SSB DEMODULATION - THE PHASING METHOD

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ACHIEVEMENTS: *modelling of a phasing-type SSB demodulator; examination of the sideband selection capabilities of a true SSB demodulator; synchronous and asynchronous demodulation of SSB; evasion of DSB sideband interference by sideband selection.*

PREREQUISITES: *completion of the experiments entitled* ***Product demodulation - synchronous and asynchronous*** *and* ***SSB generation - the phasing method*** *in this Volume would be an advantage.*

PREPARATION

This experiment is concerned with the demodulation of SSB. Any trigonometrical analyses that you may need to perform should use a single tone as the message, knowing that eventually it will be replaced by bandlimited speech. We will *not* be considering the transmission of data via SSB. As has been done in earlier demodulation experiments, a 'stolen carrier' will be used when synchronous operation is required. It will be shown that, when speech is the message, synchronous demodulation is not strictly necessary; this is fortunate, since carrier acquisition is a problem with SSB.

carrier acquisition from SSB

A pure SSB signal (without any trace of a carrier) contains no explicit information about the frequency of the carrier from which it was generated

But, for speech communications, synchronous operation of the demodulator is not essential; a local carrier within say 10 Hz of the ideal is adequate.

None-the-less, when SSB first came to popularity for mobile voice communications in the 1950s it was difficult (and, therefore, expensive) to maintain a local carrier within 10 Hz (or even 100 Hz, for that matter) of that required. Many techniques were developed for providing a local carrier of the required tolerance, including sending a trace of the carrier - a 'pilot' carrier - to which the receiver was 'locked' to give synchronous operation.

In the interim the tolerance problem was overcome by inevitable technological advances, including the advent of frequency synthesisers, and asynchronous operation became the norm.

In the 1990s the need for synchronous operation has returned, although for a different reason. Now it is desired to send data (or digitized speech) and phase coherence offers some advantages. But methods are still sought to avoid it.

Fortunately, ideal synchronous-type demodulation is not necessary when the message is speech. An error of up to 10 Hz in the local carrier is quite acceptable in most cases (see, for example, Hanson, J.V. and Hall, E.A.; 'Some results concerning the perception of musical distortion in mis-tuned single sideband links', *IEEE Trans. on Comm.*, correspondence pp.299-301, Feb. 1975). For speech communications an error of up to 100 Hz can be tolerated, although the speech may sound unnatural. You can make your own assessment in this experiment.

the synchronous demodulator

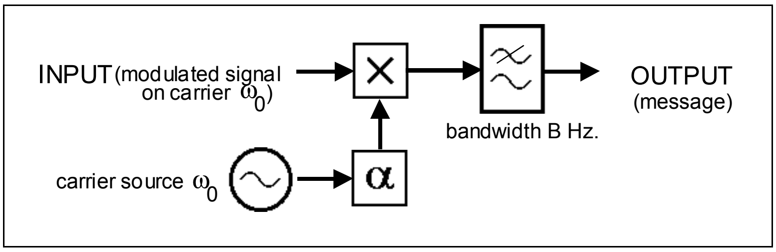

Figure 1: the synchronous demodulator

SSB demodulation can be carried out with a *synchronous demodulator*. You should remember this from the experiment entitled '*Product demodulation - synchronous and asynchronous*'. Figure 1 will remind you of the basic elements. Note that for SSB derived from speech there is no need for the phase shifter [1].

But the arrangement of Figure 1 can not be described as an SSB demodulator, since it is unable to differentiate between the upper and lower sideband of a DSBSC signal. It responds to signals in a window either side of the carrier to which it is tuned, yet the wanted SSB signal will be located on one side of this carrier, not both. The window is *too wide* - as well as responding to the signal in the wanted sideband, it will also respond to any signals in the other sideband. There may be other signals there, and there certainly will be unwanted noise. Thus the output signal-to-noise ratio will be unnecessarily worsened.

a true SSB demodulator

A true SSB demodulator must have the ability to *select* sidebands.

All the methods of SSB *generation* so far discussed have their counterparts as demodulators. In this experiment you will be examining the phasing-type demodulator. A block diagram of such a demodulator is illustrated in Figure 2.

[1] why ?

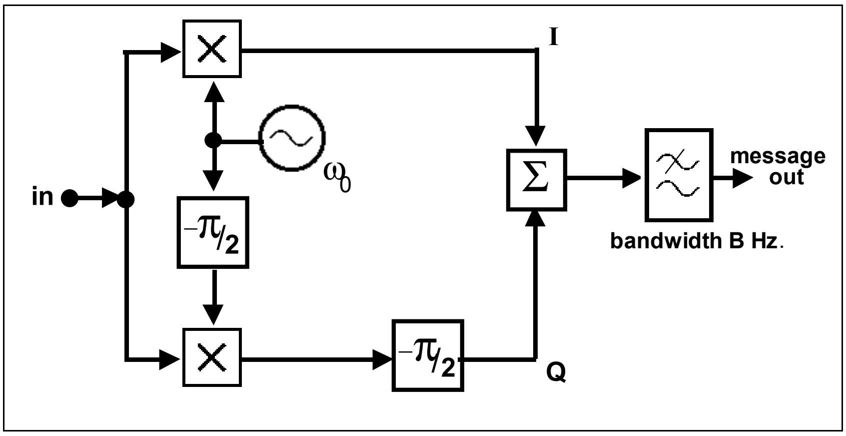

Figure 2: the ideal phasing-type SSB demodulator

principle of operation

It is convenient, for the purpose of investigating the operation of this demodulator, to use for the input signal two components, one ω_H rad/s, above ω_0, and the other at ω_L rad/s, below ω_0. This enables us to follow each sideband through the system and so to appreciate the principle of operation.

The multipliers produce both sum and difference products. The sum frequencies are at or about 2ω rad/s, and the difference (wanted) products near DC. The discussion below is simplified if we assume there are two identical filters, one each in the I (inphase) and Q (quadrature) paths, which remove the sum products.

Consider the upper path I: into the '***I***' input of the summer go *two* contributions; the first is that from the component at ω_H, the second from the component ω_L.

Two more contributions to the summer come from the lower path '***Q***'.

You can show that these four contributions are so phased that those from one side of ω_0 will add, whilst those from the other side will cancel. Thus the demodulator appears to look at only one side of the carrier.

The purpose of the adjustable phase α is to vary the phase of the local carrier source ω_0 with respect to the incoming signal, also on ω_0.

practical realization

As was discussed in the experiment entitled '***SSB** generation - the phasing method*', the physical realization of a two-terminal wide-band 90° phase shifter network (in the Q arm) presents great difficulties. So the four-terminal quadrature phase *splitter* - the QPS - is used instead. This necessitates a slight rearrangement of the scheme of Figure 2 to that illustrated in Figure 3.

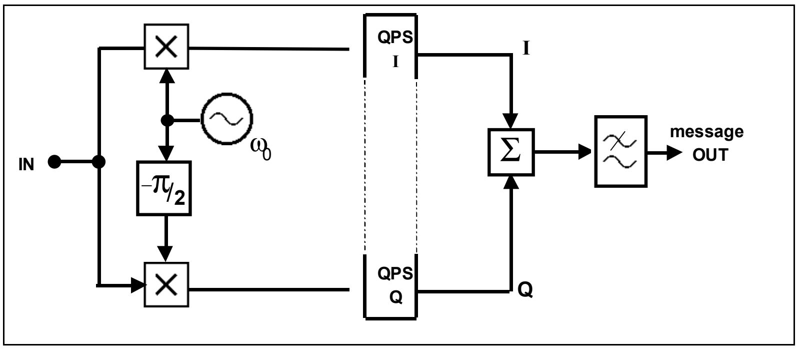

Figure 3: the practical phasing-type SSB demodulator

practical considerations

Figure 3 is a practical arrangement of a phasing-type SSB demodulator.

The $\pi/2$ phase shifter needs to introduce a 90^0 phase shift at a single frequency, so is a narrowband device, and presents no realization problems.

The QPS, on the other hand, needs to perform over the full message bandwidth, so is a wideband device.

Remember that the outputs from the multipliers contain the sum and difference frequencies of the product; the difference frequencies are those of interest, being in the message frequency band.

The sum frequencies are at twice the carrier frequency, and are of no interest. It is tempting to remove them with two filters, one at the output of each multiplier, because their presence will increase the chances of overload of the QPS. But the transfer functions of these filters would need to be *identical* across the message bandwidth, so as not to upset the *balance* of the system, and this would be a difficult practical requirement.

Being a linear system in the region of the QPS and the summing block, *two* filters in the I and Q arms (the *inputs* to the summing block) can be replaced by a *single* filter in *output* of the summing block.

The lowpass filter in the summing block output determines the bandwidth of the demodulator in the 100 kHz part of the spectrum; that is, the width of the window located *either* above *or* below the frequency ω_0. Its bandwidth must be equal to or less than the frequency range over which the QPS is designed to operate, since, outside that range, cancellation of the unwanted sideband will deteriorate.

EXPERIMENT

outline

For this experiment you will be sent three signals via the trunks; an SSB, an ISB, and a DSBSC (with superimposed interference on one sideband).

Generally speaking, if the messages are speech, or of unknown waveform, it would be very difficult (impossible ?) to differentiate between these three by viewing with an oscilloscope. For single tone messages it would easier - consider this !

You may be advised of the nature of the messages, but not at which TRUNKS outlet each signal will appear.

The aim of the experiment will be to identify each signal by using an SSB demodulator.

The unknown signals will be in the vicinity of 100 kHz, as arranged by your Laboratory Manager. They may or may not be based on a 100 kHz carrier locked to yours.

You should start the experiment using the 100 kHz sinewave from the MASTER SIGNALS module for the local carrier; but any stable carrier *near* 100 kHz would suffice. This will need to be split into two paths in quadrature. If you use the 100 kHz carriers from the MASTER SIGNALS module you might feel tempted to use the sine and cosine outputs. But fine trimming will be needed for precise balance of the demodulator, so a PHASE SHIFTER will be used instead. This has been included in the patching diagram of Figure 4.

patching the model

T1 *patch up a model to realize the arrangement of Figure 3. A possible method is shown in Figure 4. The VCO serves as the test input signal.*

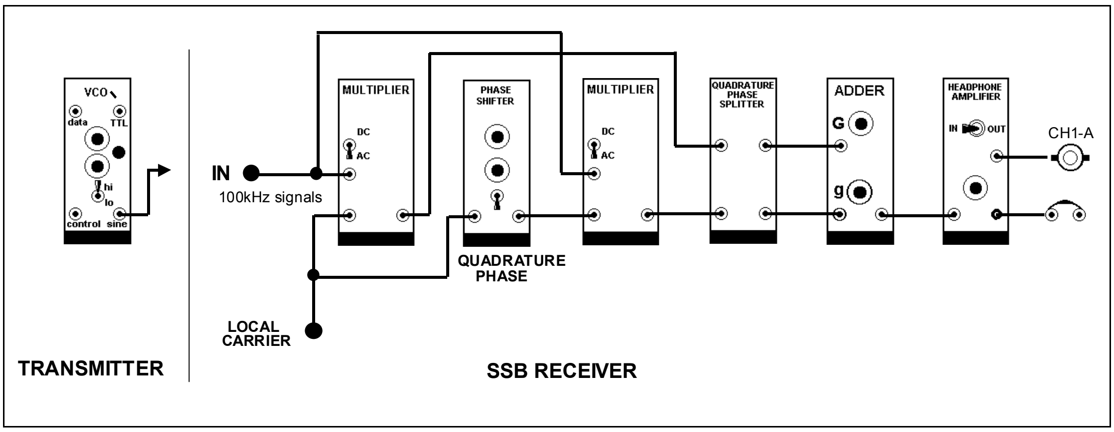

Figure 4: model of an SSB demodulator

Before the demodulator can be used it must be aligned. A suitable test input signal is required. A single component near 100 kHz is suitable; this can come from a VCO, set to one or two kilohertz above or below 100 kHz, where the unknown signals will be located, and so where your demodulator will be operating. Make sure, after demodulation, it will be able to pass through the 3 kHz LPF of the HEADPHONE AMPLIFIER module.

For example, a 98 kHz single frequency component is simulating an SSB signal, derived from a 2 kHz message, and based on a 100 kHz (suppressed) carrier.

trimming

After patching up the model the balancing procedure can commence.

T2 *set the VCO to, say, the upper sideband of 100 kHz, at 102 kHz or thereabouts.*

T3 *check that there is a signal of much the same shape and amplitude from each MULTIPLIER. These signals should be about 4 volts peak-to-peak. Their appearance will be dependent upon the oscilloscope sweep speed, and method of synchronization. They will probably appear unfamiliar to you, and unlike text book pictures of modulated signals. Do you understand why ?*

You will now examine the performance of the upper, '***P***', branch and the lower, '***Q***', branch, independently.

Remember that each branch is like a normal (asynchronous) SSB demodulator. Phasing has no influence on the output amplitude. It is only when the outputs from the two branches are *combined* that something special happens.

check the I branch

T4 *remove input **Q** from the ADDER. Adjust the output of the filter, due to **I**, to about 2 volts peak-to-peak with the appropriate ADDER gain control. It will be a sine wave. Confirm it is of the correct frequency. Confirm that adjustment of the PHASE SHIFTER has no significant effect upon its amplitude.*

check the Q branch

T5 *remove input **I** from the ADDER, and replace input **Q**. Adjust the output of the filter, due to **I**, to about 2 volts peak-to-peak with the appropriate ADDER gain control. It will be a sine wave. Confirm it is of the correct frequency. Confirm that adjustment of the PHASE SHIFTER has no significant effect upon the amplitude.*

combine branches

T6 *replace input **Q** to the ADDER. What would you expect to see ? Merely the addition of two sinewaves, of the same frequency, similar amplitude, and unknown relative phase. The resultant is also a sine wave, of same frequency, and amplitude anywhere between about zero volt, and 4 volt peak-to-peak. What would we like it to be ?*

T7 *rotate the PHASE SHIFTER front panel control. Depending upon the state of the 180^0 toggle switch you may achieve either a maximum or a minimum amplitude output from the filter. Choose the minimum.*

T8 *adjust one or other (not both) of the ADDER gain controls until there is a better minimum.*

T9 *alternate between adjustments of the PHASE control and the ADDER gain control, for the best obtainable minimum. These adjustments will not be interactive, so the procedure should converge fast.*

When the above adjustments are completed to your satisfaction you have a true SSB receiver. It has been adjusted to ignore any input on the sideband in which your test signal was located. If this was the lower sideband, then you have an upper sideband receiver. If it had been in the upper sideband, then you have a lower sideband receiver.

Note that you were advised to *null* the unwanted sideband, rather than *maximise* the wanted.

But you could have, in principle, chosen to adjust for a *maximum*. In that case, if the test signal had been in the lower sideband, then you have a lower sideband receiver. Had it been in the upper sideband, then you have an upper sideband receiver.

In practice it is customary to choose the nulling method. Think about it !

To convince yourself that what was stated above about which sideband will be selected, you should sweep the VCO from say 90 kHz to 110 kHz, while watching the output from the receiver - that is, from the 3 kHz LPF output. You will be looking for the extent of the 'window' through which the receiver looks at the RF spectrum.

T10 *do a quick sweep of the VCO over its full frequency range (or say 90 to 110 kHz). Notice that there is a 'window' about 3 kHz wide on **one side only** of 100 kHz from which there is an output from the receiver. Elsewhere there is very little.*

T11 *repeat the previous Task, this time more carefully, noting precisely the VCO and audio output frequencies involved, their relationship to each other, and to the 3 kHz LPF response. Sketch the approximate response of the SSB receiver.*

swapping sidebands

It is a simple matter to change the sideband to which the demodulator responds by flipping the $\pm 180^0$ toggle switch of the PHASE SHIFTER.

T12 *flip the $\pm 180^0$ toggle switch of the PHASE SHIFTER. Did this reverse the sideband to which the demodulator responds ? How did you prove this ? Was (slight) realignment necessary ?*

There are other methods which are often suggested for changing from one sideband to the other with the arrangement of Figure 3. Which of the following would be successful ?

1. swap *inputs* to the QPS.
2. swap *outputs* from the QPS.
3. interchange the I and Q paths of the QPS (ie, inputs *and* outputs).
4. swap signal inputs to the two MULTIPLIERS.
5. swap carrier inputs to the two MULTIPLIERS.
6. any more suggestions ?

identification of signals at TRUNKS

There are three signals at TRUNKS, all based on a 100 kHz carrier. They are:

- an SSB derived from speech.
- an ISB, at least one channel being derived from speech
- a DSBSC, derived from speech, but with added interference.

T13 *use your SSB demodulator to identify and discover as much about the signals at TRUNKS as you can.*

You should have been able to:

- verify that either sideband may be selected from the ISB
- show that the interference is on one sideband of the DSBSC, and that the other sideband may be demodulated interference-free
- identify which sideband of the DSBSC contained the interference.

asynchronous demodulation of SSB

So far you have been demodulating SSB and other signals with a *stolen* (and therefore synchronous) carrier.

There was no provision for varying the phase of the stolen carrier *before* it was split into an *inphase* and *quadrature* pair. This would have required another PHASE SHIFTER module in the arrangement of Figure 3. However, it was observed in an earlier experiment (and may be confirmed analytically) that this would change the phase of the received message, but not its amplitude, and so would go unnoticed with speech as the message.

But what if the local carrier is not synchronous - that is, if there is a small frequency error between the SSB carrier (suppressed at the transmitter), and the local carrier (supplied at the receiver) ? You can check the effect by using the analog output from a VCO in place of the 100 kHz carrier from the MASTER SIGNALS module.

T14 *replace the 100 kHz carrier from the MASTER SIGNALS module with the analog output from a VCO. Set the VCO frequency close to 100 kHz, and monitor it with the FREQUENCY COUNTER. Remember the preferred method of fine tuning the VCO is to use a small, negative DC voltage in the CONTROL VOLTAGE socket, and fine tune with the GAIN control. (refer to the* ***TIMS User Manual****).*

T15 *connect the SSB at TRUNKS to the input of the demodulator, and listen to the speech as the VCO is tuned slowly through 100 kHz. Report your findings. In particular, comment on the intelligibility and recognisability of the speech message when the frequency error δf is about 0.1 Hz, 10 Hz, and say 100 Hz.*

TUTORIAL QUESTIONS

Q1 *confirm analytically that the RF window width of the arrangement of Figure 1 is* **twice** *the bandwidth of the LPF.*

Q2 *confirm analytically that the RF window width of the arrangement of Figure 2 is* ***equal to*** *the bandwidth of the LPF.*

Q3 *the trimming procedure of the phasing-type demodulator could have chosen to* ***maximize*** *or* ***minimize*** *the filter output. Explain the difference between these two possible methods. Which would you recommend, and why ?*

Q4 *when would a true SSB demodulator (Figure 2) give superior performance to a 'normal' product (synchronous) demodulator (Figure 1), when demodulating a DSBSC. How superior ? Explain.*

Q5 *you have met all the elements of the SSB demodulator of Figure 3 in earlier experiments, so should know their characteristics. If not, measure those you require, and predict, analytically, which sideband it is 'looking at'. Check that this agrees with experiment.*

Q6 *why use a PHASE SHIFTER module for the quadrature carrier, instead of using the* inphase *and* quadrature *outputs already available from the MASTER SIGNALS module ?*

Q7 *do you think it is essential for an SSB demodulator to be synchronous when the message is speech ? What sort of frequency error do you think is acceptable ? What would be the tolerance requirements of the receiver carrier source (assuming no fine tuning control) if the SSB was radiated at 20 MHz ? Answer this questions from your own observations. See what your text book says.*