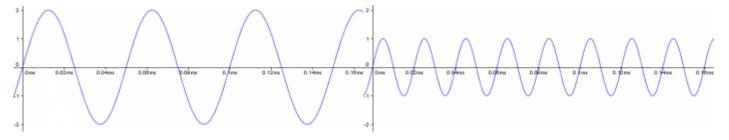
Wave and Modulation

Here we dive a little more deeply into waves and three ways a "pure" radio wave (called the *carrier*) can be modulated to encode a voice signal (called the *baseband* signal): *AM, SSB, FM*. But first, let's look at the general characteristics of a wave.

Amplitude, Period, and Frequency

Look at the following two waves. How are they different?

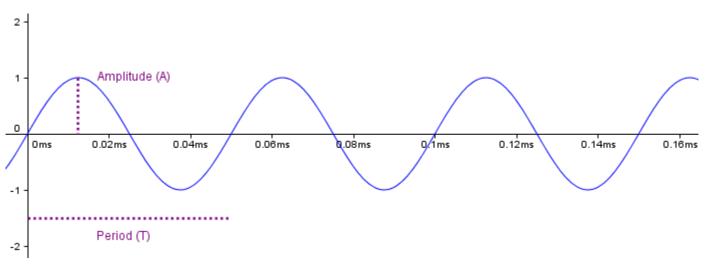


At first look:

- 1. the first one is "taller" than the second one. That is, it goes up and down higher and lower.
- 2. the first one is also "longer" than the second one. That is, it stretches sideways more. It's not as "tight".

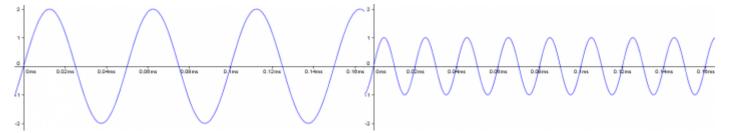
These two observations can be quantified very precisely as:

- 1. the *amplitude*: **vertical** length from the centre of the wave to its highest (or lowest) point.
- 2. the *period*: **horizontal** length of one complete cycle.



So the previous two waves have:

- 1. Amplitude = 2, Period = 0.05 ms
- 2. Amplitude = 1, Period = 0.02 ms



The amplitude is normally related to the strength of the signal (like the volume for sound).

Since the period is the amount of time it takes to complete one cycle, and the frequency (*f*) is the number of cycles in one second, the period and the frequency are inverses of each other:

<latex> $qquad \$f = \frac{1}{T} qquad \Leftrightarrow qquad T = \frac{1}{f}$

Let's pause for a minute here ...

In this course, we'll see a few formulas and it'll be tempting to memorize them but let's instead understand what they really mean...

Here:

- The period is the length of time it takes to complete one cycle and
- The frequency is the number cycles in one second.

So:

- if the period is half a second, we can fit 2 full cycles in one second.
- If the period is a quarter of a second, the frequency is 4.
- If the period is a tenth of a second, the frequency is 10.
- If the period is T seconds, the frequency is $\frac{1}{T} \le \frac{1}{0.5} = 2$, $\frac{1}{0.25} = 4$, $\frac{1}{0.1} = 10$

Right?

So for the previous two waves, the frequencies would be:

- 1. < latex>\$\$f = $frac{1}{0.05 \text{ kext} ms} = \frac{1}{0.00005 \text{ kext} s}$
- 2. <|atex>\$\$f = $frac{1}{0.02 \text{ kext} ms}$ = $frac{1}{0.00002 \text{ kext} s}$

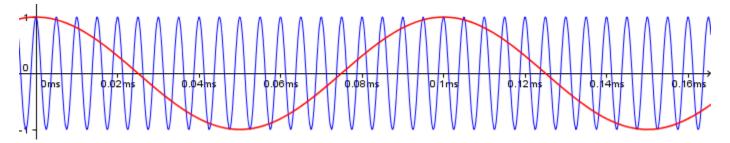
Recall that *Hz* means "cycle per seconds". That's why when we divide a number of cycles by time, we get Hertz.

Let's now look at three different ways to encode a signal on a radio wave.

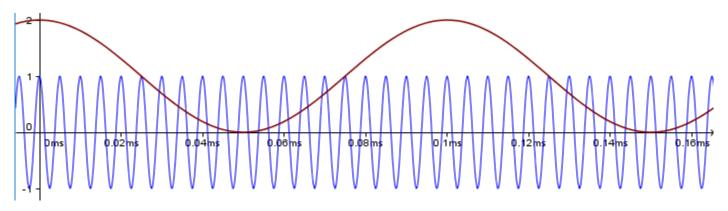
AM

AM stands for *Amplitude Modulation*. What this means is that the transmitted radio wave is obtained by changing the amplitude of a pure radio waves (the *carrier*) based on an audio signal (the *baseband*).

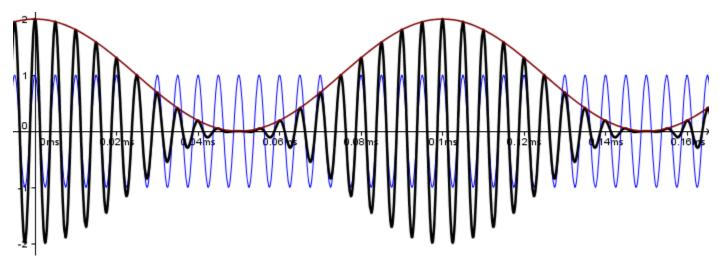
For example, let's transmit a single audio note of 10 kHz at a radio frequency of 200 kHz.¹:



Before we modulate the carrier, we raise the audio signal above zero to get an *envelope*:



Finally, we **multiply** the envelope and the carrier, which gives us a wave that has the same frequency as the carrier, but an amplitude that varies like the voice signal:



So:

AM Radio Wave = (Audio Signal + 1) × Carrier Wave

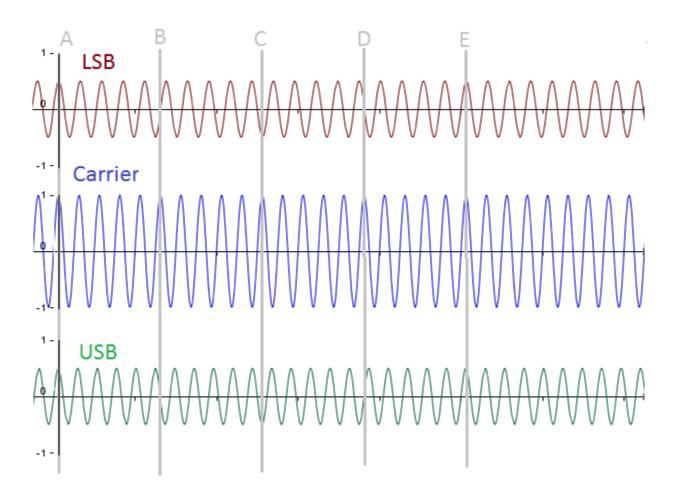
The incredible thing about the resulting AM broadcast is that the transmitted radio signal can also be seen as the *sum* of three pure sine wavesL

AM Radio Wave = LSB Wave + Carrier Wave + USB Wave

- LSB means Lower Side Band
- USB means Upper Side Band

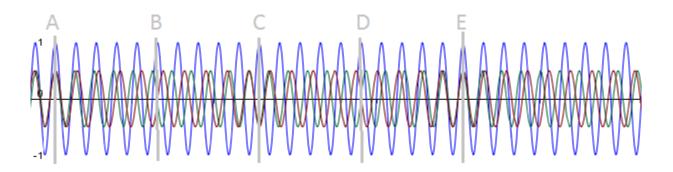
This is absolutely **not** obvious but let's see why it's at least plausible. Imagine we start with the following three

waves:



- An LSB Wave oscillating at 190 kHz with an amplitude of 0.5
- A Carrier Wave oscillating at 200 kHz with an amplitude of 1
- A USB Wave oscillating at 210 kHz with an amplitude of 0.5

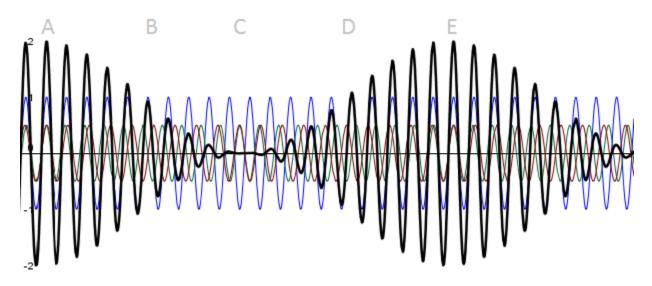
Now let's add them together:



- at point A, all three waves align so the sum is: 0.5 + 1 + 0.5 = 2
- at point B, the two side bands are opposite and cancel each other and only the carrier remains: 0 + 1 + 0 = 1

- at point C, the carrier is opposite the two side bands so the sum is: -0.5 + 1 0.5 = 0
- at point D, the same as point B is happening
- at point E, the same as point A is happening

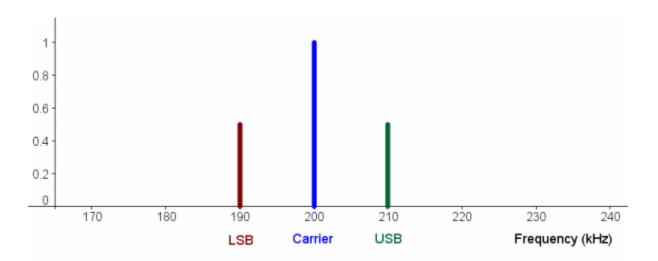
Here they are again with the final transmitted wave:



Note that the carrier is at 200 kHz exactly like the original carrier, but that the two side bands are 10 kHz lower and higher with half of the amplitude. Notice how the LSB Wave oscillates slower than the Carrier Wave, while the USB Wave oscillates faster.

Frequency Spectrum

An easier way to represent a radio signal is using a *spectroscope*. Instead of seeing the signal wave, it shows the strength of each frequency that makes up the sum of the signal. For example, the spectroscope of our 10 kHz note transmitted over a 200 kHz carrier would look like this:



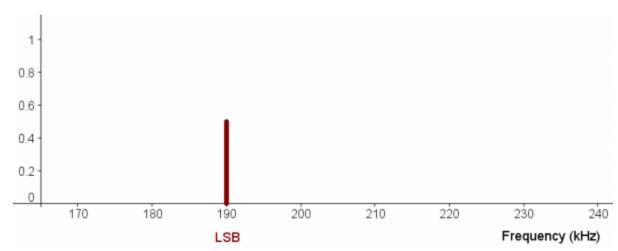
All this is saying is that the radio signal is composed of three pieces: a signal at 190 kHz with an amplitude of 0.5, a signal at 200 kHz with an amplitude of 1, and another at 210 kHz with an amplitude of 0.5.

There are three things to notice here:

- The two side bands are 10 kHz on each side of the carrier (same as the baseband signal!). It is that distance away from the carrier that represents the audio signal we want to recover.
- Most of the power is going into transmitting the carrier, which in itself doesn't carry any information, so that's a bit of a waste of energy.
- More fundamentally: even though we say that the signal is transmitted at 200 kHz, in this example, it is really contained between 190 kHz and 210 kHz. That is, it has a bandwidth of 20 kHz (210 kHz 190 kHz). This bandwidth is regulated and depends on the band used.

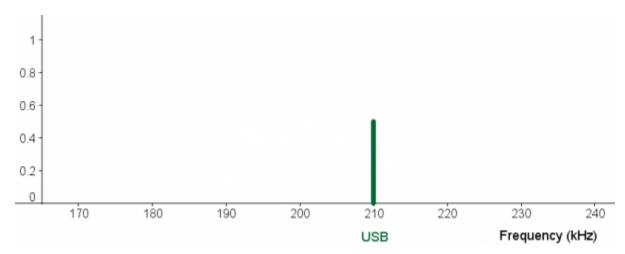
SSB

One way of saving power is to only transmit one of the side bands. In this example, the radio would be tuned to 200 kHz, but...



...for LSB, the spectroscope would look like:

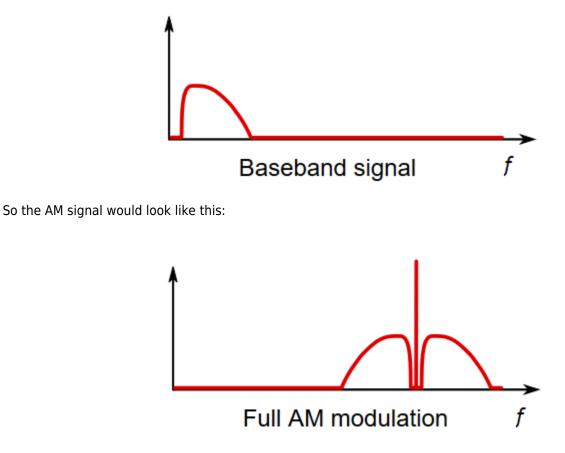
...for USB, the spectroscope would look like:



By itself, neither of these transmissions would carry the information we need (that the baseband signal was a 10 kHz note) since it's the difference between the sideband and the carrier that gives us that information. But if the receiver knows that this signal was generated by a transmitter at a frequency of 200 kHz, then the receiver can re-inject the missing carrier on its side.

This is why an AM signal is not too picky about being slightly off frequency (both the sidebands and the carrier are transmitted). But a SSB signal changes pitch if the receiver is not tuned precisely to the transmitter frequency.

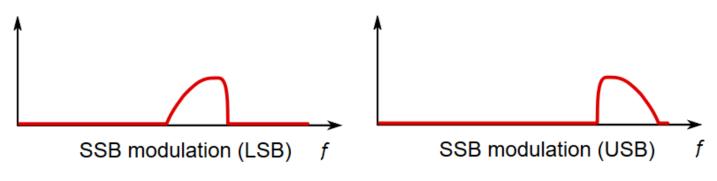
In reality, the voice we transmit contains a whole group of "notes" typically between 300 Hz and 3000 Hz (music could range between 20 Hz and 20,000 Hz). A typical voice signal (baseband) could look something like this:²⁾



Notice how:

- the carrier is a single line in the centre because unlike the sidebands, it is a pure sine wave of only *one* frequency.
- the two side bands are mirror images of each, which is why it's important that both the receiver be in the same mode as the transmitter.

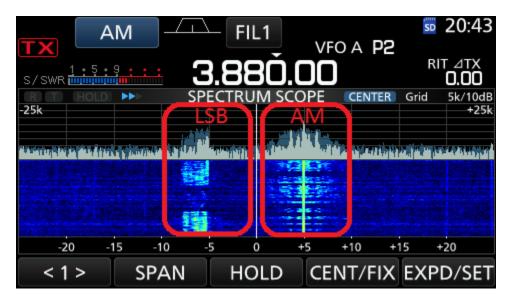
And finally, each individual sideband would look like this:



The two main advantages of using SSB (LSB or USB) are that:

- It takes less power to transmit the same information.
- It takes half the bandwidth.

Here's a screenshot of the scope from my IC-7300.



The radio is tuned to 3.880 MHz (where no one is transmitting), but there are two different conversations going on: one at 3.875 MHz using LSB, and another at 3.885 MHz using AM. The scope shows the recent history of the radio signal (called *waterfall*) where the present is at the top and the past at the bottom. Blue represent a low signal and yellow or red represent a high signal. Here are some things to notice:

	LSB (3.875 MHz)	AM (3.885 MHz)
Symmetry	The signal is on the left (low side) of where the carrier would be (at 3.875 MHz) and varies with speech.	The signal has a strong, constant carrier in the centre, and two symmetrical sides that vary with speech.
Bandwidth	About 3 kHz on the low side of 3.875 MHz	About 6 kHz (3 kHz on each side of 3.885 MHz)
Pauses	During pauses, no radio signal is transmitted.	During pauses, the carrier is still transmitted.
	An AM signal can be understood in LSB mode because it contains the lower side band required. But an LSB signal can't be understood in AM mode because both sidebands and the carrier are needed to process the signal.	

FM

FM stands for Frequency Modulation. What this means is that the transmitted radio wave is obtained by changing the frequency of the carrier based on the audio signal.

For example, let's again transmit a single audio note of 10 kHz at a radio frequency of 200 kHz using FM this time instead of AM:

×

This time, we don't simply multiply the baseband signal to the carrier (as in AM). Instead, we "compress" and "stretch" the carrier (ie, modulate its frequency) based on the baseband signal.

×

Here, the math is a bit more involved and requires at least 1st year calculus to understand but in a nutshell, if the carrier is <|atex> $c(t) = cos(2 pi f_c t)$, and the baseband signal is <|atex>, then the FM signal will be:

<latex>\$\$ \cos\Big(2 \pi f_c t + 2 \pi k \int_0^t s(\tau) d\tau\Big) \$\$</latex>

If this looks like Greek to you, don't worry; the math isn't important. The key concept to understand is that the highs and lows of the baseband signal are encoded in the horizontal compression (the frequency) of the radio wave: When the baseband is high, the radio signal is more compressed (its frequency is higher), and when the baseband is low, the radio signal is more stretched out (its frequency is lower).

Optional Details

For those interested in some of the mathematical details, see this optional page.



2)

Note that 200 kHz is lower than commercial AM broadcast and is **not** a ham radio frequency. I chose a ratio of 20:1 so we can see the effects on the graph

The next few images are from Single-sideband_modulation