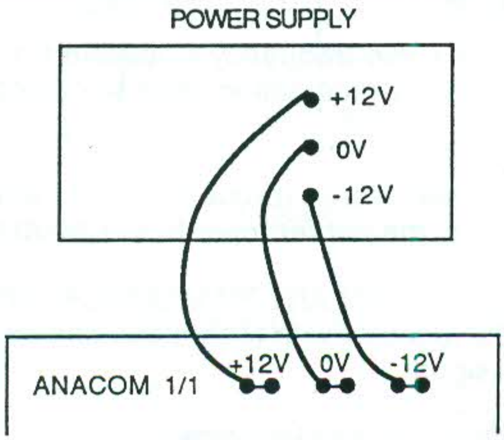


Figure 3

2. Ensure that the following initial conditions exist on the board:

   (a) AUDIO INPUT SELECT switch in INT position;
   (b) MODE switch in DSB position;
   (c) OUTPUT AMPLIFIER's GAIN preset in fully **clockwise** position;
   (d) SPEAKER switch in OFF position.

3. Turn on power to the ANACOM 1/1 board.

4. Turn the AUDIO OSCILLATOR block's AMPLITUDE preset to its fully clockwise (MAX) position, and examine the block's output (t.p.14) on an oscilloscope.

This is the audio frequency sinewave which will be used as our modulating signal. Note that the sinewave's frequency can be adjusted from about 300Hz to approximately 3.4kHz, by adjusting the AUDIO OSCILLATOR's FREQUENCY preset.

Note also that the amplitude of this audio modulating signal can be reduced to zero, by turning the AUDIO OSCILLATOR's AMPLITUDE preset to its fully **counter-clockwise** (MIN) position.

Return the AMPLITUDE preset to its MAX position before continuing.

5. Turn the BALANCE preset, in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, to its fully **clockwise** position. It is this block that we will use to perform **double-sideband amplitude modulation**.

6. Monitor, in turn, the two inputs to the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, at t.p.1 and t.p.9. Note that:

   (a) The signal at t.p.1 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 sinewave of frequency 1MHz and amplitude 120mV pk/pk approx. This is the carrier input to our double-sideband modulator.

7. Next, examine the output of the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block (at t.p.3), together with the modulating signal at t.p.1. Trigger the oscilloscope on the t.p.1 signal.

   Check that the waveforms are as shown below:

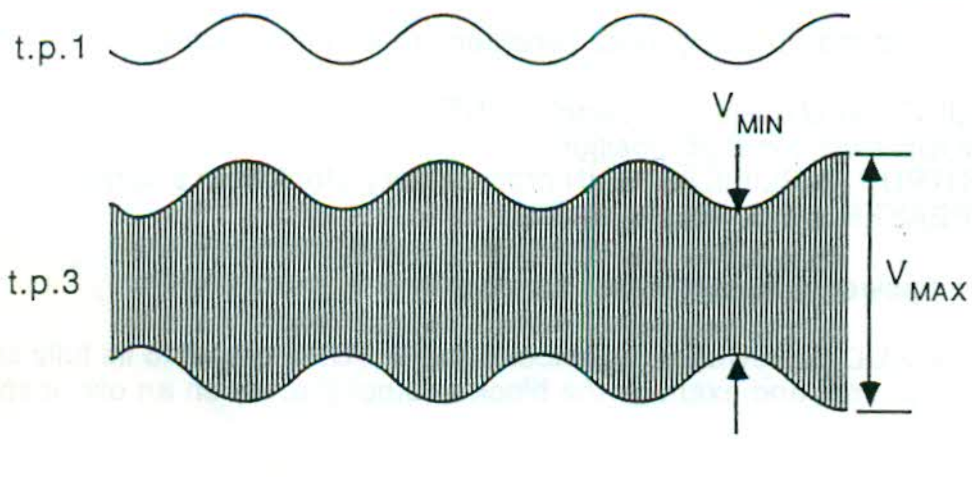


Figure 4

The output from the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block (at t.p.3) is a double-sideband AM waveform, which has been formed by amplitude-modulating the 1MHz carrier sinewave with the audio-frequency sinewave from the audio oscillator.

The frequency spectrum of this AM waveform is as shown below, where **fm** is the frequency of the audio modulating signal:

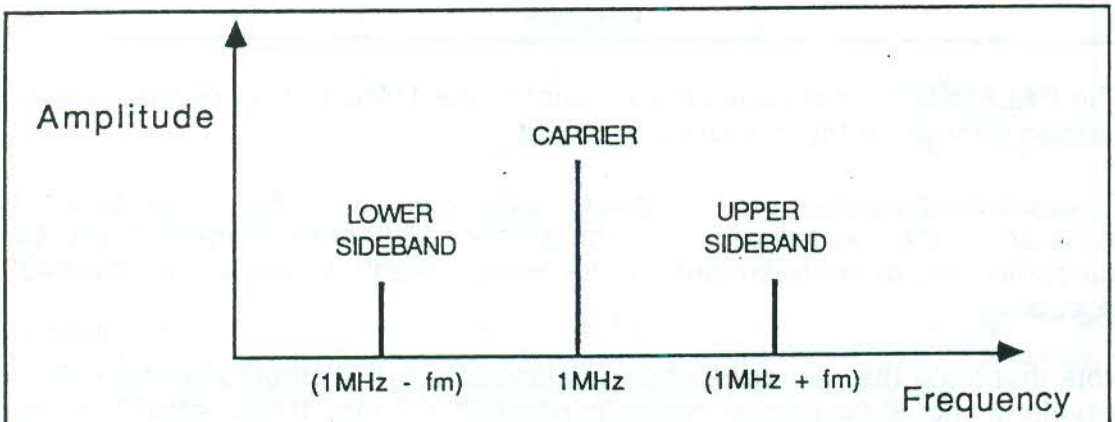


Figure 5

8. To determine the depth of modulation, measure the maximum amplitude (V_{MAX}) and the minimum amplitude (V_{MIN}) of the AM waveform at t.p.3, and use the following formula:

   $$\text{Percentage Modulation} = \frac{V_{MAX} - V_{MIN}}{V_{MAX} + V_{MIN}} \times 100\%$$

   where V_{MAX} and V_{MIN} are the maximum and minimum amplitudes shown in Fig 4.

9. Now vary the amplitude and frequency of the audio-frequency sinewave, by adjusting the AMPLITUDE and FREQUENCY presets in the AUDIO OSCILLATOR block. Note the effect that varying each preset has on the amplitude modulated waveform.

   The 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 preset to its MIN position, and note that the signal at t.p.3 becomes an unmodulated sinewave of frequency 1MHz, indicating that only the carrier component now remains.

   Return the AMPLITUDE preset to its maximum position before continuing.

10. Now turn the BALANCE preset in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, until the signal at t.p.3 is as shown below:

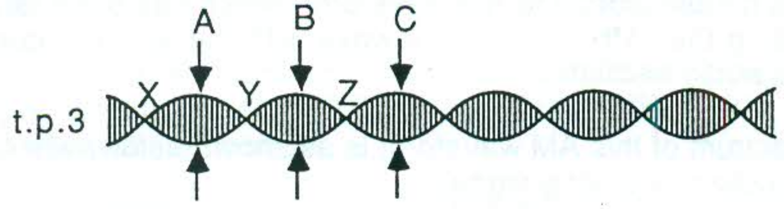


Figure 6

The BALANCE preset varies the amount of the 1MHz carrier component which is passed through to the modulator's output.

By adjusting the preset 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 t.p.3'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 bandpass filter in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block must be adjusted so that it passes both sidebands equally. This is achieved by carefully trimming transformer T1 (using the trimmer tool provided), until the waveform's amplitude is as close to zero as possible at the minimum points.

The waveform at t.p.3 is known as a **double-sideband suppressed carrier** (DSBSC) waveform, and its frequency spectrum is as shown overleaf:

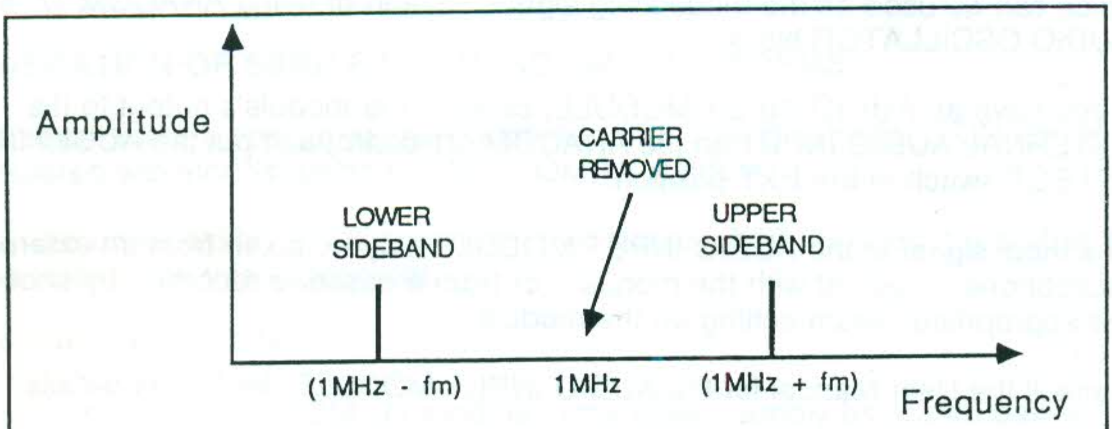


Figure 7

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

10. Change the amplitude and frequency of the modulating audio signal (by adjusting the AUDIO OSCILLATOR block's AMPLITUDE and FREQUENCY presets), and note the effect that these changes have 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 this by turning the AMPLITUDE preset 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 preset to its MAX position before continuing.

11. Examine the output from the OUTPUT AMPLIFIER block (t.p.13), together with the audio modulating signal (at t.p.1), triggering the 'scope with the latter. Note that the DSBSC waveform appears, amplified slightly, at t.p.13. As we will see later, it is the OUTPUT AMPLIFIER's output signal which will be transmitted to the receiver.

12. By using the optional AUDIO INPUT MODULE (L.J. Order Code CT7), the human voice can be used as the modulating signal, instead of using ANACOM 1/1's AUDIO OSCILLATOR block.

    If you have an AUDIO INPUT MODULE, connect the module's output to the EXTERNAL AUDIO INPUT on the ANACOM 1/1 board, and put the AUDIO INPUT SELECT switch in the EXT position.

    The input signal to the AUDIO INPUT MODULE may be taken from an external microphone (supplied with the module), or from a cassette recorder, by choosing the appropriate switch setting on the module.

    Consult the User Manual for the AUDIO INPUT MODULE, for further details.

GENERATION OF SINGLE SIDEBAND AM WAVEFORMS

This experiment investigates the generation of Single Sideband (SSB) amplitude modulated waveforms, using the ANACOM 1/1 module.

To avoid unnecessary loading of monitored signals, X10 oscilloscope probes should be used throughout this experiment.

EXPERIMENTATION

1. Connect the ANACOM 1/1 module to the power supply as shown below:

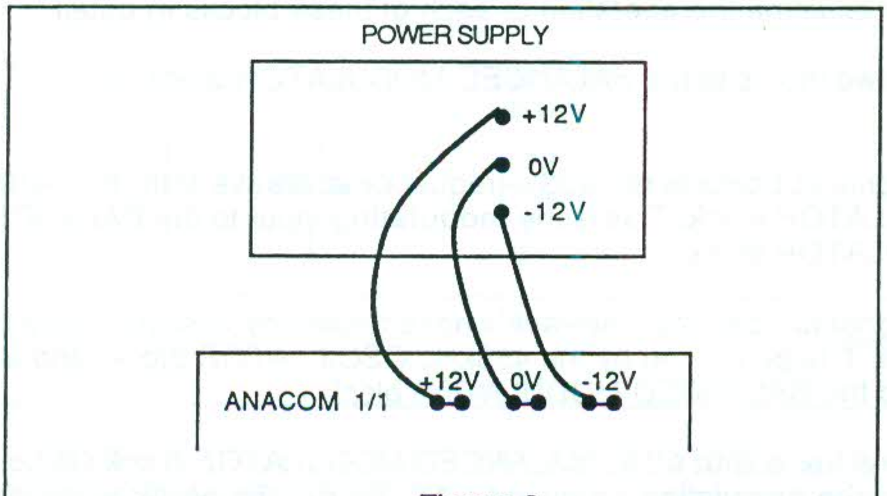


Figure 8

2. Ensure that the following initial conditions exist on the board:

   (a) AUDIO INPUT SELECT switch in INT position;
   (b) MODE switch in SSB position;
   (c) OUTPUT AMPLIFIER's GAIN preset in fully **clockwise** position;
   (d) SPEAKER switch in OFF position.

3. Turn on power to the ANACOM 1/1 board.

4. Turn the AUDIO OSCILLATOR block's AMPLITUDE preset to its fully **clockwise** (MAX) position, and examine the block's output (t.p.14) on an oscilloscope.

   This is the audio frequency sinewave which will be used as our modulating signal. Note that the sinewave's frequency can be adjusted from about 300Hz to approximately 3.4kHz, by adjusting the AUDIO OSCILLATOR's FREQUENCY preset.

Note also that the amplitude of this audio modulating signal can be reduced to zero, by turning the AUDIO OSCILLATOR's AMPLITUDE preset to its fully **counter-clockwise** (MIN) position.

Leave the AMPLITUDE preset in its fully **clockwise** position, and adjust the FREQUENCY preset for an audio frequency of 2kHz, before continuing.

5. To achieve single-sideband amplitude modulation, we will utilize the following three blocks on the ANACOM 1/1 module:

   BALANCED MODULATOR
   CERAMIC BANDPASS FILTER
   BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2

   We will now examine the operation of each of these blocks in detail.

6. Monitor the two inputs to the BALANCED MODULATOR block, at t.p.15 and t.p.6, noting that:

   (a) The signal at t.p.15 is the audio-frequency sinewave from the AUDIO OSCILLATOR block. This is the **modulating** input to the BALANCED MODULATOR block.

   (b) The signal at t.p.6 is a sinewave whose frequency is slightly less than 455kHz. It is generated by the 455kHz OSCILLATOR block, and is the **carrier** input to the BALANCED MODULATOR block.

7. Next, examine the output of the BALANCED MODULATOR block (at t.p.17), together with the modulating signal at t.p.15. Trigger the oscilloscope on the modulating signal.

   Check that the waveforms are as shown below:

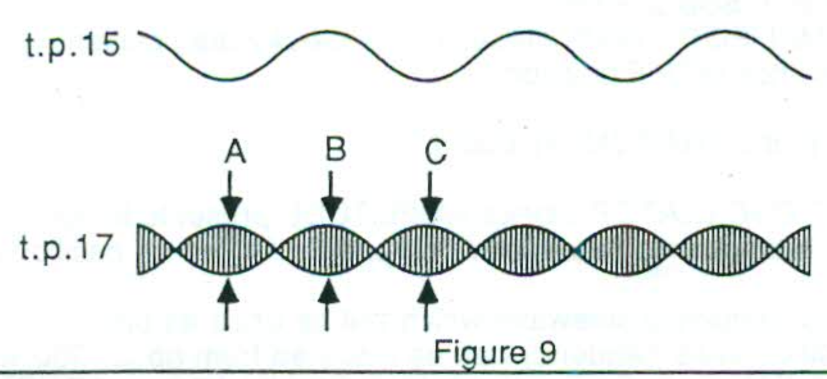


Figure 9

Note that it may be necessary to adjust the BALANCED MODULATOR block's BALANCE preset, in order to ensure that the peaks of t.p.17's waveform envelope (labelled A,B,C, etc in the above diagram) all have equal amplitude.

You will recall that the waveform at t.p.17 was encountered in the previous experiment - this is a **double-sideband suppressed carrier** (DSBSC) AM waveform, and it has been obtained by amplitude-modulating the carrier sinewave at t.p.6 (of frequency f_c) with the audio-frequency modulating signal at t.p.15 (of frequency f_m), and then removing the carrier component from the resulting AM signal, by adjusting the BALANCE preset.

The frequency spectrum of this DSBSC waveform is shown below:

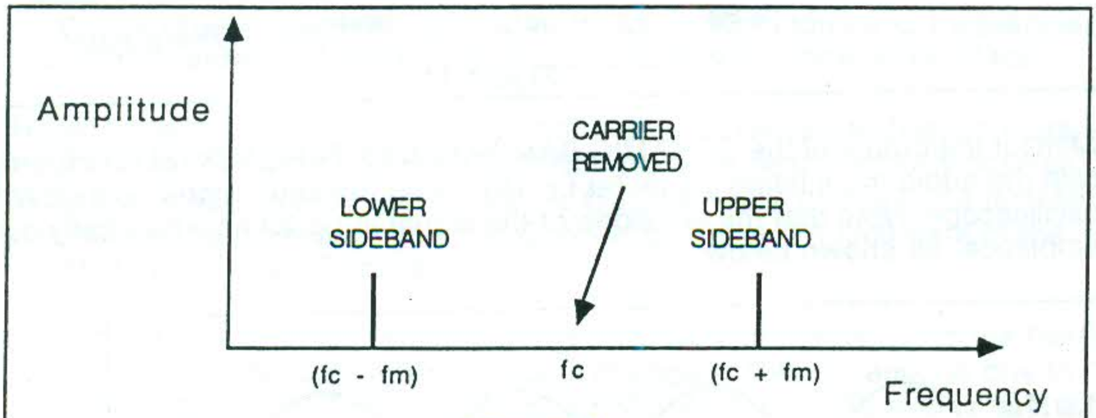


Figure 10

8. The DSBSC output from the BALANCED MODULATOR block is next passed on to the CERAMIC BANDPASS FILTER block, whose purpose is to **pass** the **upper sideband**, but **block** the **lower sideband**. We will now investigate how this is achieved.

   First note that the ceramic bandpass filter has a narrow passband of only a few kilohertz, centered around 455kHz.

   It was mentioned earlier that the frequency of the carrier input to the BALANCED MODULATOR block has been arranged to be **slightly less than** 455kHz. In fact, the carrier frequency is carefully chosen so that, whatever the modulating frequency f_m, the upper sideband (at f_c+f_m) will fall inside the filter's passband, while the lower sideband (at f_c-f_m) always falls outside.

   Consequently, the upper sideband will suffer little attenuation, while the lower sideband will be heavily attenuated to such an extent that it can be ignored.

This is shown in the frequency spectrum below:

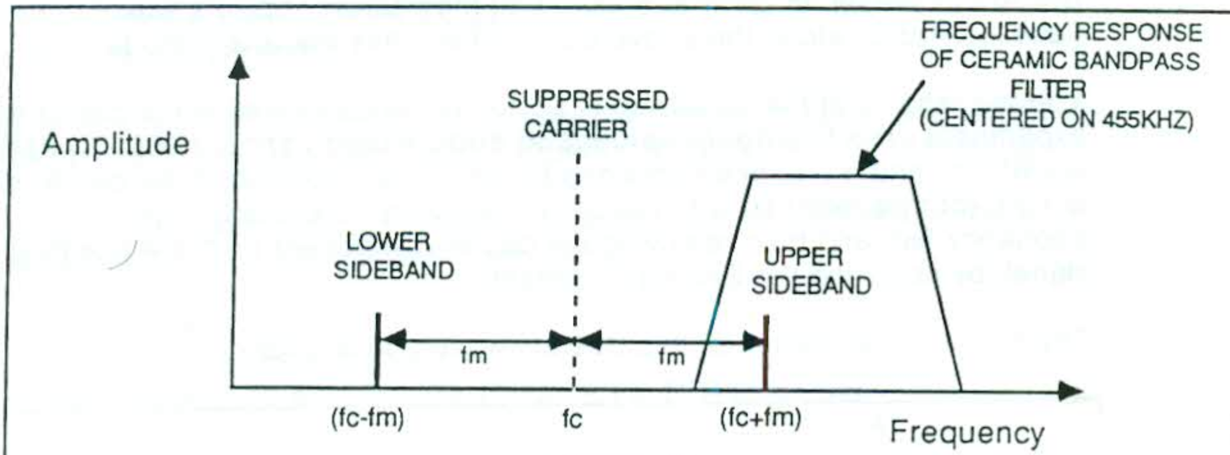


Figure 11

9. Monitor the output of the CERAMIC BANDPASS FILTER block (at t.p.20), together with the audio modulating signal (at t.p.15), using the latter signal to trigger the oscilloscope. Note that the envelope of the signal at t.p.20 now has fairly constant amplitude, as shown below:

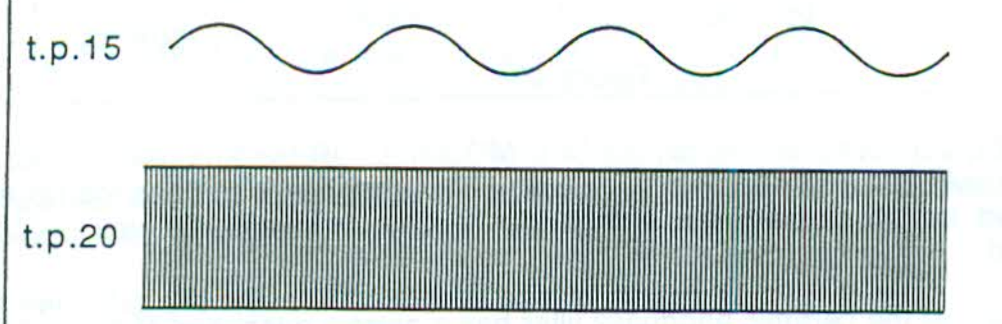


Figure 12

If the amplitude of the signal at t.p.20 is **not** reasonably constant, adjust the BALANCE preset in the BALANCED MODULATOR block to minimize variations in the signal's amplitude.

If a constant-amplitude waveform **still** cannot be obtained, then the frequency of the 455kHz OSCILLATOR needs to be trimmed. To do this, follow the procedure given in Chapter 8, entitled 'Adjustment of the Transmitter's Tuned Circuits'.

10. Now trigger the oscilloscope with the CERAMIC BANDPASS FILTER's output signal (t.p.20), and note that the signal is a good, clean sinewave, indicating that the filter has passed the upper sideband only.

    Next, turn the AUDIO OSCILLATOR block's FREQUENCY preset throughout its range. Note that for most audio frequencies, the waveform is a good, clean sinewave, indicating that the lower sideband has been totally rejected by the filter.

    For low audio frequencies, you may notice that the monitored signal is not such a pure sinusoid. This is because the upper and lower sidebands are now very close to each other, and the filter can no longer completely remove the lower sideband. Nevertheless, the lower sideband's amplitude is sufficiently small compared with the upper sideband, that its presence can be ignored.

    Since the upper sideband dominates for all audio modulating frequencies, we say that **single-sideband (SSB) amplitude-modulation** has taken place.

    Note: If the monitored waveform is **not** a good sinewave at **higher** modulating frequencies (i.e. when the FREQUENCY preset is near the MAX position), then it is likely that the frequency of the 455kHz OSCILLATOR needs to be trimmed. To do this, follow the procedure given in Chapter 8, entitled 'Adjustment of the Transmitter's Tuned Circuits'.

11. Note that there is some variation in the amplitude of the signal at the filter's output (t.p.20), as the modulating frequency is changed. This variation is due to the frequency response of the CERAMIC BANDPASS FILTER, and is best explained by considering the spectrum of the filter's input signal at the MIN and MAX positions of the FREQUENCY preset, as shown overleaf:

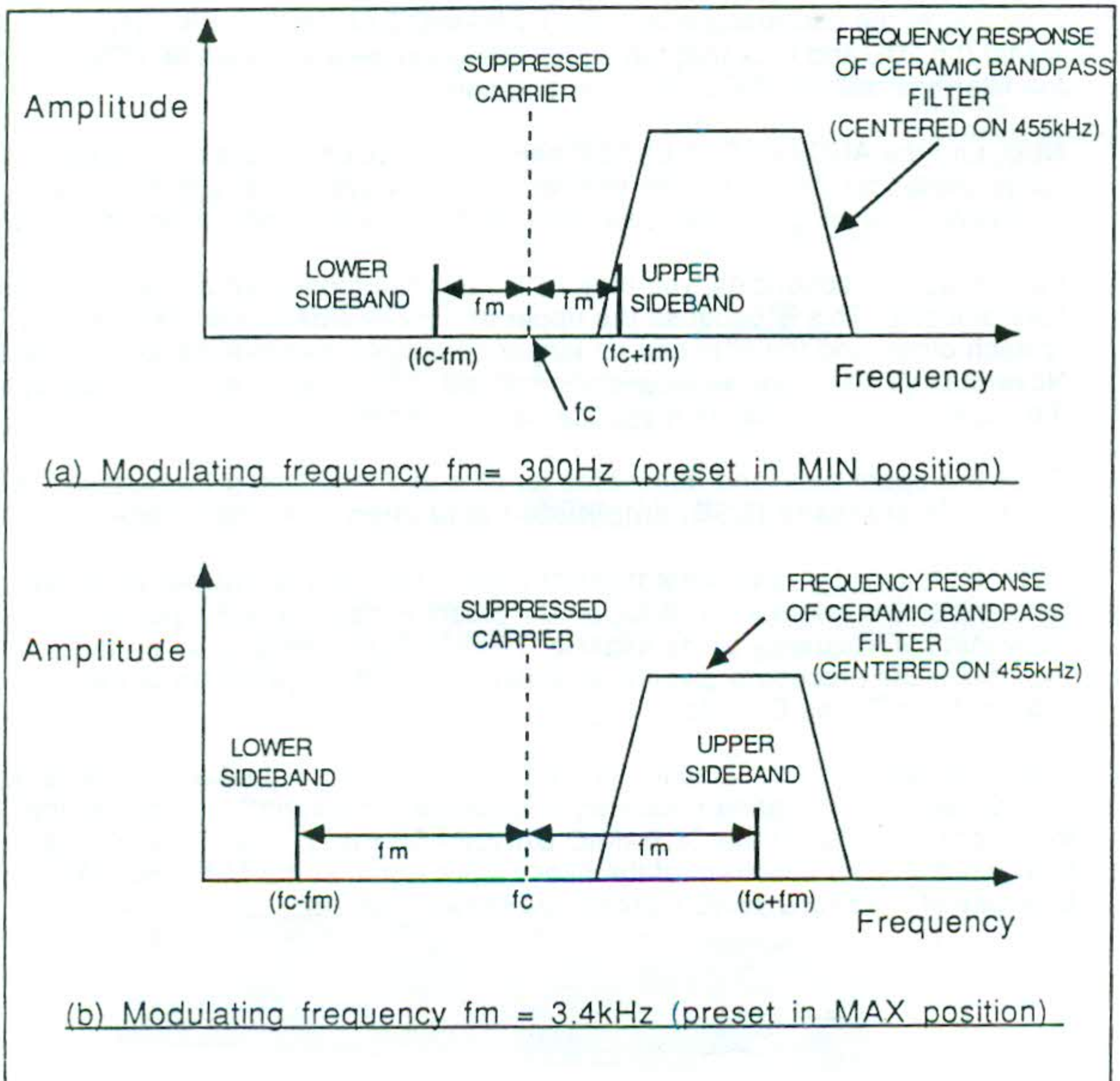
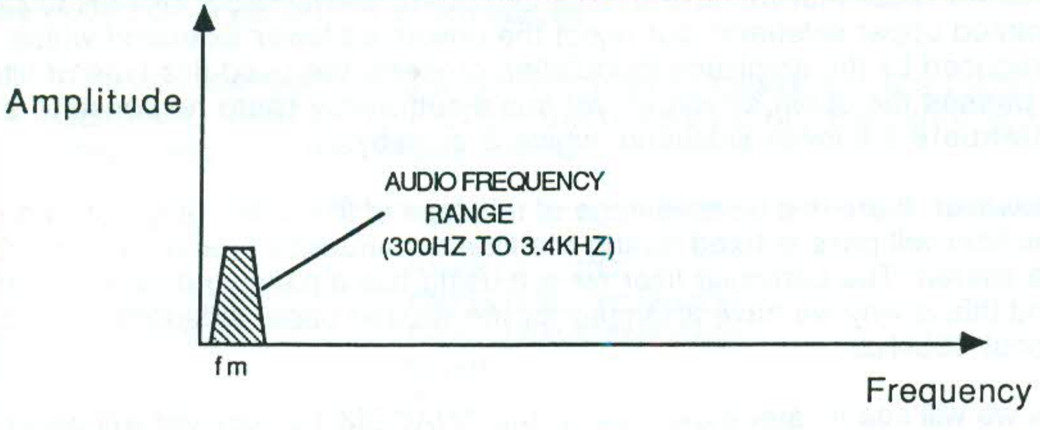


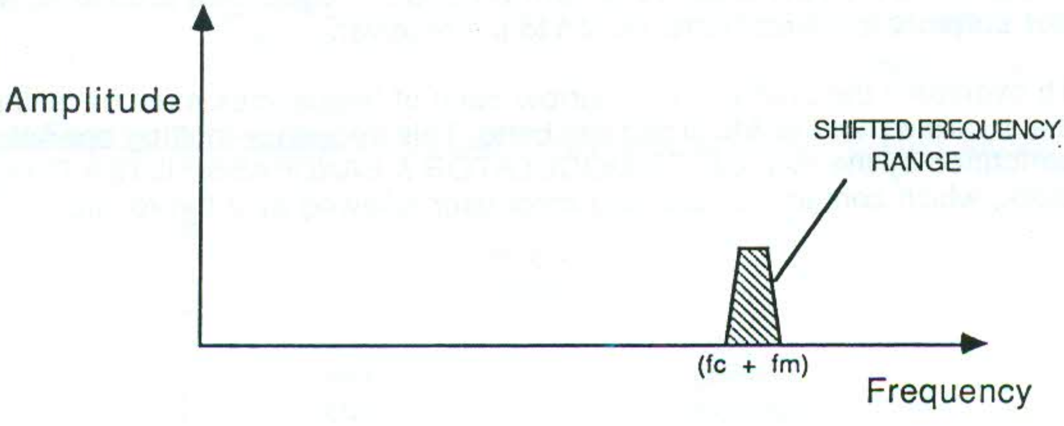
Figure 13

Notice that, since the upper sideband cuts the rising edge of the filter's frequency response when fm=300Hz, there will be a certain amount of signal attenuation when the FREQUENCY preset is in its MIN position.

12. Note that, by passing **only** the upper sideband (of frequency ($f_c + f_m$)), all we have actually done is to **shift** our audio modulating signal (of frequency f_m) **up** in frequency by an amount equal to the carrier frequency f_c. This is shown overleaf:



(a) Range of frequencies available from AUDIO OSCILLATOR



(b) Corresponding range of output frequencies from CERAMIC BANDPASS FILTER block

Figure 14

13. With the AUDIO OSCILLATOR block's FREQUENCY preset roughly in its midway position (arrowhead pointing towards the top of the PCB), turn the block's AMPLITUDE preset to its MIN position, and note that the amplitude of the signal at the CERAMIC BANDPASS FILTER's output (t.p.20) drops to zero.

    This highlights one of the main advantages of SSB amplitude modulation - if there is no modulating signal, then the amplitude of the SSB waveform drops to zero, so that no power is wasted.

    Return the AMPLITUDE preset to its MAX position before continuing.

14. You will recall that we have used a CERAMIC BANDPASS FILTER to pass the wanted upper sideband, but reject the unwanted lower sideband which was also produced by the amplitude modulation process. We used this type of filter because it **passes** the upper sideband, yet has a sufficiently sharp response to **strongly attenuate** the lower sideband, which is closeby.

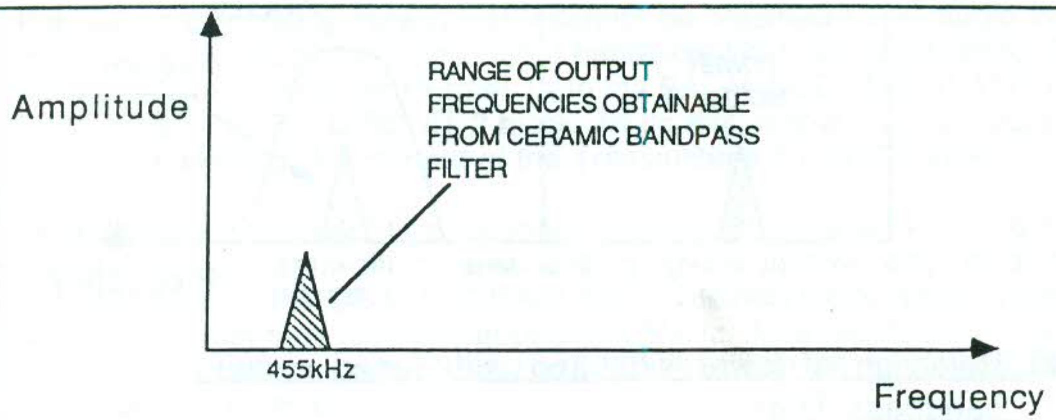
However, there is a disadvantage of this type of filter - the range of frequencies that the filter will pass is **fixed** during the filter's manufacture, and cannot subsequently be altered. The particular filter we are using has a passband centered on 455kHz, and this is why we have arranged for the wanted upper sideband to also be at about 455kHz.

As we will see in later experiments, the ANACOM 1/2 receiver will accept radio-frequency signals in the AM broadcast band, i.e. signals which fall in the frequency range of 525kHz to 1605kHz. However, since the SSB output from the CERAMIC BANDPASS FILTER occupies a narrow band of frequencies around 455kHz, it is **not suitable** for direct transmission to the receiver.

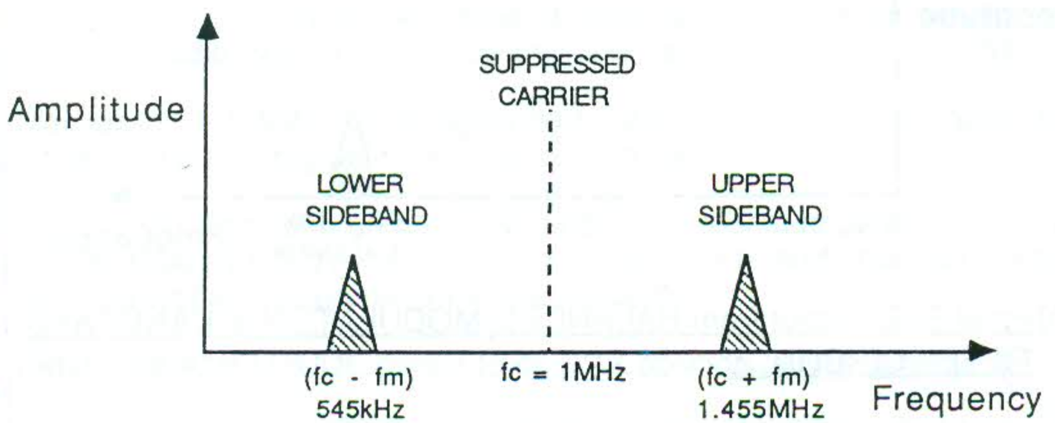
To overcome the problem, this narrow band of frequencies must be 'shifted up' so that it falls within the AM broadcast band. This frequency-shifting operation is performed by the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2 block, which contains a balanced modulator followed by a tuned circuit.

The operation is performed in two stages:

(1) By amplitude-modulating a 1MHz carrier sinewave with the output from the CERAMIC BANDPASS FILTER, and 'balancing out' the carrier component. This is shown in the diagram below:



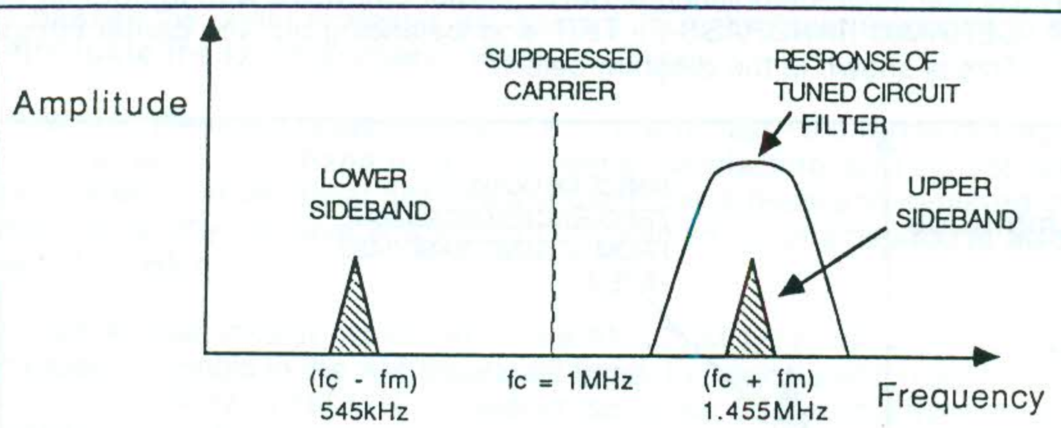
(a) Spectrum of output from CERAMIC BANDPASS FILTER block



(b) Spectrum obtained by modulating 1MHz carrier with output from CERAMIC BANDPASS FILTER

Figure 15

(2) By passing the Upper Sideband, and blocking the Lower Sideband, using a tuned circuit bandpass filter, as shown below:



(a) Rejection of lower sideband with tuned circuit bandpass filter.

(b) Final SSB output from BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2.

Figure 16

Note that since there is a large gap between the upper and lower sidebands (a gap of about 910kHz), a bandpass filter with a very sharp response is **not needed** to reject the lower sideband - a simple tuned-ciruit bandpass filter is quite sufficient.

15. Now examine the output of the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2 block (at t.p.22), and check that the waveform is a good sinewave of frequency approximately 1.455MHz. This indicates that only the upper sideband is being passed by the block.

Check that the waveform is a reasonably good sinusoid for all audio modulating frequencies (i.e. all positions of the AUDIO OSCILLATOR's FREQUENCY preset). I this is not the case, it may be that the BALANCE preset (in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2 block) needs adjusting, to remov any residual carrier component at 1MHz.

If a reasonably clean sinewave **still** cannot be obtained for all audio frequencies, then the response of the tuned circuit bandpass filter needs adjusting. This is achieved by adjusting transformer T4 in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2 block. To do this, follow the procedure given in Chapter 8, entitled 'Adjustment of the Transmitter's Tuned Circuits'.

Once the signal at t.p.22 **is** a reasonably good sinewave for all audio frequencies, we have achieved our objective of 'shifting up' the narrow range of output frequencies from the CERAMIC BANDPASS FILTER block (which were around 455kHz), so that they are now around 1.455MHz. As a result, they now fall within the AM broadcast range of 525kHz to 1.605MHz, and will be detectable by the ANACOM 1/2 receiver.

When the modulating audio signal is swept over its entire range (a range of 3.4kHz - 300Hz = 3.1kHz), the SSB waveform at t.p.22 sweeps over the **same** frequency range. So single-sideband modulation has simply served to **shift** our range of audio frequencies up so they are centered around 1.455MHz.

16. Monitor the 1.455MHz SSB signal (at t.p.22), together with the audio modulating signal (at t.p.15), triggering the 'scope with the latter.

    Reduce the amplitude of the audio modulating signal to zero (by means of the AUDIO OSCILLATOR block's AMPLITUDE preset), and note that the amplitude of the SSB signal also drops to zero, as expected.

    Return the AMPLITUDE preset to its MAX position before continuing.

17. Examine the final SSB output (at t.p.22) together with the output from the OUTPUT AMPLIFIER block (t.p.13). Note that the final SSB waveform appears, amplified slightly, at t.p.13. As we will see later, it is the OUTPUT AMPLIFIER's output signal which will be transmitted to the receiver.

18. By using the optional AUDIO INPUT MODULE (L.J. Order Code CT7), the human voice can be used as the audio modulating signal, instead of using ANACOM 1/1's AUDIO OSCILLATOR block.

    If you have an AUDIO INPUT MODULE, connect the module's output to the EXTERNAL AUDIO INPUT on the ANACOM 1/1 board, and put the AUDIO INPUT SELECT switch in the EXT position.

The input signal to the AUDIO INPUT MODULE may be taken from an external microphone (supplied with the module), or from a cassette recorder, by choosing the appropriate switch setting on the module.

Consult the User Manual for the AUDIO INPUT MODULE, for further details.

RECEPTION OF DOUBLE SIDEBAND AM WAVEFORMS

This experiment investigates the reception and demodulation of AM waveforms by the ANACOM 1/2 module. Both AM broadcast signals, and AM transmissions from ANACOM 1/1, will be examined, and the operation of Automatic Gain Control at the Receiver will be investigated.

To avoid unnecessary loading of monitored signals, X10 oscilloscope probes should be used throughout this experiment.

EXPERIMENTATION

1. Position the ANACOM 1/1 and 1/2 modules, with the ANACOM 1/1 board on the left, and a gap of about three inches between them. Then connect them to the power supply as shown below:

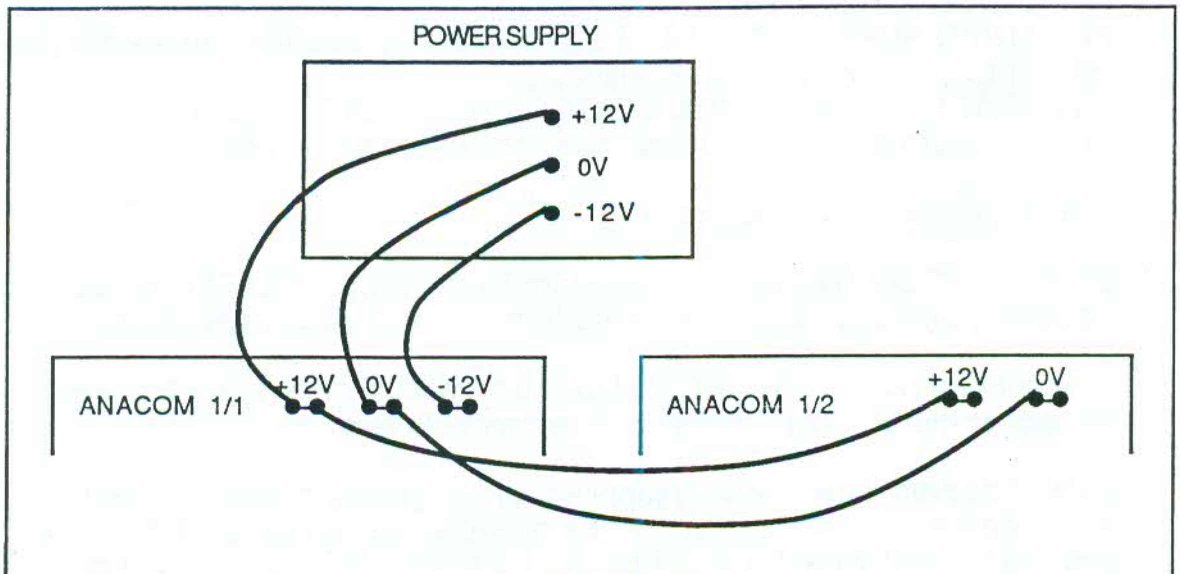


Figure 17

2. Ensure that the following initial conditions exist on the ANACOM 1/1 board:

   (a) AUDIO OSCILLATOR's AMPLITUDE preset in fully **clockwise** position;
   (b) AUDIO INPUT SELECT switch in INT position;
   (c) BALANCE preset in BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block, in fully **clockwise** position;
   (d) MODE switch in DSB position;
   (e) OUTPUT AMPLIFIER's GAIN preset in fully **counter-clockwise** position;
   (f) TX OUTPUT SELECT switch in ANT. position;
   (g) AUDIO AMPLIFIER's VOLUME preset 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 ANACOM 1/2 board:

   (a) RX INPUT SELECT switch in ANT. position;
   (b) R.F. AMPLIFIER's TUNED CIRCUIT SELECT switch in INT position;
   (c) R.F. AMPLIFIER's GAIN preset in fully **clockwise** position;
   (d) AGC switch in IN position;
   (e) DETECTOR switch in DIODE position;
   (f) AUDIO AMPLIFIER's VOLUME preset 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. On the ANACOM 1/2 module, slowly turn the AUDIO AMPLIFIER's VOLUME preset clockwise, until sounds can be heard from the on-board loudspeaker.

   Next, turn the vernier TUNING dial until a broadcast station can be heard clearly, and adjust the VOLUME control to a comfortable level.

   Note: if desired, headphones (supplied with the module) may be used instead of the on-board loudspeaker. To use the headphones, simply plug the headphone jack into the AUDIO AMPLIFIER block's HEADPHONES socket, and put the SPEAKER switch in the OFF position. The volume from the headphones is still controlled by the block's VOLUME preset.

6. The first stage, or 'front end', of the ANACOM 1/2 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 amplifiy the **wanted** frequency, but also those frequencies which 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 t.p.12), 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's 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, which will be explained later.

7. 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 455kHz **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** (I.F. for short). This frequency relationship is shown below, for some arbitrary position of the TUNING dial:

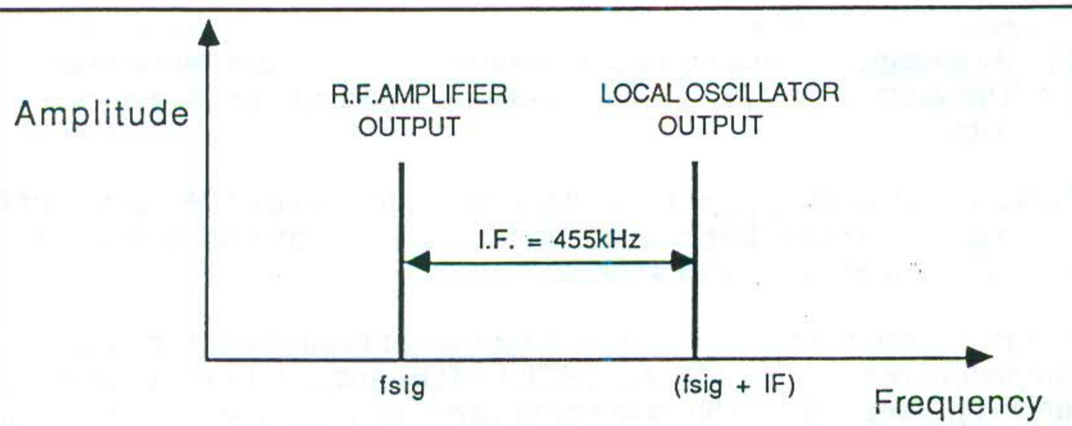


Figure 18

Examine the output of the LOCAL OSCILLATOR block, and check that its frequency varies as the TUNING dial is turned.

Re-tune the Receiver to a radio station before continuing.

8. The operation of the MIXER stage is basically to **shift** the wanted signal down to the I.F. frequency, irrespective of the position of the TUNING dial. This is achieved in two stages:

   (a) by mixing the LOCAL OSCILLATOR's output sinewave 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, fsum = (2f_{sig} + IF)

   The difference between the original two frequencies, f_{diff} = (f_{sig} + IF - f_{sig}) = IF

These three frequency components are shown below:

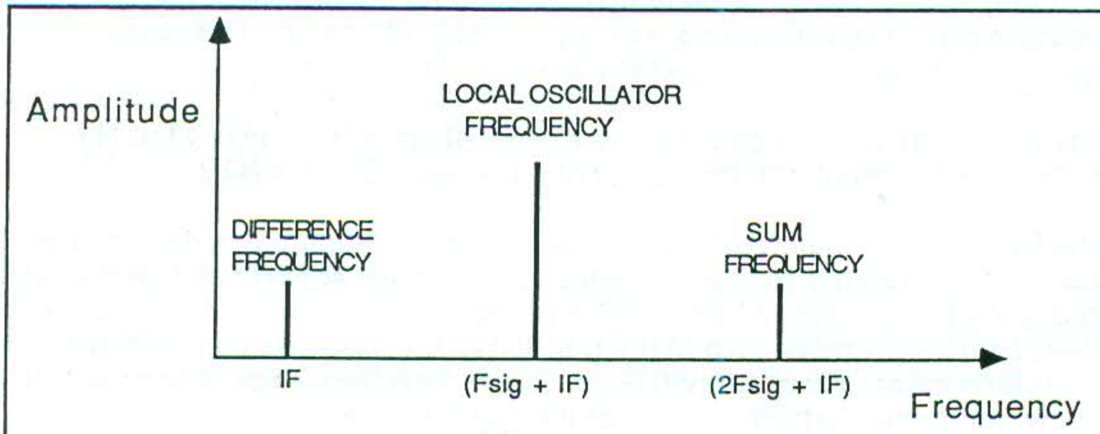


Figure 19

   (b) By strongly attenuating **all** components except the difference frequency, I.F.. This is done by putting a narrow-bandwidth bandpass 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 455kHz (the I.F. frequency), and the sidebands of the AM signal are now either side of 455kHz.

9. Note that, since the mixer's bandpass filter is not highly selective, it will **not completely remove** the LOCAL OSCILLATOR 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 (at t.p.20) 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 bandpass filter.

10. 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 I.F. frequency of 455kHz, is still very small in comparison to the LOCAL OSCILLATOR component.

    What we need to do now is to preferentially amplify frequencies around 455kHz, without amplifying the higher-frequency LOCAL OSCILLATOR and SUM components.

    This selective amplification is achieved by using two I.F. AMPLIFIER stages, I.F. AMPLIFIER 1 and I.F. AMPLIFIER 2, which are designed to amplify strongly a narrow band of frequencies around 455kHz, without amplifying frequencies on either side of this narrow band.

    These I.F. AMPLIFIERS are basically **tuned amplifiers** which have been pre-tuned to the I.F. frequency - they have a bandwidth just wide enough to amplify the 455kHz carrier and the AM sidebands either side of it. Any frequencies outside this narrow frequency band will not be amplified.

    Examine the output of I.F. AMPLIFIER 1 (at t.p.24) with an a.c.- coupled oscilloscope channel, and note that:

    (1) The overall amplitude of the signal is much larger than the signal amplitude at the MIXER's output, indicating that voltage amplification has occurred.

    (2) The dominant component of the signal is now at 455kHz, irrespective of which station you have tuned into. This implies that the wanted signal, at the I.F. frequency, has been amplified to a level where it dominates over the unwanted components.

    (3) The envelope of the signal is modulated in amplitude, according to the sound information being transmitted by the station you have tuned into.

11. Examine the output of I.F. AMPLIFIER 2 (t.p.28) with an a.c.- coupled oscilloscope channel, noting that the amplitude of the signal has been further amplified by this second I.F. amplifier stage.

I.F. AMPLIFIER 2 has once again preferentially amplified signals around the I.F. frequency (455kHz), so that:

(1) The unwanted LOCAL OSCILLATOR and SUM components from the MIXER are now so small in comparison, that they can be ignored totally, and

(2) 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 I.F. AMPLIFIER 2 (t.p.28) is therefore composed almost entirely of a 455kHz carrier, and the A.M. sidebands either side of it, carrying the wanted audio information.

12. The next step is tc extract this audio information from the amplitude variations of the signal at the output of I.F. 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 (t.p.31), together with the output from I.F. AMPLIFIER 2 (at t.p.28). Note that the signal at the DIODE DETECTOR's output:

(1) Follows the amplitude variations of the incoming signal, as required;

(2) Contains some ripple at the I.F. frequency of 455kHz, and

(3) Has a positive d.c. offset, equal to half the average peak-to-peak amplitude of of the incoming signal. We will see how we make use of this offset later on, when we look at Automatic Gain Control (AGC).

13. The final stage of the receiver is the AUDIO AMPLIFIER block. The block contains a simple low-pass filter which passes only audio frequencies, and removes the high-frequency ripple from the DIODE DETECTOR's output signal. This filtered audio signal is applied to the input of an audio power amplifier, which drives the on-board loudspeaker (and the headphones, if these are used). The final result is the sound you are listening to!

The audio signal which drives the loudspeaker can be monitored at t.p.39 (providing that the AUDIO AMPLIFIER block's VOLUME preset is not in its minimum volume position). Compare this signal with that at the DIODE DETECTOR's output (t.p.31), and note how the AUDIO AMPLIFIER block's low-pass filter has 'cleaned up' the audio signal.

You may notice that the output from the AUDIO AMPLIFIER block (t.p.39) is inverted with respect to the signal at the output of the DIODE DETECTOR (t.p.31) - this inversion is performed by the audio power amplifier I.C., and in no way affects the sound produced by the Receiver.

14. Now that we have examined the basic principles of operation of the ANACOM 1/2 receiver for the reception and demodulation of AM broadcast signals, we will try receiving the A.M. signal from the ANACOM 1/1 Transmitter.

    Presently, the gain of ANACOM 1/1's OUTPUT AMPLIFIER block is zero, so that there is no output from theTransmitter. Now turn the GAIN preset in ANACOM 1/1's OUTPUT AMPLIFIER block to its fully clockwise (maximum gain) position, so that the Transmitter generates an AM signal.

    On the ANACOM 1/1 module, examine the Transmitter's output signal (t.p.13), together with the audio modulating signal (t.p.1), triggering the 'scope with the audio signal.

    Since ANACOM 1/1's TX OUTPUT SELECT switch is in the ANT. position, the A.M. signal at t.p.13 is fed to the transmitter's antenna. Prove this by touching ANACOM 1/1's antenna, and noting that the loading caused by your hand reduces the amplitude of the A.M. waveform at t.p.13.

    The antenna will propagate this AM signal over a maximum distance of about 4 feet. We will now attempt to receive the propagated AM waveform with the ANACOM 1/2 board, by using the receiver's on-board antenna.

    Note: If more than one ANACOM 1 Transmitter/Receiver system is in use at one time, it is possible that there may be **interference** between nearby transmitters if antenna propagation is used. To eliminate this problem, use a **screened phono-to-phono link** (supplied) between each Transmitter/Receiver pair, connecting it between ANACOM 1/1's TX. OUTPUT socket and ANACOM 1/2's RX. INPUT socket. If you do this, make sure that the Transmitter's TX. OUTPUT SELECT switch, and the Receiver's RX. INPUT SELECT switch, are **both** in the SKT. position, then follow the steps below as though antenna propagation were being used.

15. On the ANACOM 1/1 module, turn the VOLUME preset (in the AUDIO AMPLIFIER block) **clockwise**, until you can hear the tone of the AUDIO OSCILLATOR's output signal, from the on-board loudspeaker.

    Note: if desired, headphones (supplied with the module) may be used instead of the on-board loudspeaker. To use the headphones, simply plug the headphone jack into the AUDIO AMPLIFIER block's HEADPHONES socket, and put the SPEAKER switch in the OFF position. The volume from the headphones is still controlled by the block's VOLUME preset.

Turn the VOLUME preset to the fully **counter-clockwise** (minimum volume) position before continuing.

16. On the ANACOM 1/2 Receiver, adjust the VOLUME preset so that the receiver's output can be clearly heard. Then adjust the receiver's TUNING dial until the tone generated at the Transmitter is also clearly audible at the Receiver (This should be when the TUNING dial is set to about 55-65), and adjust the Receiver's VOLUME preset until the tone is at a comfortable level.

    Check that you are tuned into the Transmitter's output signal, by varying ANACOM 1/1's FREQUENCY preset (in the AUDIO OSCILLATOR block), and noting that the tone generated by the Receiver changes.

    The ANACOM 1/2 Receiver is now tuned into the AM signal generated by the ANACOM 1/1 Transmitter. Briefly check that the waveforms, at the outputs of the following Receiver blocks, are as expected:

R.F. AMPLIFIER	(t.p.12)
MIXER	(t.p.20)
I.F. AMPLIFIER 1	(t.p.24)
I.F. AMPLIFIER 2	(t.p.28)
DIODE DETECTOR	(t.p.31)
AUDIO AMPLIFIER	(t.p.39)

17. We will now investigate the operation of the Receiver's AGC (**Automatic Gain Control**) CIRCUIT. The AGC CIRCUIT prevents the receiver from overloading when it is tuned into a strong A.M. broadcast signal, by monitoring the d.c. bias voltage at the output of the DIODE DETECTOR.

    You may recall, from our earlier investigation of the DIODE DETECTOR, that this positive bias voltage is proportional to the **average peak-to-peak amplitude** of the signal at the detector's input, which in turn depends on the **carrier strength** of the incoming AM signal at the Receiver's input.

    So, the **stronger** the A.M. signal being received, the **greater** the bias voltage at the DIODE DETECTOR's output. The AGC CIRCUIT then uses this d.c. offset to control the **gain** of the R.F. AMPLIFIER and I.F. AMPLIFIER 1 blocks.

    The AGC CIRCUIT has little effect on circuit operation, provided that the d.c. level at the DIODE DETECTOR's output is below about 0.7 volts.

If the strength of the incoming signal now increases, the **average amplitude** of the signal at the DIODE DETECTOR's input will become larger, and the d.c. level at the detector's output will attempt to rise **above** 0.7 volts. This will cause the AGC CIRCUIT to **reduce** the gain of the R.F. AMPLIFIER and I.F. AMPLIFIER 1 stages. Consequently, the **overall gain** of the receiver **decreases**, reducing the average amplitude of the signal at the DIODE DETECTOR's input, and maintaining the d.c. level of the detector's output at 0.7V.

The overall result is that the **average amplitude** of the AM signal at the input to the DIODE DETECTOR is maintained at a constant level. This has two advantages:

(1) The receiver cannot overload, even when the incoming AM signal is very strong;

(2) Providing the incoming signal strength is sufficient to bring the receiver's AGC CIRCUIT into operation, then the AGC CIRCUIT will compensate for slow fluctuations in signal strength at the receiver's input, by maintaining the audio output from the DIODE DETECTOR at a constant level.

18. The Receiver's AGC CIRCUIT is currently in operation. To examine its behavior, monitor the output of I.F. AMPLIFIER 2 (t.p.28), together with the output of the DIODE DETECTOR (at t.p.31).

Note that the d.c. offset at the DIODE DETECTOR's output is about +0.7 volts, indicating that the incoming signal to the receiver is a strong one. Check that neither of the monitored signals shows any sign of overloading, in spite of the strength of the incoming AM signal from ANACOM 1/1. This is because the AGC CIRCUIT is maintaining the input signal to the DIODE DETECTOR, and the output signal from the DIODE DETECTOR, at a constant level.

Now turn the GAIN preset, in ANACOM 1/1's OUTPUT AMPLIFIER block, slowly **counter-clockwise**, thereby reducing the strength of the transmitted signal. Note that the monitored signals do not decrease in amplitude, until the GAIN preset is almost in its fully **counter-clockwise** (minimum gain) position. This is because the AGC CIRCUIT will compensate for changes in the strength of the incoming signal, so long as the signal strength is large enough to keep the AGC CIRCUIT in operation.

As the OUTPUT AMPLIFIER's GAIN preset approaches the minimum gain position, the output of the Receiver's DIODE DETECTOR block begins to drop below 0.7 volts, and the AGC CIRCUIT no longer has any effect on the Receiver's operation.

Return the OUTPUT AMPLIFIER's GAIN preset slowly to its fully **clockwise** (maximum gain) position, noting the point at which the AGC CIRCUIT comes back into operation.

19. We will now prevent the AGC CIRCUIT from controlling the gain of the Receiver, by disconnecting it from the R.F. AMPLIFIER and I.F. AMPLIFIER 1 blocks. Do this by putting the Receiver's AGC switch in the OUT position, and note the effect on the two monitored waveforms.

    Providing the Receiver has been tuned in correctly, the results of removing AGC should be:

    (1) Overloading of the signal at the DIODE DETECTOR's input, so that all amplitude variations are removed;

    (2) No audio signal at the DIODE DETECTOR's output;

    (3) A d.c. level of greater than 0.7 volts at the DIODE DETECTOR's output.

    Note: If these results are not obtained, it is likely that the Receiver has not been tuned in properly to the transmitted signal. To overcome this, slightly retune the Receiver's TUNING dial, until the above results are obtained.

    Slowly turn the GAIN preset, in the Transmitter's OUTPUT AMPLIFIER block, **counter-clockwise** until the transmitted signal strength is low enough to avoid overloading at the Receiver. Note that the desired waveforms can be re-obtained at the monitored test points of the Receiver.

    Return the OUTPUT AMPLIFIER's GAIN preset (on the ANACOM 1/1 Transmitter) to its fully **clockwise** position, so that overloading occurs once more.

    Finally, check that the desired waveforms can also be obtained by manually reducing the gain of the receiver's R.F. AMPLIFIER. Do this by turning the GAIN preset, in ANACOM 1/2's R.F. AMPLIFIER block, slowly **counter-clockwise**, until the desired signals are once again present at the monitored test points.

    Finally, return the R.F. AMPLIFIER's GAIN preset to its fully clockwise (maximum) position, and return the Receiver's AGC switch to its IN position, so that the AGC CIRCUIT once again takes over. Note that the desired waveforms are once again obtained.

20. By using the optional AUDIO INPUT MODULE (L.J. Order Code CT7), the human voice can be used as the Transmitter's audio modulating signal, instead of using ANACOM 1/1's AUDIO OSCILLATOR block.

    If you have an AUDIO INPUT MODULE, connect the module's output to the EXTERNAL AUDIO INPUT on the ANACOM 1/1 board, and put the AUDIO INPUT SELECT switch in the EXT position.

The input signal to the AUDIO INPUT MODULE may be taken from an external microphone (supplied with the module), or from a cassette recorder, by choosing the appropriate switch setting on the module.

Consult the User Manual for the AUDIO INPUT MODULE, for further details.

RECEPTION OF SINGLE SIDEBAND AM WAVEFORMS

This experiment investigates the reception and demodulation of the single-sideband amplitude-modulated waveforms generated by ANACOM 1/1, using the ANACOM 1/2 Receiver module.

To avoid unnecessary loading of monitored signals, X10 oscilloscope probes should be used throughout this experiment.

EXPERIMENTATION

1. Position the ANACOM 1/1 and 1/2 modules, with the ANACOM 1/1 board on the left, and a gap of about three inches between them. Then connect them to the power supply as shown below:

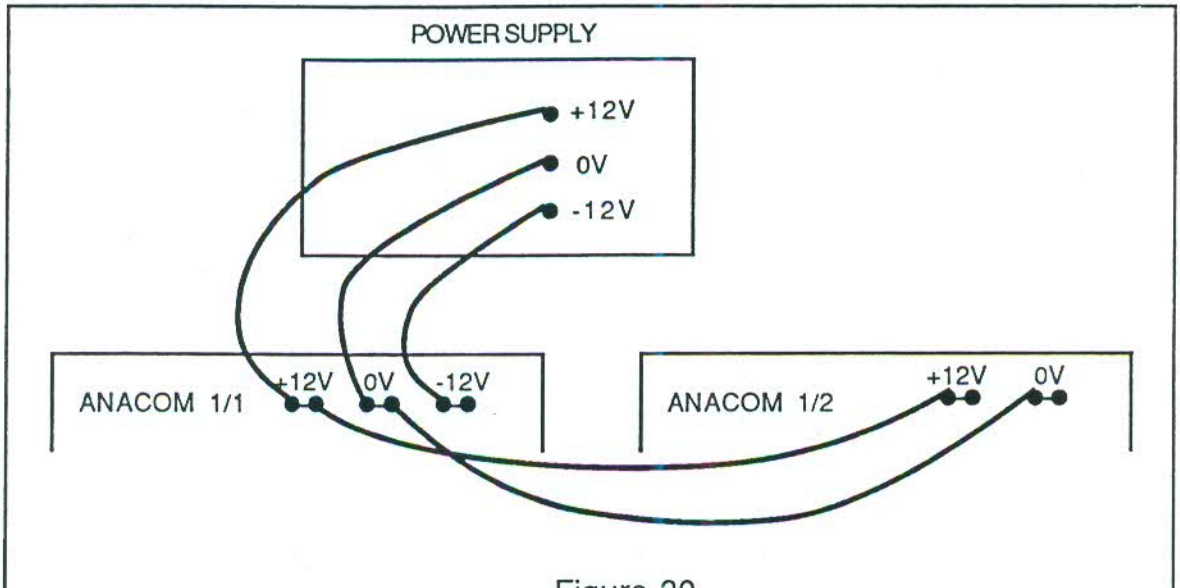


Figure 20

2. Ensure that the following initial conditions exist on the ANACOM 1/1 board:

   (a) AUDIO OSCILLATOR's AMPLITUDE preset in fully **clockwise** position;
   (b) AUDIO INPUT SELECT switch in INT position;
   (c) MODE switch in SSB position;
   (d) OUTPUT AMPLIFIER's GAIN preset in fully **clockwise** position;
   (e) TX OUTPUT SELECT switch in ANT. position;
   (f) AUDIO AMPLIFIER's VOLUME preset in fully **counter-clockwise** position;
   (g) SPEAKER switch in ON position;
   (h) On-board antenna in vertical position, and fully extended.

3. Ensure that the following initial conditions exist on the ANACOM 1/2 board:

   (a) RX INPUT SELECT switch in ANT. position;
   (b) R.F. AMPLIFIER's TUNED CIRCUIT SELECT switch in INT position;
   (c) R.F. AMPLIFIER's GAIN preset in fully **clockwise** position;
   (d) AGC switch in OUT position;
   (e) DETECTOR switch in PRODUCT position;
   (f) AUDIO AMPLIFIER's VOLUME preset in fully **counter-clockwise** position;
   (g) SPEAKER switch in ON position;
   (h) BEAT FREQUENCY OSCILLATOR switch in ON position;
   (i) On-board antenna in vertical position, and fully extended.

4. Turn on power to the modules.

5. On the ANACOM 1/1 module, examine the Transmitter's output signal (t.p.13), and make sure that this is a good SSB waveform, by checking that the signal is a reasonably good sinewave for all positions of the AUDIO OSCILLATOR's FREQUENCY preset.

   Note: The amplitude of the Transmitter's output signal will change as the preset is turned; also, the monitored sinewave may be slightly less pure at low modulating frequencies. These characteristics are due to the fact that the CERAMIC BANDPASS FILTER is not a perfect filter, and they will have negligible effect on the quality of the Receiver's audio output.

   If the monitored waveform is not a good sinewave at **higher** modulating frequencies (i.e. when the FREQUENCY preset is near the MAX position), try adjusting the BALANCE presets in the following two blocks, in order to ensure that the 455kHz and 1MHz carrier components have been completely balanced out:

   (a) BALANCED MODULATOR block, and
   (b) BALANCED MODULATOR & BANDPASS CIRCUIT 2 block.

   If the waveform at t.p.13 is **still** not a good sinewave at higher modulating frequencies, then it is likely that the frequency of ANACOM 1/1's 455kHz OSCILLATOR block needs adjusting. To do this, follow the procedure given in Chapter 8, entitled 'Adjustment of the Transmitter's Tuned Circuits'.

6. Turn ANACOM 1/1's AMPLITUDE preset (in the AUDIO OSCILLATOR block) to its fully **counter-clockwise** (minimum amplitude) position, and note that the amplitude of the monitored output signal from ANACOM 1/1 (at t.p.13) drops to zero. This illustrates that the SSB waveform **contains no carrier** - if the amplitude of the modulating audio signal drops to zero, so does the amplitude of the transmitted SSB signal.

In ANACOM 1/1's AUDIO OSCILLATOR block, return the AMPLITUDE preset to its fully clockwise (MAX) position, and put the FREQUENCY preset in its midway position (arrowhead pointing towards top of PCB), before continuing.

7. We will now transmit the SSB waveform to the ANACOM 1/2 Receiver.

   Since ANACOM 1/1's TX OUTPUT SELECT switch is in the ANT. position, the SSB signal at t.p.13 is fed to the transmitter's antenna. Prove this by touching ANACOM 1/1's antenna, and noting that the loading caused by your hand reduces the amplitude of the SSB waveform at t.p.13.

   The antenna will propagate this SSB waveform over a maximum distance of about 4 feet .

   We will now attempt to receive the propagated SSB waveform with the ANACOM 1/2 board, by using the receiver's on-board antenna.

   Note: If more than one ANACOM 1 Transmitter/Receiver system is in use at one time, it is possible that there may be **interference** between nearby transmitters if antenna propagation is used. To eliminate this problem, use a **screened phono-to-phono link** (supplied) between each Transmitter/Receiver pair, connecting it between ANACOM 1/1's TX. OUTPUT socket and ANACOM 1/2's RX. INPUT socket. If you do this, make sure that the Transmitter's TX. OUTPUT SELECT switch, and the Receiver's RX. INPUT SELECT switch, are **both** in the SKT. position, then follow the steps below as though antenna propagation were being used.

8. On the ANACOM 1/2 module, monitor the output of the I.F. AMPLIFIER 2 block (t.p.28), and turn the TUNING dial until the amplitude of the monitored signal is at its greatest . This should occur at about 85-95 on the TUNING dial.

   Check that you are tuned into the SSB signal, by turning ANACOM 1/1's AMPLITUDE preset (in the AUDIO OSCILLATOR block) to its MIN position, and checking that the monitored signal amplitude drops to zero.

   Return the AMPLITUDE preset to its MAX position before continuing.

9. Since the incoming SSB signal contains **no carrier component**, the Receiver's AGC CIRCUIT cannot make use of incoming carrier amplitude, in order to control the Receiver's gain. This means that the Receiver's AGC CIRCUIT cannot be used for SSB reception, and must be switched out.

Consequently, it is very important to avoid overloading the Receiver by transmitting an SSB signal which is too large for the Receiver to handle. To ensure that overloading does not occur:

(1) Turn the GAIN preset, in ANACOM 1/1's OUTPUT AMPLIFIER block, so that the preset's arrowhead is horizontal, and pointing to the left. This ensures that the amplitude of the transmitted SSB signal is small.

(2) On the ANACOM 1/2 module, fine-tune the TUNING dial until the amplitude of the monitored signal (at t.p. 28) is at its greatest.

(3) Adjust the GAIN preset, in ANACOM 1/2's R.F. AMPLIFIER block, until the amplitude of the monitored signal is about 2 volts pk/pk.

(4) Repeat steps (2) and (3).

There should now be no risk of the ANACOM 1/2 Receiver overloading.

10. For SSB reception, the following blocks of the Receiver operate in the same way as they did for the reception of Double-Sideband AM signals:

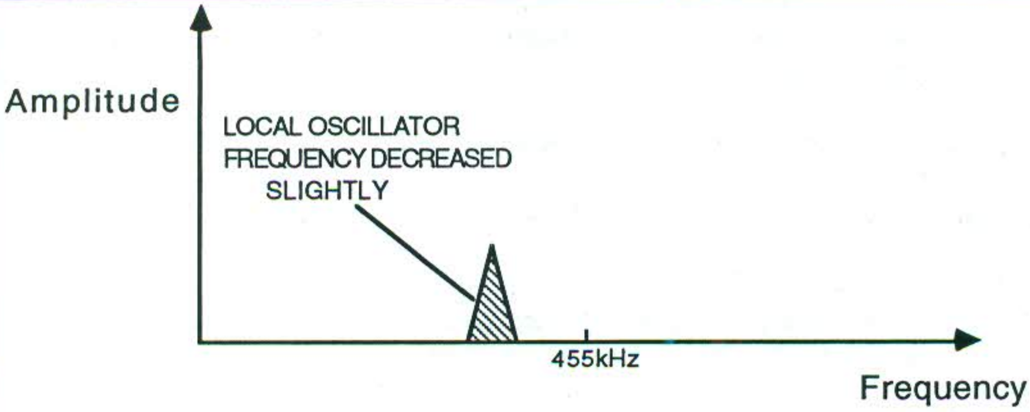
R.F. AMPLIFIER
LOCAL OSCILLATOR
MIXER
I.F. AMPLIFIER 1
I.F. AMPLIFIER 2

Since we have already discussed the operation of these blocks, we will only concern ourselves with how we demodulate the SSB signal at the output of I.F. AMPLIFIER 2.

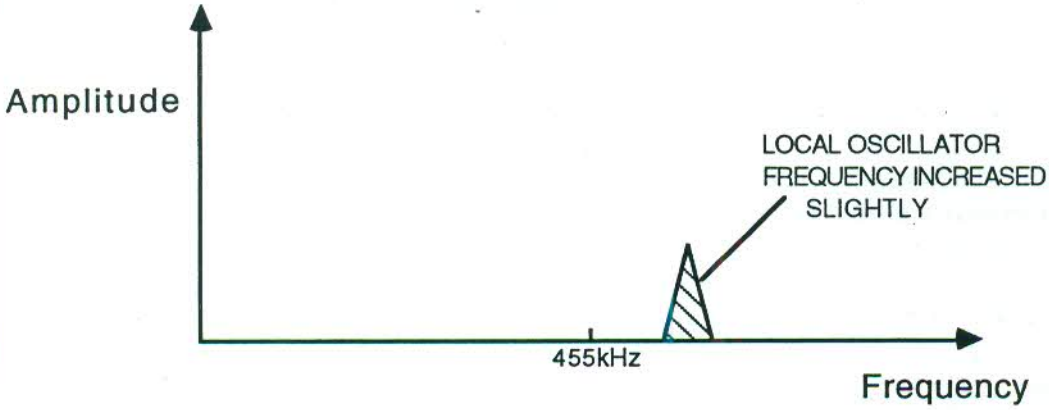
11. The Receiver's BEAT FREQUENCY OSCILLATOR (BFO) produces a sinewave at the I.F. frequency of 455kHz. This 455kHz sinewave is input to the Receiver's PRODUCT DETECTOR block, where it is mixed with the SSB signal from I.F. AMPLIFIER 2.

The actual frequency of the output signal from I.F. AMPLIFIER 2 will lie within a limited range of frequencies, which lie in the region of 455kHz. The output signal can be varied over this limited range of frequencies, by adjusting the frequency of the Transmitter's modulating signal over its full range.

In addition, the position of the limited range of frequencies from I.F. AMPLIFIER 2 will depend on the **exact frequency** of the Receiver's LOCAL OSCILLATOR output. If the oscillator's frequency is varied slightly from its present frequency, this range of frequencies can be moved both above, and below, 455kHz. This is illustrated overleaf:



(a) Frequency range of I.F.AMPLIFIER 2's output with slightly reduced LOCAL OSCILLATOR frequency



(b) Frequency range of I.F.AMPLIFIER 2's output with slightly increased LOCAL OSCILLATOR frequency

Figure 21

The PRODUCT DETECTOR block mixes the output from the BFO with the output from I.F. AMPLIFIER 2. The mixing process results in the the generation of two new frequency components:

(1) A component whose frequency is the **sum** of the two input frequencies;

(2) A component whose frequency is the **difference** between the two input frequencies.

A low-pass filter at the output of the PRODUCT DETECTOR rejects all frequencies except the difference frequency. Consequently, **any slight difference** in frequency between the BFO's output and I.F. AMPLIFIER 2's output will result in an **audio frequency** at the PRODUCT DETECTOR's output. This audio frequency is then converted into sound by the Receiver's AUDIO AMPLIFIER block.

To demodulate our incoming SSB signal, we tune the Receiver's LOCAL OSCILLATOR so that the output frequency range from I.F. AMPLIFIER 2 is slightly **below** the 455kHz BFO frequency (as shown in part (a) of the last diagram), such that the **difference frequency** generated by the PRODUCT DETECTOR is the same as the original Transmitter audio modulating frequency. Then, as the frequency of the Transmitter's modulating signal changes, the output from the PRODUCT DETECTOR should follow it.

12. Monitor the output of ANACOM 1/2's BEAT FREQUENCY OSCILLATOR block (t.p.46), and note that this carries a sinewave of 455kHz.

    On the ANACOM 1/2 Receiver, adjust the VOLUME preset so that the receiver's output is clearly audible.

    Note: if desired, headphones (supplied with the module) may be used instead of the on-board loudspeaker. To use the headphones, simply plug the headphone jack into the AUDIO AMPLIFIER block's HEADPHONES socket, and put the SPEAKER switch in the OFF position. The volume from the headphones is still controlled by the block's VOLUME preset.

    Slowly turn the TUNING dial, and notice that the tone at the Receiver's output changes. This is because the frequency of the output signal from I.F. AMPLIFIER 2 changes as the dial is turned. Since the output frequency from the BFO is fixed, this results in a changing difference frequency at the PRODUCT DETECTOR's output, as the TUNING dial is turned.

    Slowly turn ANACOM 1/2's TUNING dial from about 100 down to 80 on the scale, and note that, as the dial is turned through this region, there is an area where the frequency of the tone generated by ANACOM 1/2 drops to a minimum, and then increases again. The dial position at which the frequency of the tone is a minimum is known as the **minimum frequency postion**.

    By tuning though this region, you have caused the frequency of I.F. AMPLIFIER 2's output to change from being **above** the BFO frequency, to being **below** it. Hence the decreasing, then increasing, tone at the Receiver's output, as the difference frequency decreases, reaches a minimum, and then increases again.

13. On the ANACOM 1/1 module, turn the VOLUME preset (in the AUDIO AMPLIFIER block) **clockwise**, until you can hear the tone of the AUDIO OSCILLATOR's output signal, **in addition to** the tone from the ANACOM 1/2 board.

    With the Receiver's TUNING dial on the **counter-clockwise** side of the minimum frequency position (i.e. using dial positions **lower** than the minimum frequency position), find the position where the two tones are approximately the same.

    Then, turn the FREQUENCY preset in ANACOM 1/1's AUDIO OSCILLATOR block, throughout its range, noting that the frequency of the tone generated by ANACOM 1/2 remains close to that generated by ANACOM 1/1, for **all preset positons.**

    Demodulation of the SSB signal has now been achieved, so the VOLUME preset in the transmitter's AUDIO AMPLIFIER block can now be returned to its fully **counter-clockwise** (minimum) position.

    Note: If the TUNING dial is tuned on the **clockwise** side of the minimum frequency position, rather than the counter-clockwise side, a position **will still be found** where the transmitter and receiver tones are approximately the same. However, if the transmitter's audio frequency is then increased, the receiver's audio frequency will decrease, and vice-versa. The reason for this is that the frequency of I.F. AMPLIFIER 2's output is now **above** the BFO frequency, instead of below it, converting all high-frequency components in the Transmitter's modulating waveform into low-frequency components, and vice-versa.

    Consequently, SSB demodulation is not achieved with the TUNING dial on the clockwise side of the minimum frequency position.

14. On the ANACOM 1/2 module, monitor the output of the PRODUCT DETECTOR block (at t.p.37), together with the output of the AUDIO AMPLIFIER block (t.p.39), triggering the 'scope with the latter signal. **Note**: there will be no signal at t.p.39 if the AUDIO AMPLIFIER's VOLUME preset is in its fully counter-clockwise (minimum) position.

    Vary the frequency of the Transmitter's audio modulating signal by adjusting the AUDIO OSCILLATOR's FREQUENCY preset on the ANACOM 1/1 module. Note how the product detector's output changes as the modulating frequency is changed.

    Also, try briefly reducing the amplitude of the Transmitter's modulating signal to zero (by turning the AUDIO OSCILLATOR's AMPLITUDE preset fully clockwise), and note that the Receiver's output amplitude also drops to zero.

15. With the Receiver's TUNING dial adjusted for correct demodulation of the transmitted SSB signal, you may notice that there is a slight drift in the tone generated by the Receiver. This is due to small frequency drifts in the Transmitter and Receiver oscillator circuits, leading to changes in the difference frequency produced by the PRODUCT DETECTOR.

    Oscillator drift is a serious problem in SSB communication, since it shifts all the frequency components which make up the Receiver's audio output signal, by the same amount. If we try to use our SSB communications system to transmit music, then oscillator drift will cause the harmonic relationship between notes to be lost.

    This makes SSB useless for transmitting music, although it is perfectly adequate for speech.

16. In practice, it would not be possible to align the Receiver to the Transmitter by comparing tones, since the Receiver's operator would not have access to the original audio modulating signal!

    Correct alignment of the Receiver to the Transmitter would actually by achieved by sending speech over the communications system, rather than a single tone. The Receiver operator would then tune the Receiver's TUNING dial to obtain clear, intelligible speech at the Receiver's output.

    By using the optional AUDIO INPUT MODULE (L.J. Order Code CT7), the human voice can be used as the Transmitter's audio modulating signal, instead of using ANACOM 1/1's AUDIO OSCILLATOR block.

    If you have an AUDIO INPUT MODULE, connect the module's output to the EXTERNAL AUDIO INPUT on the ANACOM 1/1 board, and put the AUDIO INPUT SELECT switch in the EXT position.

    The input signal to the AUDIO INPUT MODULE may be taken from an external microphone (supplied with the module), or from a cassette recorder, by choosing the appropriate switch setting on the module.

    Consult the User Manual for the AUDIO INPUT MODULE, for further details.

Investigation of Image Frequencies

This experiment investigates the concept of **image frequencies**, using the ANACOM 1/2 module.In order to explain what these are, here is a short review of how our AM receiver operates:

You will recall that the frequency of the Receiver's LOCAL OSCILLATOR is arranged to be **higher** than the selected signal frequency by a **constant amount**, irrespective of the frequency of the selected station. The amount of that constant difference frequency is chosen to be 455kHz, the Intermediate Frequency.

The Receiver's MIXER block mixes the output from the R.F. AMPLIFIER block with the output from the LOCAL OSCILLATOR, to extract the 455kHz difference frequency between the two signals. This frequency is then passed on to the I.F. AMPLIFIERs, which preferentially amplify signals around 455kHz.

The output from I.F. AMPLIFIER 2 is then passed on to the detector, in order to recover the original audio signal.

Now suppose the R.F. AMPLIFIER contained **no frequency selectivity** at all, so that it amplified **all incoming frequencies** equally. The I.F. AMPLIFIERs will preferentially amplify output signals from the MIXER which are around 455kHz, so they will pass the 455kHz difference frequency which results from mixing the wanted incoming frequency with the output from the LOCAL OSCILLATOR.

Now consider **some other** signal at the R.F. AMPLIFIER's output, whose frequency is 455kHz **above** the frequency of the LOCAL OSCILLATOR. When this frequency is mixed with the output from the local oscillator, a difference frequency of 455kHz will once again be generated. This will be preferentially amplified by the I.F. AMPLIFIER stages, and will appear at the detector's input, in addition to the wanted signal.

The incoming signal whose frequency is exactly 455kHz above the frequency of the LOCAL OSCILLATOR, is known as the **image frequency**.

The frequency spectrum below shows the **wanted** and **image** frequencies at the R.F. AMPLIFIER's output, together with the frequency of the output from the LOCAL OSCILLATOR:

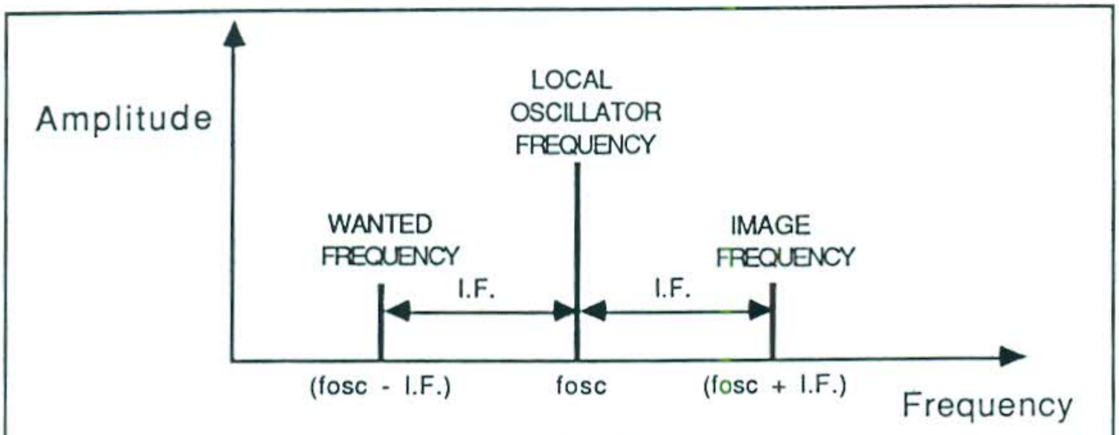


Figure 22

For every wanted frequency there is a corresponding image frequency. If there is a strong station at, or near, the image frequency, it will result in irritating whistling noises at the receiver's output, which will spoil the reception of the wanted station.

Experimentation:

1. Connect supplies to the ANACOM 1/2 module as shown below:

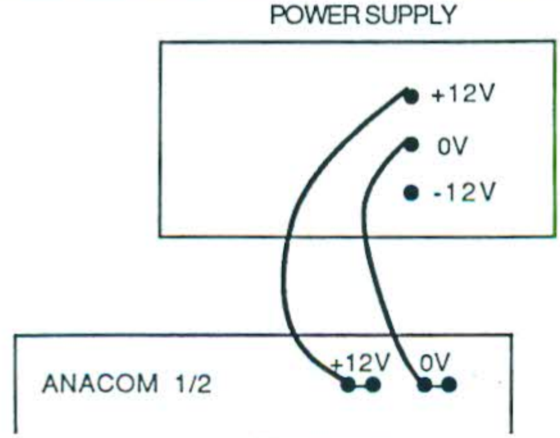


Figure 23

2. Ensure that the following initial conditions exist on the ANACOM 1/2 board:

   (a) R.F. AMPLIFIER's TUNED CIRCUIT SELECT switch in EXT position;
   (b) R.F. AMPLIFIER's GAIN preset in fully **clockwise** position;
   (c) AGC switch in IN position;
   (d) BEAT FREQUENCY OSCILLATOR switch in OFF position.

3. Connect a signal generator to the TUNED CIRCUIT INPUTS on the left-hand side of the ANACOM 1/2 board, as shown below:

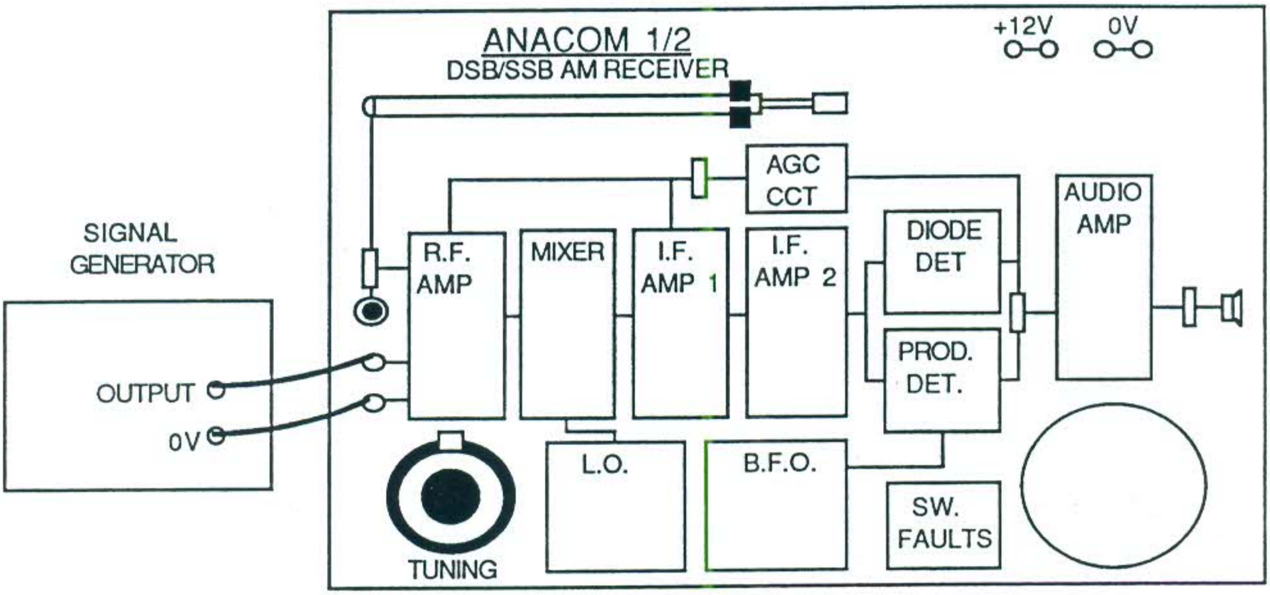


Figure 24

4. Set the signal generator to produce a sinewave output, and adjust the amplitude of the output until the signal at t.p.7 (in the R.F.AMPLIFIER block) is 50mV pk/pk.

5. With the TUNED CIRCUIT SELECT switch in the EXT. position, the R.F. AMPLIFIER block no longer uses its internal tuned circuit - in other words, it has no frequeency selectivity at all. Consequently, any signal fed into the R.F. AMPLIFIER's TUNED CIRCUIT INPUTS will be amplified by the same amount, irrespective of its frequency.

   This should enable us to see image frequencies at the output of I.F. AMPLIFIER 2. Monitor this output (at t.p.28), and adjust the signal generator's output frequency to 1MHz. This is our **wanted** frequency.

6. Turn ANACOM 1/2's TUNING dial so that the monitored signal has maximum amplitude. This should occur at about 55-65 on the dial.

   We are now clearly receiving the wanted signal.

7. Slowly increase the signal generator's output frequency until the monitored signal's amplitude is once again a maximum. This should occur at 910kHz (2 x 455kHz) above the wanted frequency, i.e. at 1.91MHz.

   This incoming signal is at the **image** frequency.

8. We have just shown that the Receiver will allow **both** the wanted frequency, and the image frequency, through to the detector's input, if the R.F. AMPLIFIER is not frequency selective.

   To overcome this, it is important to block the image frequency **before** mixing takes place - this is the reason why we need a frequency-selective R.F. AMPLIFIER block.

   If the R.F. AMPLIFIER's internal tuned circuit is used (as it has been for all previous experiments), both the R.F. AMPLIFIER , and the LOCAL OSCILLATOR, are tuned simultaneously by means of the TUNING control. Tuning the R.F. AMPLIFIER's tuned circuit ensures that the wanted frequency is always passed, but the image frequency (910kHz above the wanted frequency) is greatly attenuated.

   The response of this tuned circuit is shown below:

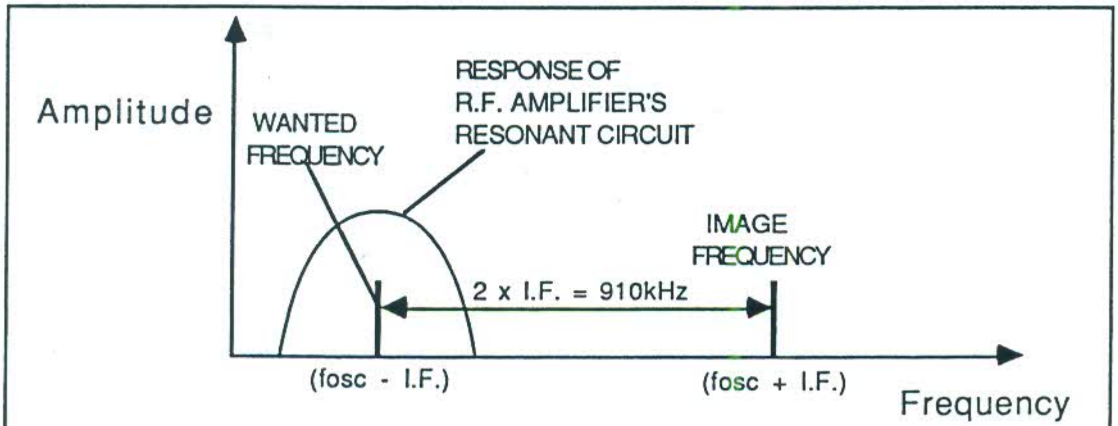


Figure 25

9. To use the R.F. AMPLIFIER's internal tuned circuit:

   (1) Put the TUNED CIRCUIT SELECT switch in the INT position;
   (2) Put the RX. INPUT SELECT switch in the ANT. position;
   (3) Ensure that the on-board antenna is upright and fully extended

10. Return the signal generator's output frequency to 1MHz, and trail the generator's output lead so that it runs close to the antenna without touching it. This 1MHz signal should now be picked up by the antenna, and passed on to the R.F. AMPLIFIER's tuned circuit.

    Re-tune the Receiver's TUNING dial so that the monitored signal (at t.p.28) is a maximum. This should again occur at about 55-65 on the dial.

11. Vary the signal generator's output frequency around 1.91MHz, and check that image frequency has now been removed by the R.F. AMPLIFIERs tuned circuit.

Use of the MEDIUM WAVE TUNED CIRCUIT module with ANACOM 1/2

This experiment describes how the optional MEDIUM WAVE TUNED CIRCUIT module (LJ stock no. DS1A) can be used in place of the tuned circuit in ANACOM 1/2's R.F. AMPLIFIER block.

This allows the performance of ferrite rod antennas to be examined.

Experimentation

1. Connect the MEDIUM WAVE TUNED CIRCUIT module to the ANACOM 1/2 module as shown below:

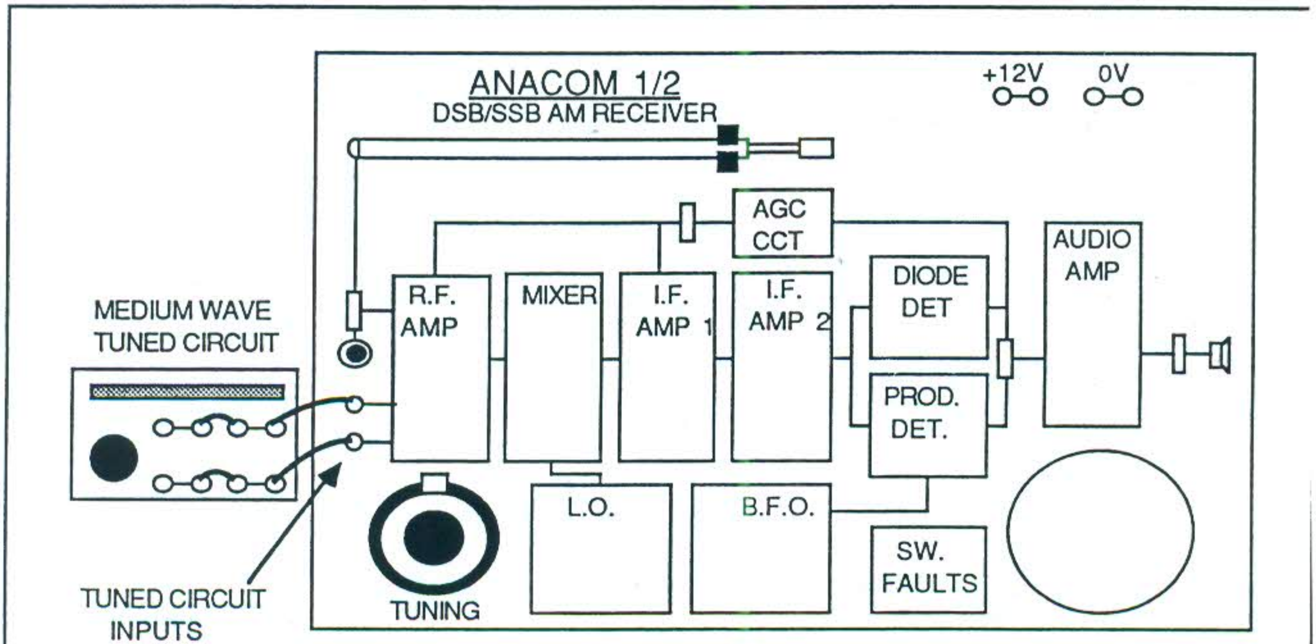


Figure 26

The inductor and variable capacitor on the MEDIUM WAVE TUNED CIRCUIT module are connected in parallel, to form a tuned circuit whose resonant frequency is adjustable. The inductor has a ferrite rod core, which also acts as an antenna.

2. Ensure that the ANACOM 1/2 module's TUNED CIRCUIT SELECT switch is in the EXT. position. The tuned circuit on the MEDIUM WAVE TUNED CIRCUIT module will now be used in place of the R.F. AMPLIFIER's telescopic antenna and internal tuned circuit.

3. Investigate the reception of AM broadcast signals, and DSB/SSB AM signals from ANACOM 1/1, by using the MEDIUM WAVE TUNED CIRCUIT module as the R.F. AMPLIFIER's tuned circuit.

   To tune the Receiver, first tune ANACOM 1/2's TUNING dial into the signal required (this tunes the Receiver's LOCAL OSCILLATOR), then tune the variable capacitor on the MEDIUM WAVE TUNED CIRCUIT module so that maximum receiver output is obtained.

Adjustment of Transmitter Tuned Circuits

This chapter describes how to adjust ANACOM 1/1's tuned circuits for correct operation

Where signals are to be monitored with an oscilloscope, the 'scope's input channels should be a.c.-coupled, unless otherwise indicated. Ensure that X10 oscilloscope probes are used throughout.

A frequency counter should be used for all frequency measurements.

Use the trimming tool, supplied with the ANACOM 1 modules, for trimming inductors. **Never** use a screwdriver, as this may damage the inductor's core. Also, take care not to turn any inductor's core past its end stop, as this may also result in damage.

1MHz CRYSTAL OSCILLATOR Tuned Circuit

Monitor t.p.9 on the ANACOM 1/1 board, while using a trimmer tool to adjust transformer T3 in the 1MHz CRYSTAL OSCILLATOR block.

By carefully tuning T3 throughout its range of adjustment, check that the monitored signal is a d.c. level at both ends of the adjustment range, and a small-amplitude, high-frequency sinewave in the middle of the range.

Tune T3 so that it is in the centre of the 'sinewave' region. Check that the monitored sinewave has an amplitude of approximately 120mV pk/pk and a frequency of 1MHz (±100Hz).

BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 Tuned Circuit

On the ANACOM 1/1 board, put the AUDIO INPUT SELECT switch in the INT position, then turn the AUDIO OSCILLATOR block's AMPLITUDE preset to its fully **clockwise** (MAX) position.

Put the MODE switch in the DSB position, then turn the BALANCE preset in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block to its fully **counter-clockwise** position.

Monitor test points 1 and 3, triggering the 'scope with the t.p.1 signal, and check that the waveforms appear as shown below:

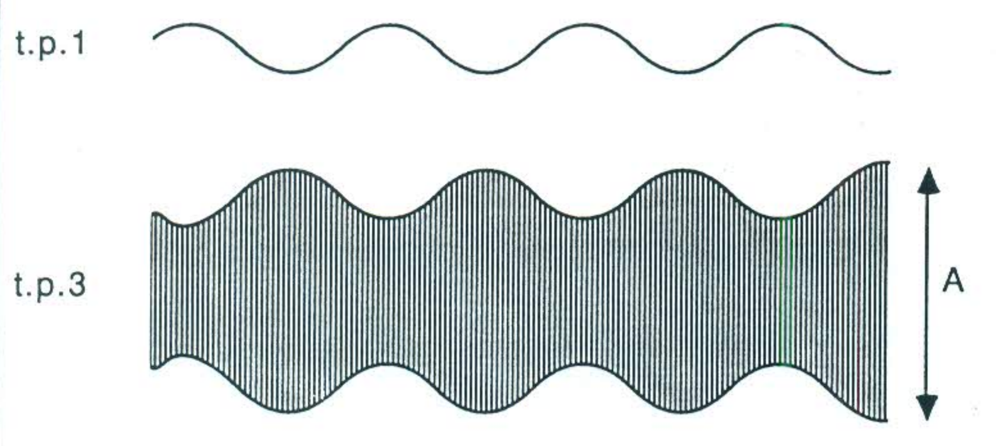


Figure 27

Tune transformer T1 until the amplitude of the waveform on t.p.3 is at its maximum.

While continuing to monitor test point 3, turn the BALANCE PRESET (in the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block) in a clockwise direction, until the monitored waveform is as shown below:

Figure 28

The amplitude of this waveform is a minimum at points A, B, C, etc, as indicated above.

The waveform's amplitude at these points should be as close to zero as possible - to ensure that this is the case, fine-tune transformer T1 until the amplitude of the waveform at these points is as small as possible.

455kHz OSCILLATOR Tuned Circuit

On the ANACOM 1/1 board, monitor t.p.15 and t.p.17 (triggering on t.p.15), and check that the AUDIO INPUT SELECT switch is in the INT position

In the AUDIO OSCILLATOR block, turn the AMPLITUDE preset to its MAX position, and the FREQUENCY preset to its MIN position.

Adjust the BALANCE preset in the BALANCED MODULATOR block until the waveform at t.p.17 is as shown in Figure 28 above, taking care to ensure that adjacent peaks of the waveform's envelope have the same amplitude.

Next, examine test points 15 and 20 on the ANACOM 1/1 board, again triggering the 'scope from t.p.15. Tune transformer T2 (in the 455kHz OSCILLATOR block) until the waveform at t.p.20 is also as shown in Figure 28 above.

Note the overall amplitude of the waveform, then **very slowly** turn T2 **clockwise**, until the overall amplitude is **one fifth (1/5)** of what it was.

The 455kHz OSCILLATOR's tuned circuit should now be correctly adjusted.

<u>BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2 Tuned Circuit</u>

Check that the AUDIO INPUT SELECT switch is in the INT position, and turn the AUDIO OSCILLATOR block's AMPLITUDE and FREQUENCY presets to their MAX positions.

Put the MODE switch in the SSB position, and monitor t.p.22, the output from the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2 block.

Tune transformer T4 (in BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2) until the monitored signal has maximum amplitude.

Adjust the following presets until the monitored signal is a good, clean, sinewave:

(1) BALANCE preset in BALANCED MODULATOR block;

(2) BALANCE preset in BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2 block.

Now fine-tune T4 so that the monitored sinewave's amplitude is once again a maximum.

Adjustment of Receiver Tuned Circuits

This chapter describes how to adjust ANACOM 1/2's tuned circuits for correct operation

Where signals are to be monitored with an oscilloscope, the 'scope's input channels should be a.c.-coupled, unless otherwise indicated. Ensure that X10 oscilloscope probes are used throughout.

A frequency counter should be used for all frequency measurements.

Use the trimming tool, supplied with the ANACOM 1 modules, for trimming inductors. **Never** use a screwdriver, as this may damage the inductor's core. Also, take care not to turn any inductor's core past its end stop, as this may also result in damage.

Adjustment of R.F. AMPLIFIER tuned circuit

Ensure that the following conditions exist on the ANACOM 1/2 module:

(a) BEAT FREQUENCY OSCILLATOR switch in OFF position;
(b) AGC switch in OUT position;
(c) TUNED CIRCUIT SELECT switch (in R.F. AMPLIFIER block) in INT position;
(d) RX. INPUT SELECT switch in ANT. position;
(e) GAIN preset (in the R.F. AMPLIFIER block) in its midway position (arrowhead on preset pointing towards top of board).
(f) On-board antenna upright and extended.

Set up a signal generator so that its output is a sinewave of amplitude 50mV pk/pk and trail the output lead so that it runs close to the antenna (without touching it). Monitor t.p.12 (the output from the R.F. AMPLIFIER block), and follow the steps below:

(a) Turn the vernier TUNING dial to position 25 (i.e. so that the dial's pointer is midway between the '20' and '30' marks on the scale), adjust the signal generator for an output frequency of 615kHz (±1kHz), and tune transformer T1 until the amplitude of the monitored signal is a maximum.

(b) Turn the TUNING dial to position 75, adjust the signal generator for an output frequency of 1220kHz (±1kHz), and tune trimmer capacitor TC1 until the amplitude of the monitored signal is a maximum.

(c) Repeat steps (a) and (b).

Finally, return the R.F. AMPLIFIER's GAIN preset to its fully **clockwise** (MAX) position.

Adjustment of LOCAL OSCILLATOR tuned circuit

Monitor the frequency at t.p.40 (the output of the LOCAL OSCILLATOR block), and follow the steps below:

(a) Turn the TUNING dial to position '0', and tune transformer T5 (in the LOCAL OSCILLATOR block) until the monitored frequency is 980kHz (±1kHz).

(b) Turn the TUNING dial to '100', and tune trimmer capacitor TC2 until the monitored frequency is 2060kHz (±2kHz).

(c) Repeat steps (a) and (b).

Adjustment of MIXER and I.F. AMPLIFIER tuned circuits

Ensure that the BEAT FREQUENCY OSCILLATOR switch is in the OFF position, and the AGC switch is in the OUT position

Set the signal generator up for a sinewave output of amplitude 0.1V pk/pk, and frequency 455kHz (±0.5kHz). Connect the signal generator to t.p. 14 in ANACOM 1/2's MIXER block, and insert Switched Fault 3 (which switches off the LOCAL OSCILLATOR).

Next follow the steps below:

(a) Monitor t.p.28 (output of I.F. AMPLIFIER 2), and tune transformer T2 (in the MIXER block) until the amplitude of the monitored signal is a maximum.

(b) Tune transformer T3 (in the I.F. AMPLIFIER 1 block) until the amplitude of the monitored signal is a maximum.

(c) Tune transformer T4 (in the I.F. AMPLIFIER 2 block) until the amplitude of the monitored signal is once again a maximum.

(d) Repeat steps (a), (b) and (c).

(e) Finally, remove Switched Fault 3.

Adjustment of BEAT FREQUENCY OSCILLATOR tuned circuit

Put the BEAT FREQUENCY OSCILLATOR switch in the ON position, and monitor t.p.46, the output from the BEAT FREQUENCY OSCILLATOR.

Tune transformer T6 until the frequency of the monitored sinewave is 455kHz (±0.5kHz).

Finally, return the BEAT FREQUENCY OSCILLATOR switch to the OFF position.

ANACOM 1 Switched Faults

This section lists the faults on the ANACOM 1/1 and ANACOM 1/2 modules.

There are 8 fault switches on each module, and they are hidden behind locked covers. To remove each cover, use the key provided. Insert it into the socket on the top of the cover, and turn it counter-clockwise.

To replace a fault cover, locate the cover's locking pin in the hole in the support pillar, and turn the key fully clockwise in the cover's socket. To remove the key, turn it counter-clockwise slightly.

ANACOM 1/1 Switched Faults

1. Prevents the 1MHz OSCILLATOR from oscillating, by disconnecting the tuned circuit's primary winding from the +12 volts supply.

2. Disables the output of the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block (at t.p. 3), by disconnecting the tuned circuit's primary winding from the +12 volts supply.

3. Causes the output of the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 1 block (at t.p.3) to become a double-sideband suppressed carrier (DSBSC) signal, irrespective of the position of the block's BALANCE preset. The fault disconnects the BALANCE preset's slider from the -12 volt supply.

4. Causes the output frequency from the AUDIO OSCILLATOR block (at t.p.14) to drop to 150Hz, irrespective of the block's FREQUENCY preset position. The fault disconnects the 56K resistor, in the FREQUENCY preset's divider chain, from 0 volts, so that the FM SWEEP input to the 8038 (pin 8) is pulled up to +12 volts.

5. Stops the 455kHz OSCILLATOR, by shorting out the 18K resistor in the transistor's base bias chain. This causes the bias on the transistor's base (t.p.6) to drop to 0 volts.

6. Prevents the carrier component at the output of the BALANCED MODULATOR block from being 'balanced out' by the block's BALANCE preset, so that a DSBSC waveform cannot be obtained at t.p.17. This is achieved by shorting the 'SIG -' pin of the 1496 (pin 4), to 0 volts.

7. Shorts together the input (t.p.18) and output (t.p.19) of the ceramic filter in the CERAMIC BANDPASS FILTER block, allowing **both** sidebands of the BALANCED MODULATOR block's output signal to reach t.p.20.

8. Disables the output of the BALANCED MODULATOR & BANDPASS FILTER CIRCUIT 2 block (at t.p. 22), by shorting the BIAS input (pin 5) of the 1496 to 0 volts.

ANACOM 1/2 Switched Faults

1. Disables the R.F. AMPLIFIER block, by open-circuiting the transistor's base bias chain. This causes the bias voltage on the transistor's base (at t.p.10) to drop to 0 volts.

2. Disables the output from the MIXER block (t.p.20), by open-circuiting the 1K emitter resistor of the modulating transistor.

3. This open-circuit fault stops the LOCAL OSCILLATOR from working, by removing the bias voltage (at t.p.41) from the transistor's base.

4. This open-circuit fault disables the output from the DIODE DETECTOR block (t.p.31), by removing the d.c. bias (at t.p.30) from the diode's anode.

5. Disables the output from the I.F. AMPLIFIER 1 block (t.p.24), by shorting the transistor's emitter (t.p.23) to the +12 volts supply.

6. This fault disables the PRODUCT DETECTOR block, by shorting the base of the block's output transistor (at t.p.34) to 0 volts.

7. Shorts to 0 volts the AGC control input to the R.F. AMPLIFIER and I.F. AMPLIFIER 1 blocks (at t.p.1 and t.p.2), disabling both blocks.

8. Shorts the inverting input (pin 2) of the AUDIO AMPLIFIER block's LM386 power amplifier I.C. to 0 volts, so that there is no audio output from the block.