

# Modulation Methods SSB and DSB

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SSB or Single Sideband, is a type of AM without the carrier and one sideband. DSB or double sideband is AM with the carrier suppressed, but both upper and lower sidebands are used. DSB is compatible with SSB receivers, the receiver merely rejecting the unwanted or redundant sideband. The use of both sidebands to carry two separate channels of information is called ISB, or independent sideband. It was somewhat popular with hams in the early 1960s as AM was gradually yielding to SSB, as a DSB transmitter is relatively simple to build. DSB is seldom used today, but it was a cheap way back then to gradually phase over to SSB, as SSB receivers could handle it and the unwelcome carrier signal was absent. We will not discuss DSB any further as it is considered obsolete as voice transmission method in HF communications work. It is still used in FM stereo transmission for the 38 kHz audio channel difference (L-R) subcarrier. This will be covered in a later column.

SSB or single sideband, was known in the early days of radio, but circuit techniques and hardware to generate it did not become readily available until after WW2. There was a transatlantic telephone circuit operating on about 55 kHz in the long wave band during the 1920s, which used SSB transmission. Experimentally minded amateur radio operators (hams) experimented with SSB after WW2, while AM was still “king”. However, the gradual shift to SSB started during the late 1950s. During the early 1960s, reasonably priced manufactured SSB equipment became available to amateurs, and a gradual changeover to SSB took place. By 1970 AM was mainly used on the 28 Mhz and VHF amateur bands, and it was called “Ancient Modulation”. Even on the VHF bands, FM (Frequency Modulation) took over during the 1970s, and by 1980 AM was pretty scarce. It has made a small comeback since the late 1980s, since SSB equipment using LSI chips and microprocessors has become smaller, sophisticated, and too complex and forbidding for home experimentation. The old vacuum tube AM equipment has enjoyed somewhat of a revival, as it lends itself to experimentation by amateurs and those interested in restoring and operating old time vintage equipment. AM activity can still be found near the 3.9 and 29 Mhz frequencies in the 75 meter and 10 meter ham bands, with some local AM work also at 50.4 MHz in the 6 meter band. The military has also long since converted to SSB for its HF communications work. With the exception of international broadcasting, HF voice communications is practically all SSB. And international short wave broadcasting is going this route and also toward digital radio. But AM is still the simplest and cheapest from a reception standpoint, and is almost universally used worldwide for broadcasting and air to

ground VHF-UHF voice communications. However, the use of SSB allows superior weak signal reception and less transmitter power for the same results.

Most SSB exciters first generate a DSB signal which is then processed into SSB. An AM signal with one sideband partially suppressed is called VSB or vestigial sideband. This is widely used in television transmission to reduce bandwidth while still allowing AM detection schemes to be used. A SSB signal can be transmitted with a carrier to reduce occupied bandwidth, and this is called CSSB or compatible SSB. It has little advantage over AM other than the reduction in bandwidth and selective fading effects. Selective fading is a phenomenon in radio transmission where the fading of a signal at the receiver is very frequency selective, usually due to radio wave cancellation effects caused by phase differences from multipath transmission and ionospheric effects. It acts as a sharp notch filter which continuously and randomly varies in center frequency, and randomly nulls out one sideband, then the carrier, then the other sideband, and then might reverse direction. This randomly moving “notch filter” causes the fading and intermittent audio distortion heard on received AM signals. This can be easily heard on distant AM broadcast stations during the nighttime hours, when multipath effects from skywave and groundwave signals cause this phenomenon. SSB is less susceptible to this as there is no carrier and other sideband to deal with. Therefore SSB transmission usually only exhibits rising and falling signal levels, with little extra distortion as compared to AM.

Referring to Fig 1, SSB is generated as follows: Audio information at the transmitter input is first fed into an amplifier and possibly a speech compressor or clipper. This serves to increase average modulation level. A word of caution here. Unlike an AM signal, in which the envelope has the same waveform as the modulating waveform, the envelope waveform of a SSB signal has no direct simple relationship to the modulating signal (see Fig 2), and using clipping of peaks may and can introduce undesirable effects and actually degrade the signal. Compression, on the other hand, largely preserves the waveshape of the modulating signal, mainly affecting its amplitude, and can be effective in boosting the average modulation level. The lesson here is to avoid the all too commonly heard overclipped and overcompressed signals that are strong but nearly unreadable. The idea that “if enough is enough, then more is better, and too much is just right” does not apply in this case. Next the audio should be bandlimited to eliminate products outside the intended bandwidth. Typically this will be 200 to 3500 Hz for speech, although 2500 Hz is sometimes used as an upper limit. Next, the audio is fed to a balanced modulator that is also driven with an RF carrier at the SSB generation frequency, or sometimes called the transmitting IF frequency. In many instances this is the same the receiver IF frequency, as often done in transceiver systems, where the same circuitry is used for modulation and demodulation. The output of the balanced modulator (actually a mixer) is a double sideband suppressed carrier signal, since the carrier is cancelled out. In the absence of a modulating signal, the output is ideally zero. In practical balanced modulators, about 30 to 40 dB suppression of the carrier is obtained. There is usually some provision provided for optimizing carrier

suppression in most circuits, although with modern solid state diode doubly balanced mixer assemblies inherent suppression is good enough and no adjustment is necessary.

Next, the output of the mixer or modulator is fed to a sharp cutoff filter. This filter may be made up of L-C elements (in the 10-50 KHz range) or mechanical resonators (455 or 500 KHz), or most often made from quartz crystals. Crystal filters are available at many popular frequencies as off the shelf assemblies. 1.65, 3.0, 5, 9, 10.7, and 21.4 Mhz are common SSB IF frequencies that are stock crystal filters, and many other frequencies are also used. The 5 to 9 Mhz range seems most popular, as crystal filters for this range are readily made. The filter should have a bandwidth (for speech) of about 2.1 to 3 kHz, and should have a center frequency about 1.5 KHz above or below the carrier frequency, and should have 20 to 30 dB rejection at the carrier frequency. The filter should cut off sharply on the carrier side and should have 40 dB or better rejection of the unwanted sideband. This is why crystals are used in these filters as we need the very high Q values to physically realize this kind of rejection and bandwidth. The filter is generally one of the more expensive components in a SSB system.

A SSB generator of this type can generate a SSB signal of either lower or upper sideband. This is a function of the filter response characteristics. If capability to generate a signal of either sideband is needed, there are several approaches. First, two separate filters can be used with a switching arrangement to select the desired sideband. Alternately, a filter with a symmetrical response curve that has a very sharp cutoff on each side can be used, and the carrier oscillator can be shifted to either side of the filter. A scheme that was popular some years ago used a filter at 9.000 MHz that had a symmetrical response plus and minus 1.2 kHz from each side of center at 9.000 Mhz. This gave a bandwidth of 2.4 KHz for the signal, and two separate crystals were provided in the carrier oscillator, one at 8998.5 for USB generation, and another at 9001.5 for lower sideband generation. This had the disadvantage of having a 1.5 KHz nominal error in the 9.000 Mhz nominal frequency, but this was corrected by shifting the LO signal plus or minus 1.5 kHz to compensate, so the final output frequency would be correct. This is most easily done in today's transmitters via software in the microcontroller programming used for controlling the frequency synthesizer, so the entire operation is transparent and automatic. Another method using only one filter involves using a mixer. One scheme used in the past was to generate the SSB signal at 455 KHz using a mechanical filter as the SSB filter, as follows: The output from the SSB generator is 455 kHz USB. Next, the 455 KHz signal is mixed with the fourth harmonic of the carrier,  $4 \times 455$  KHz or 1820 KHz, giving an USB signal of 2275 KHz, which is the IF frequency used in this system. If a LSB signal is desired, the sixth harmonic of 455 KHz at 2730 KHz is used, and the 455 KHz USB signal when mixed with 2730 KHz results in a 2275 KHz SSB signal as before, but now this is a LSB signal. This is because we are taking the difference rather than the sum. In sum mixing, the output is the sum of the IF signal and the LO signal. If the IF increases in frequency so does the sum of the two signals. In difference mixing when the IF signal increases in frequency the resulting

sum of the IF and LO will decrease in frequency. This results in inversion of the SSB signal about the carrier frequency (In this case 2275 Khz). This system has the disadvantage of needing a mixer and extra stages to generate the X4 and X6 signals needed to mix with the generated SSB signal, and corresponding switching arrangements, and the extra cost and complexity must be weighed against the cost of an extra filter. In amateur radio HF transceivers, a commonly used technique is the use of one symmetrical filter with corresponding offsetting the LO, as mentioned before, since the software needed in the synthesizer costs nothing once written and debugged, and takes no physical room. The sharp symmetrical filter is cheaper than two separate filters as well, and transceiver design is simplified, as the same conditions apply to both receive and transmit.

The output of the filter is a SSB signal at the IF frequency. This signal is then mixed with a very stable and pure local oscillator signal from a very stable VFO or frequency synthesizer. This is done in a high level very linear mixer to produce the desired SSB output frequency. A filter system then removes unwanted mixer products and the resulting SSB signal is then amplified to the final transmitter power output level, which may be a few watts to many kilowatts. A very linear amplifier must be used to prevent the generation of intermodulation distortion products that will appear as unwanted components and interference on the transmitted signal. Linear amplifiers may be vacuum tube or solid state. For very high power levels, (about 500 watts or more) vacuum tube technology is still the technology of choice. Most transmitters and transceivers in the 100 watt class use solid state bipolar or power FET devices. Higher power solid state amplifiers above a few hundred watts generally need large and heavy heatsinks, RF power combiners, and several large expensive transistors, together with a high current, low voltage supply. It is difficult to get these large quantities of heat out of relatively small chip areas while keeping chip temperatures reasonable. Often sophisticated protection circuitry is needed to protect the transistors against load faults and power spikes. Vacuum tubes do not have this problem, and only a simple cooling fan is required in most cases. There are a few solid state 500-1000 watt amplifiers sold by SSB radio manufacturers, but vacuum tube amplifiers are usually smaller, can be just as, or more efficient than solid state, and more reliable, with much better immunity to load faults, such as high VSWR due to broken, mismatched or shorted antennas. A vacuum tube can usually stand a severe fault for a few seconds, while transistors can fail in microseconds. This is why high power applications are often better implemented with vacuum tubes. The large, expensive, and heavy 60 Hz transformer type high voltage supplies from the old days can now be replaced with much smaller and lighter highly efficient switching type solid state supplies, but tubes still are better suited for the RF circuitry. The vacuum tube still is king here, and may always be, for high power levels. However, for low power (200 watts or less) and portable transmitter use, solid state is undeniably the best approach.

Another approach to SSB generation is called the phasing method. In this approach a clever phase cancellation technique is used. This method eliminates the need for a sharp SSB filter and is potentially lower in cost. (See Fig 3) First, the

audio signal after processing and bandlimiting (very important in this approach) is split into two components, equal in amplitude but exactly 90 degrees apart in phase. This is the difficult part, as a network is needed that provides a 90 degree phaseshift within plus or minus 1 or 1.5 degrees over the entire audio range of 300 to 3000 Hertz. There are classes of R-C networks that have this property, generally involving precision components. In practice, each audio component is fed to a separate network and while the individual network phaseshifts vary over the audio frequency range, the difference between their outputs stays within a degree of 90 degrees, with constant amplitude. The synthesis of these networks is beyond the scope of this article. A schematic of a typical network is shown. It is rather simple, but requires precision components. The degree of unwanted sideband suppression depends on it. Next, the two audio channels 90 degrees apart are fed to identical balanced modulators or doubly balanced mixers that are driven with two carriers identical in frequency but also exactly 90 degrees apart in phase. This is easily done since the carrier frequency is generally fixed. A network consisting of R-C or L-C circuits can provide this 90 degree phaseshift, or a divide by two frequency divider can be used. Two JK flip flops driven by two identical clock signal square waves 180 degrees apart will produce two outputs at half the input frequency and 90 degrees apart in phase. Next, the outputs of the two mixers are combined. It can be shown that the output will consist of only one sideband, (See Fig 3) since the double sideband signals from each mixer will have phase relationships such that one of the sideband components will have opposite phase with respect to the other, and the other will be in phase. Sideband selection is accomplished simply by reversing the phase of either one audio or one carrier channel. In practice the audio channel method is usually used. While a good method, the phasing method requires accurate component matching, narrow tolerances, and accurate setup. Nevertheless, it has been successfully used in amateur radio equipment, mainly in the past when separate transmitters and receivers were used. Today transceivers are mainly used, and the phasing method is not used as a filter is still needed anyway for the receiver section. In the future digital signal processing will undoubtedly be used instead, eliminating or simplifying the filter required. While other methods exist, undoubtedly most SSB generation will be done for a while using the filter method. SSB crystal filters have come down somewhat in price due to manufacturing and design improvements, as well as increasing market demand, keeping the filter method the most popular approach. This could change as digital signal processing techniques may replace the filter as the most common method of SSB generation in the future.

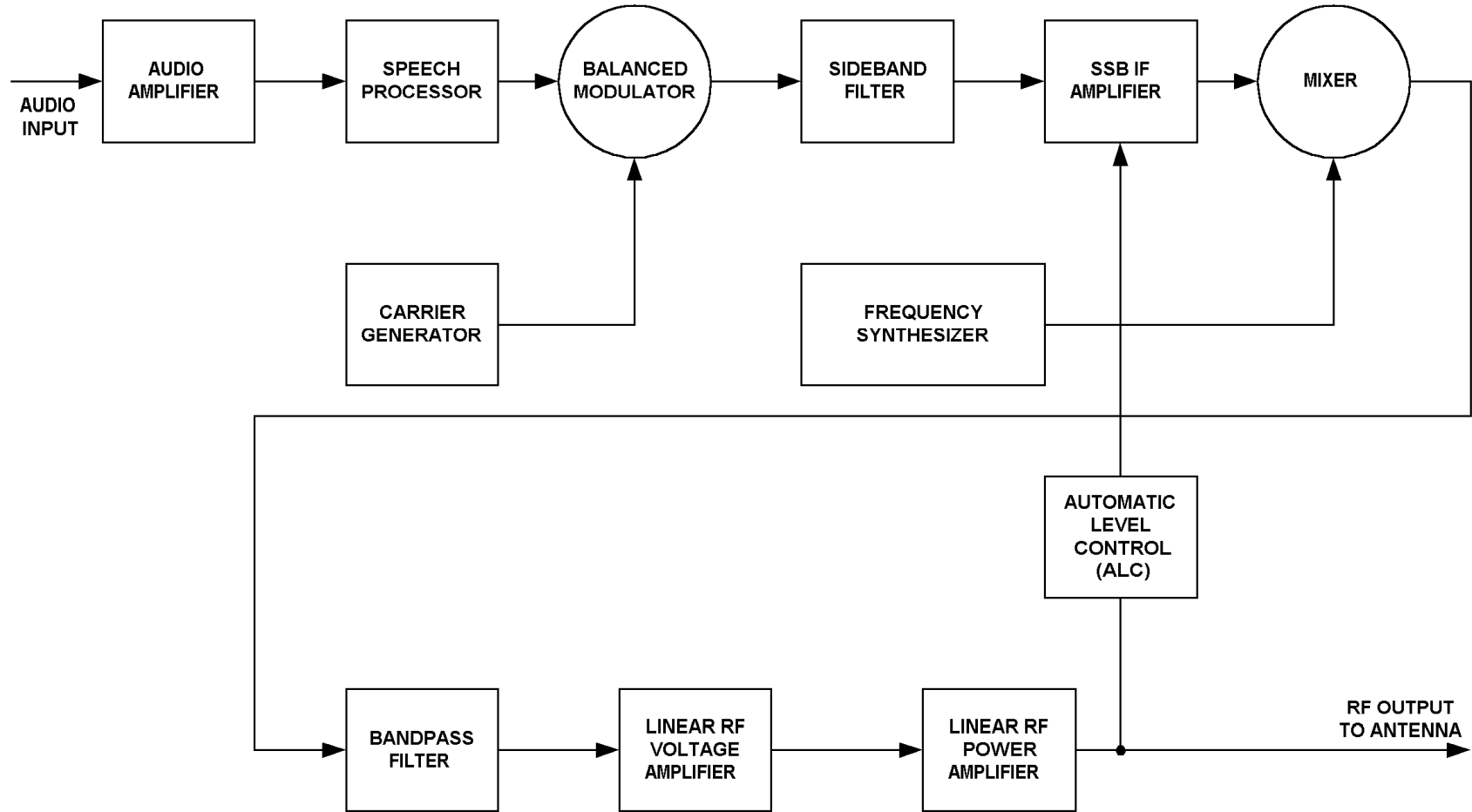
Reception of SSB signals generally follows the reverse of the generation process. A look at the spectrum of a voice SSB signal will show that it is simply the input audio input spectrum shifted up into the RF region. For example, consider a 10.000 Mhz voice frequency SSB signal. If the USB mode is used, the transmitter will produce a signal having frequency components of 10.0003 to 10.0030 Mhz, or simply 300 to 3000 Hz (0.3 to 3 KHz) shifted arithmetically higher in frequency by 10 Mhz. To receive this signal, we must simply shift it back down to the audio region. For LSB, the transmitted spectrum is also inverted, the higher voice

frequency components producing lower transmitter frequency components. A simple mixer (in this application commonly called a product detector, same mixer circuit, different name) can be used for this function and indeed a receiver can be built in which an antenna is connected to a mixer that is fed with an LO. If the LO is exactly the same frequency as the suppressed carrier of the input SSB signal from the antenna, the product detector output will be the original audio that modulated the SSB transmitter. This type of detector when used with an antenna and a suitable audio amplifier will make up a receiver commonly called a direct conversion receiver. Useful for SSB and CW (Morse code) reception, this scheme is popular for low cost ham radio receiver construction and eliminates much RF circuitry. The LO must be stable and have good noise characteristics, and a low noise audio amplifier is necessary, but sensitivities around a microvolt can be obtained. The bandwidth is that of the audio amplifier. Disadvantages of this receiver are lack of sideband selection, poor RF selectivity, lack of AM reception capability due to LO beating with the AM carrier, and susceptibility to RF overload, as generally no AGC is used. However this receiver provides a lot of performance with very little circuitry and is superior to and easier to use than a regenerative receiver for SSB and CW reception.

The carrier must be reinserted at the detector within a few hertz of the original carrier otherwise the frequency of the audio output will be shifted from the original by an amount equal to this difference. For speech 50 hertz is acceptable, but for quality 10 Hertz is desirable, and for music or where frequencies are critical, one hertz will be better. In order to assist in this, a pilot carrier may be transmitted. This is a residual sample of the original carrier sent at a known level, i.e. -30 or -40 dB down so that it is not very noticeable. A phase locked loop at the receiver is used to lock on to this pilot carrier, ensuring accurate tuning. Modern SSB equipment used by amateur radio operators can easily hold frequency within 10 Hz so this is not often done. If the reinserted carrier is way off the SSB signal will sound like gibberish, often called the "Donald Duck" sound. If the carrier is way off, (a few Khz) and is placed on the opposite side of the signal, the recovered audio may actually be spectrally inverted, so that original low speech frequencies (300-400 Hz) are now at the high end of the audio band, near 3000 Hz, and the original audio components at the 3000 Hz end of the audio spectrum are now shifted down to the 300 Hz region. This is called "inverted speech" and this concept is used elsewhere to scramble an audio signal for privacy or security purposes. In practice this scrambling is done with special circuitry. An article by the authors of this column appeared on page 37 of the December 1993 issue of Electronics Now Magazine, using digital techniques to accomplish this.

Other than the requirements for an accurate and stable LO frequency and a product detector, a SSB receiver is generally a standard superheterodyne receiver with high RF performance in areas of dynamic range, noise floor, and stability, with special AGC circuitry, as there is no carrier for AGC reference, as exists in an AM receiver. A SSB receiver usually has a separate envelope or synchronous detector for AM reception anyway, and has switchable AGC for each reception mode. In a

**transceiver system, often the same circuitry, run "backwards" is used for SSB generation and detection. This is called bilateral circuitry and will not be discussed here. The interested reader is referred to books such as the ARRL Radio Amateurs Handbook or the RSGB Handbook for details of this type of circuitry. This discussion of SSB techniques has necessarily been brief. Entire books have been written on SSB, but this is impossible to cover in a short article. The next part of this discussion will discuss frequency modulation methods and techniques**

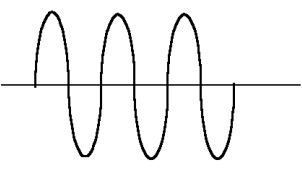


**BLOCK DIAGRAM SSB TRANSMITTER USING FILTER METHOD**

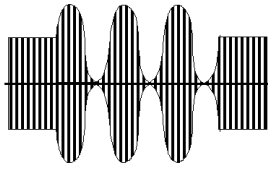


**AF WAVEFORMS**

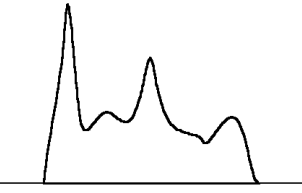
**RF WAVEFORMS**



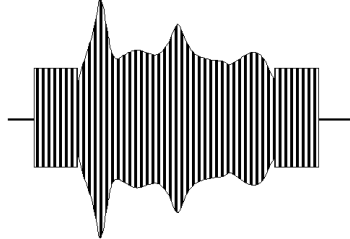
AM  
TRANSMITTER



ENVELOPE OF RF SIGNAL  
CORRESPONDS TO  
MODULATING WAVEFORM

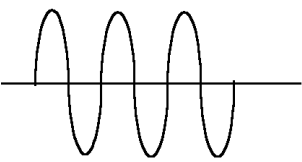


AM  
TRANSMITTER

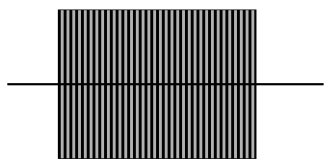


**AF WAVEFORMS**

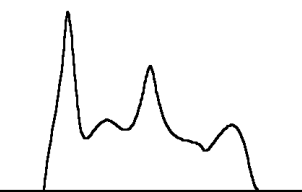
**RF WAVEFORMS**



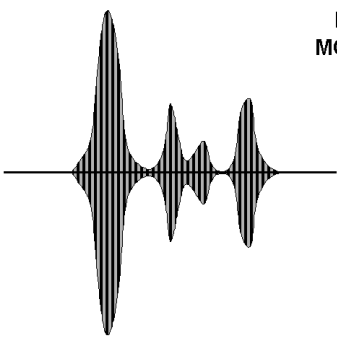
SSB  
TRANSMITTER

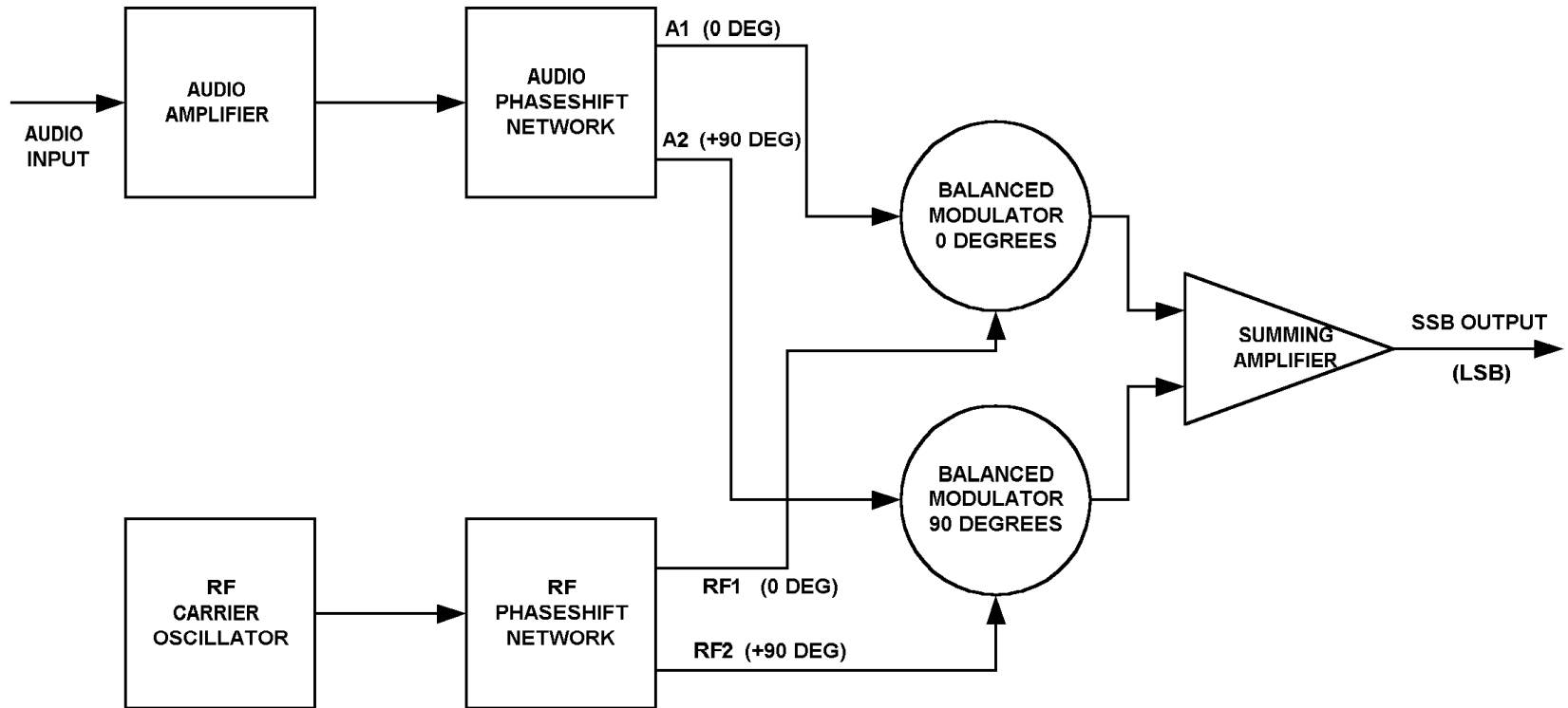


ENVELOPE OF RF SIGNAL DOES  
NOT CORRESPOND TO  
MODULATING WAVEFORM



SSB  
TRANSMITTER





**IF:**      $A1 = \sin W_m T$   
            $A2 = \cos W_m T$   
           and  
            $RF1 = \sin W_c T$   
            $RF2 = \cos W_c T$

$$\sin A \sin B = \frac{1}{2} \cos(A-B) - \frac{1}{2} \cos(A+B)$$

$$\cos A \cos B = \frac{1}{2} \cos(A+B) - \frac{1}{2} \cos(A-B)$$

**THEN:**

$$\text{SSB OUTPUT} = \frac{1}{2} \cos(W_c T - W_m T) - \frac{1}{2} \cos(W_c T + W_m T) + \frac{1}{2} \cos(W_c T + W_m T) + \frac{1}{2} \cos(W_c T - W_m T)$$

**SUMMING TERMS AND SUBSTITUTING:**

$$\text{SSB OUTPUT} = \frac{1}{2} (\text{LSB}) - \frac{1}{2} (\text{USB}) + \frac{1}{2} (\text{USB}) + \frac{1}{2} (\text{LSB}) = \text{LSB ONLY}$$

## PHASING METHOD OF SSB GENERATION