

CRYSTAL SETS TO SIDEBAND

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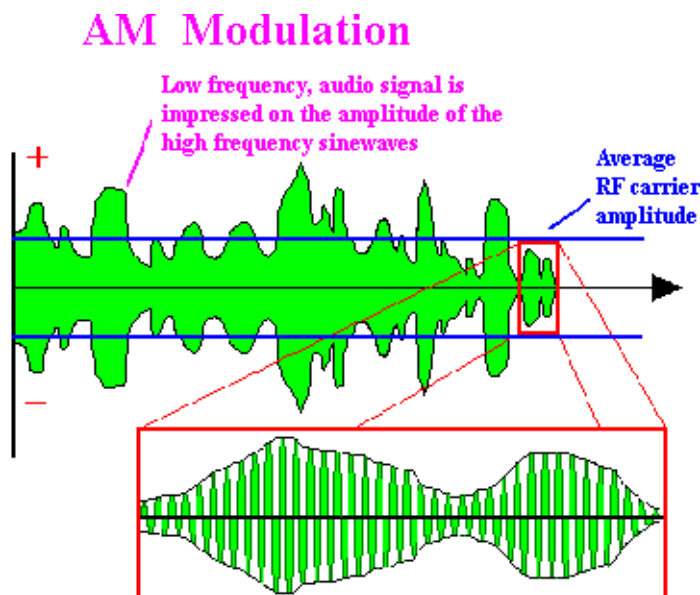
Chapter 17A

ANCIENT MODULATION & RELATED Topics

When I returned to ham radio in 1997, my ham friends told me that amplitude modulation (AM) was extinct. I was under the impression that SSB was the only mode of HF phone permitted. Later I learned that AM isn't actually illegal and there are a few diehards using AM on the 75 and 10 meter phone bands. I've also heard AM stations on 15 and 160 meters. In short, you might find a use for it. Besides, it's an interesting challenge to AM-modulate a transistorized transmitter.

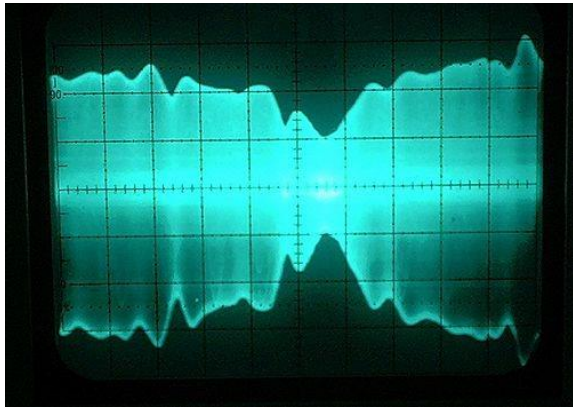
Homebuilt AM

Back in the vacuum tube days many of us built our own transmitters and AM modulators. My first AM transmitter was a modified Heathkit DX-20. The DX-20 was a 50 watt, CW-only, kit-built, vacuum tube transmitter to which I added a homebuilt AM modulator. Unlike SSB, AM could be added onto an existing CW transmitter. Rather than generate a low power AM radio signal and then amplify it with a linear amplifier, the usual method was to AM-modulate the final amplifier of the CW transmitter. This is done by impressing audio waveform onto the DC power line going to the last amplifier stage. For example, if you want to AM modulate a CW QRP like the ones described in Chapters 6 and 11, you can do that. Of course you'll need to buy crystals to cover the phone band or you could build a VFO like those in Chapter 10. However, before you go to all that work, maybe you had better find an AM equipped friend to talk to.

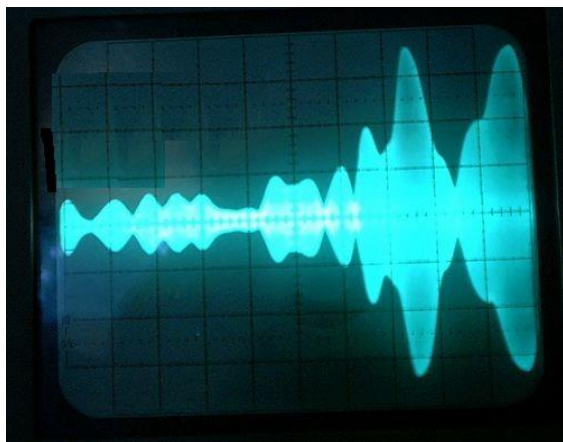


On an oscilloscope, the hallmark of AM is that, when you are **not** speaking, the RF carrier

wave runs continuously at an average power. That is, in AM the highest peak power and zero power only occur at the very highest voice peaks. Although I could see these transient peaks on the scope, they are statistically rare and I couldn't catch one. I also couldn't record a zero power level. The waveform below was typical of what I saw.



In contrast to AM, the RF output amplitude in SSB is always nearly **zero** whenever you aren't talking. Notice in the SSB oscilloscope picture below that each RF blip representing the audio starts from zero. It doesn't start from a halfway, continuous carrier level.



Plate, screen and cathode modulators

Formerly, there were three common methods of AM modulation. The "Mercedes" (highest quality) method was to use a *plate modulator transformer*. These large iron transformers impressed the audio signal onto the DC supply current. That is, as you talked, the DC input current rose and fell above and below the level of what it would be for a CW sinewave. For a 100 watt transmitter, this transformer was about the size of a softball, weighed a ton and cost like crazy. The transformer was driven with a big audio amplifier that put out at least 50% of the CW carrier power. In other words, the plate modulator circuitry was nearly as large and expensive as the rest of the transmitter.

"PLATE" OR "COLLECTOR" MODULATOR

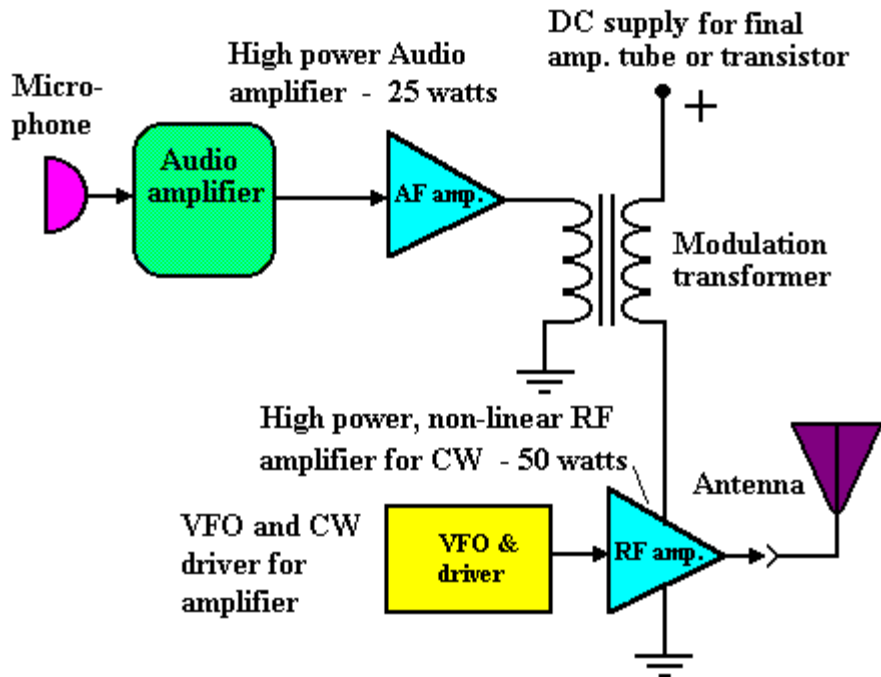


Plate modulation is the most desirable method because the power of the audio amplifier is added to the DC current input to the final amplifier of the AM transmitter. A high current version of the audio waveform is added to and subtracted from the normally constant DC current used to power the CW output amplifier. Voice peaks can rise to twice the power of the transmitter in CW mode. However, if more than 100% modulation is attempted, the voice minimums impressed on the CW carrier will try to go "below zero." In other words, the transmitter output will drop below the audio waveform minimum, breaking up the audio. This "over-modulation" sounds distorted and unpleasant. If the modulator is capable of over-modulation, the microphone gain is simply turned down to maintain peaks that are 200% of the carrier (CW) level. This is regulated by watching a "VU" output meter and keeping the maximum deflections of the needle out of the red zone on the right of the scale. "Disk jockeys" on commercial radio stations always have VU meters in front of them to monitor their speech and music volumes.

Screen modulation

The "Toyota" and "Bicycle" approaches to AM modulation were to modulate the gain of the final amplifier tube by impressing the audio on the screen or cathode, respectively. Screen modulators maintained the carrier level at 50% of the CW level. Then audio modulation varied the power level between 0% and 100%. Screen modulators usually sounded good, but were limited to 100% of the CW power on voice peaks. Since peak transients in voice waveforms are rare, the signal was almost never as powerful as the CW signal it was modulating.

Cathode modulation

Cathode modulation, sometimes called Heizing modulation, impressed an audio waveform across a large inductor connected between the final RF amplifier tube cathode and ground. As

you know, the current through an inductor can't change quickly. Because the inductor was large, several henries, the audio voltage appeared between cathode and ground, subtracting from the voltage on the plate of the vacuum tube. This meant that the RF voltage on plate the rose and fell the reverse of the amplitude of the audio signal. In other words, Heizing modulation produced "down modulation." *The power decreased on the voice peaks* instead of rising. When a receiving operator watched his receiver's S-meter, the needle when down instead of up on voice peaks. It sounded surprisingly normal, but was an inefficient use of RF power output.

Cathode modulation required far less audio power than plate modulation and, when I was in high school, it was easy to build and afford. As I recall, I found the schematic in the Popular Electronics magazine. I once used it to talk to a fellow in England, 5,000 miles away. He could barely hear me, so I switched to CW and we finished the QSO with perfect copy. Hi !

A 10 METER AM HANDHELD

RECREATING A GOLDEN-OLDIE IN 2018

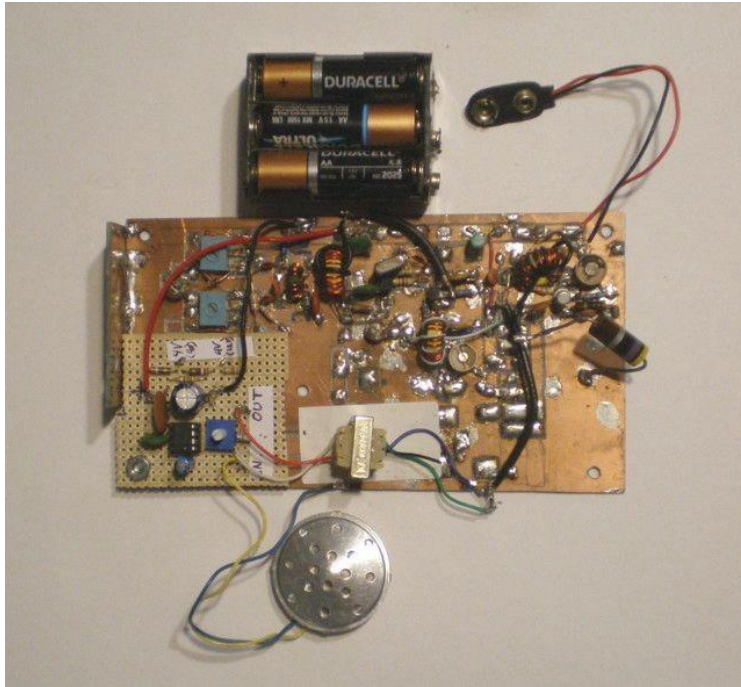
I know the modern world has little or no use for AM modulation, but just for fun I built a 10 meter AM transmitter. Earlier in the book I mentioned the 10 meter AM-modulated handheld transceivers that my friends and I built while we were in high school. The design was straight out of the 1946 ARRL handbook. The circuitry was accomplished with just two small glass vacuum tubes! A 3V4 pentode was the audio amplifier while a 3A5 dual triode served as both transmitter and regenerative receiver. In Chapter 7B we learned that the oscillating frequency of a regenerative receiver is the operating frequency. Therefore the same oscillator can become a transmitter just by connecting it directly to the antenna. As I recall, there was a small speaker transformer that served as the plate modulation transformer. A small triple-pole, double-throw switch served as the push-to-talk switch. The switch moved the circuit blocks around to convert the receiver into a transmitter. It had a simple inductor-loaded whip antenna that stuck up about 18 inches. The metal enclosure of the handheld and the operator's body helped to serve as a counterpoise for the antenna.

I don't claim that those old walkie-talkies were free of deficiencies, but I managed to talk 2,000 miles while sitting on the chimney of my parents' house. Long range with feeble transmitters and tiny antennas was possible because of the fantastic sunspot numbers of the late 1950s - 250 spots or more. Maybe, if you are *really* young, you might see propagation like that again. Of course our walkie-talkies also worked well for talking about a mile or so, much like modern "Family Radio Service" hand-helds.

As you would expect, these old time regen walkie-talkies were quite unstable and wandered up and down the band, unacceptable in the modern world. There was also a subtle difference in frequency between receive and transmit. When we conversed with our friends, the frequency "walked the band" in addition to the basic instability. I forget whether they walked up or down, but we were constantly tweaking the tiny tuning capacitor to keep up with the conversation. Bear in mind that there were no cell phones back then. Walkie-talkies were only available to the Army, Marines and Dick Tracy. Dick Tracy was a cartoon detective who chatted with his police station using his science-fiction wrist TV. We young hams were really hot stuff with our walkie-talkies.

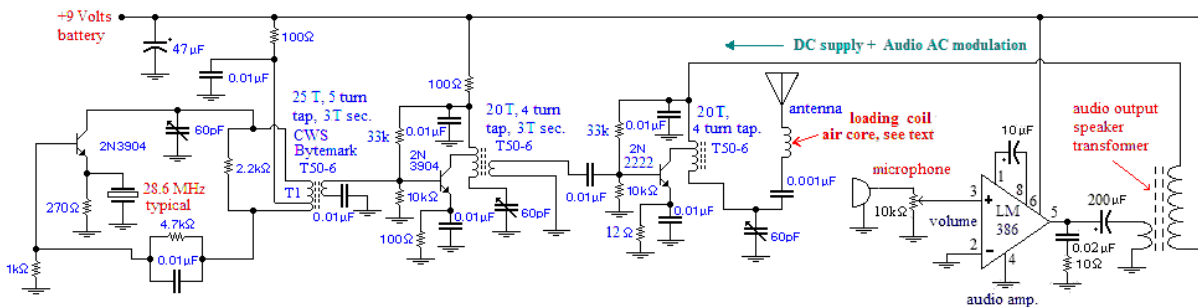
Reminiscing about those wonderful old radios made me wonder if I could produce a modern version usable on the air. I will probably never have anyone to talk to, so until then, I have no plans to package it properly. I built the transmitter on an old PC board. Since audio amplifiers are rather generic, I put it on a separate little board so I can use it again in another prototype. I plan to build the receiver breadboard on a separate board. That way I can test the reception from its complimentary transmitter.

Breadboard prototype of the handheld 10 meter AM transmitter



First of all, the trim pots at the upper left of the board are experiments with making a phase-canceling SSB, which so far hasn't worked. If I ever get the SSB working, that would make this project truly practical! Here's the schematic for the AM transmitter:

10 Meter Handheld Transmitter



As you can see, this is a "collector-modulated" AM design. An audio amplifier drives a transformer that impresses an audio waveform onto the final amplifier DC supply line. It does not modulate the RF oscillator or buffer stages. The microphone on the right side drives an LM386 integrated circuit audio amplifier. The LM386 drives a small speaker transformer which

impresses the AM modulation onto the DC input to the final RF amplifier. The microphone can be a crystal type, as in the photograph above. I tried a smaller, "electret" type which also worked well. The electret needs a 4 volt DC supply which can be simply a resistive divider. I used a 3.9K resistor in series with a 5.1 K resistor. Six AA cells power the transmitter and modulator for RF power of roughly 200 milliWatts.

The transmitter oscillator is crystal controlled and can operate between 28.3 MHz and 29.0 MHz which is the 10 meter SSB/AM phone band. This corrects the frequency instability of the old 1946 ARRL design. 29.0 MHz to 29.7 MHz is supposed to be reserved for FM. I once worked briefly for a little company that made animal tracking collars. Wild animal tracking collars were not a growth industry and sadly the company died. One result of this experience was that I inherited a collection of VHF overtone crystals. Voila! The primary frequency of an 86 MHz third overtone crystal was exactly in the right frequency range.

Antennas

For testing I transmitted into a dummy load which is the 2 watt resistor on the right side of the photo. If the antenna were the usual 50 ohm load like my breadboard, the final RF transformer would have a secondary winding with four or five turns. However, in the real world we need a practical small, 10 meter antenna. If size weren't an issue, I would use a quarter wavelength, 8 foot tall, vertical whip antenna. The whip can be shortened by using an air-core (not powdered iron) "loading coil." The loading coil "slows" the traveling RF wave down so that a much shorter structure will resonate as if it were longer. An air core is used because we want the inductor to leak RF out into the world and not confine it in the powdered iron.

Any short whip antenna will be very high impedance, at least hundreds of ohms. High impedance means that the voltage at the base of the antenna should be as high as possible. The simplest way to do this was to connect the antenna to the tuning capacitor at the bottom of the final amplifier resonating tapped inductor. Tuning this capacitor will also tune the antenna. Another approach would be to build a 10 meter version of the QRP antenna tuner shown in Chapter 9 which can load very high impedances. The method diagramed above is much less complicated.

The 1946 ARRL Handbook handheld antenna was about 18" tall. Unfortunately, I can't find a copy of that article. As I recall it was perhaps 10 or twenty turns of wire around a half inch diameter plastic or wooden coil form. The antenna whip was screwed into the top the form. Do not wrap the coil around the metal whip, the metal will drastically change the inductance of the coil. When I finish a practical receiver for this project, I'll experiment with the proper number of turns for the best performance over say, 100 yards distance. A coil with a sliding tap would make this adjustment easy. It would be like the Verizon cell phone man on TV... "Can you hear me now?"

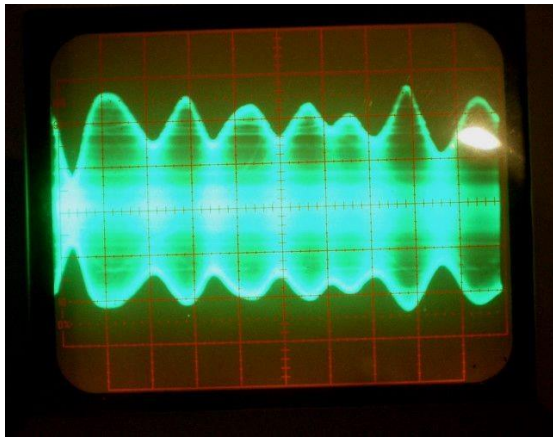
The AM modulation transformer

An obstacle to building plate-modulated or collector-modulated AM transmitters is that adequate modulation transformers are extinct in modern catalogs. Another barrier was that I didn't understand how a modulation transformer should be designed. Should both windings be high inductance? Low inductance? One of each? I could rationalize it different ways. I resorted to trial and error:

I first assumed I needed two low voltage, high current windings. I made one by using two huge filament transformers hooked together back to back. The only windings I wanted were the two low voltage 6.3 VAC windings, so I wired the two 120 VAC windings together. This double transformer made the equivalent of a single transformer. It worked fairly well - if you don't mind a pound of iron transformers to accomplish simple 1:1 voltage isolation. I had trouble getting more than about 30 percent modulation.

Next I tried a single filament transformer, 120 VAC to 6.3 VAC, with the low voltage winding driven by the audio amplifier. This worked great producing 100% modulation, but of course it was much too heavy for a handheld. I reversed the transformer driving the high impedance winding with the audio signal and the low impedance winding modulating the RF amplifier. This produced virtually no modulation at all.

Next I tried some tiny speaker transformers which I had rescued from transistor radios. The smallest ones worked poorly, but the larger one shown above worked well. I believe the input to output impedance ratio is about 1000 ohms to 8 ohms. Remember that these are not resistances, but rather approximate AC resistance (impedance) at audio frequencies. DC resistances are typically numbers like 80 ohms to 2 ohms. Resistance depends on the internal wire diameter, not the winding inductance. When I tried the centertap on the high impedance winding, the modulation decreased noticeably. So you should use the entire winding. The little iron core is 3/4" by 3/4" in its longest dimensions. Here is a typical RF waveform recorded on the dummy load resistor:



The speech shown above is from a Walkman headphone held up against the microphone. I listened to the speech on my ham receiver tuned to 28.6 MHz. Using headphones the speech quality sounds excellent to me. As you would expect, when I tried to listen to the receiver using the loudspeaker, the audio feedback produced an awful screech.

I noticed that when I turned up the microphone gain to produce frequent 100% modulation peaks, the speech often sounded "scratchy." Another difficulty I had with testing was that my bench power supply contributed high levels of 60 Hz hum on my 28 MHz signal. I couldn't see the hum on the scope waveform, but when I powered the transmitter with the 9 volt battery shown above, the hum vanished and the modulation sounded first rate in my big ham receiver.

Obviously, my next project should be a compatible miniature receiver. Because the signal

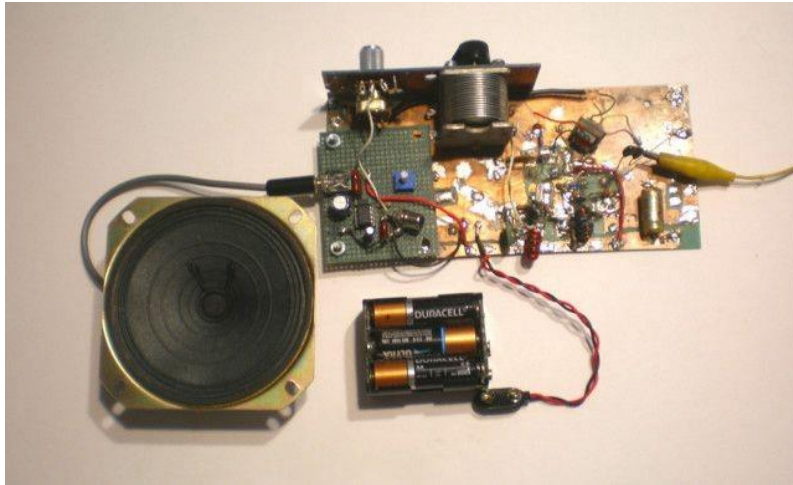
is AM, a regenerative receiver is the simplest choice, much like the one in Chapter 7B. If I can get the SSB phase canceling generation to work, a product detector would be a better choice.

A REGENERATIVE 10 METER RECEIVER

Crystal Controlled Regeneration

Here is my first attempt at a 10 meter AM receiver prototype which, hopefully, could be merged gracefully with the above transmitter to make a complete walkie-talkie. Notice that if successful, it can also be used to receive SSB or even FM.

While I was working with regenerative receivers, I was impressed that the oscillator was simply a self-oscillating amplifier with an L-C tuning circuit. Why couldn't the L-C circuit be a quartz crystal? Perhaps the frequency channel could be changed just by swapping crystals, without any L-C tuning needed. That would make tuning around for a specific frequency unnecessary. I have never heard of such a circuit. Who knows? I may be one of only 10,000 guys who have invented this before.



Prototype breadboard 10 meter regenerative receiver

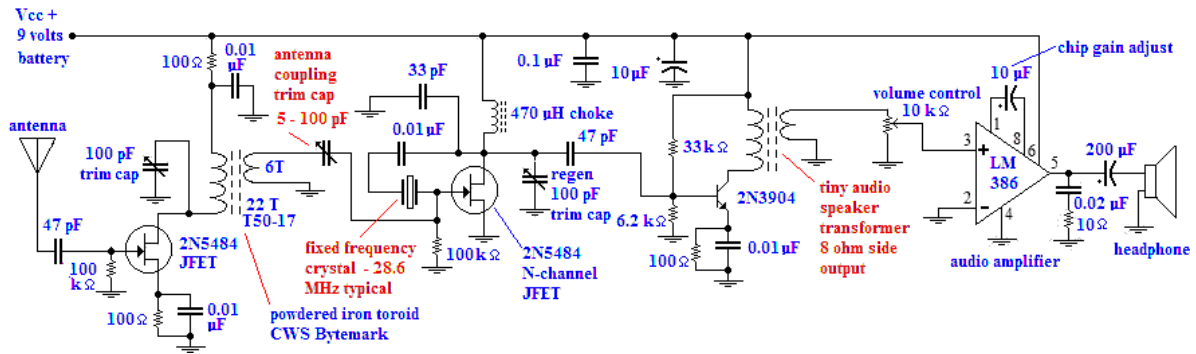
The antenna is the yellow test lead coming in from the right. The green perf-board is the audio amplifier using an LM386. I added the little blue pot to prevent the audio amplifier from oscillating at the highest volume settings. Obviously the enormous speaker isn't desirable for a hand-held. However, it sure makes loud, clear sound. Also, the large 365 pF variable capacitor can be replaced with a smaller 100 pF variable.

At first I tried the direct approach by substituting a crystal for the L-C circuit. I used the JFET transistor regen amplifier circuit from the regen receiver presented in Chapter 7B. It refused to oscillate. I've noticed that most JFET crystal oscillators use the drain-to-gate crystal placement. This configuration puts a relatively high voltage across the crystal which causes it to heat. Heat causes the frequency to drift, so I usually don't use this circuit. If heating causes noticeable drifting, we can reduce the voltage across the crystal by putting a small capacitor in series with the crystal. A small capacitor reduces the oscillation amplitude, so I began with a large capacitor relative to 10 meters, 0.01 μ F. I have not yet seen any reason to change. Crystal

drift is not nearly as important in receivers as it is in transmitters.

As with the regen in Chapter 7B, the oscillation amplitude, the regeneration, can be adjusted by the variable capacitor that shunts the drain RF voltage to ground. In my prototype I used a large 365 pF capacitor which was at least 3 times bigger than necessary. 100 pF would be much more appropriate. Hopefully in a well-tested, hand-held design, this capacitor could be a tiny trimmer capacitor set to an optimum level and would not need routine adjustment.

10 Meter Crystal Controlled Regenerative Receiver



Unfortunately, I didn't have another matched 86 MHz overtone crystal with a primary frequency in the 10 meter phone band so I couldn't properly test it with the transmitter. Using a 28.0 MHz crystal I could receive acceptable audio from the 10 meter transmitter prototype described earlier, but only at close range. To do this, I had to retune the RF pre-amplifier L-C circuit to 28.6 MHz and increase the coupling capacitance trimmer to overwhelm the crystal. This caused the regeneration to oscillate at 28.6 MHz. Obviously, this is not how it's supposed to operate, but it did demonstrate that the receiver works.

Next I retuned the input pre-amp to 28.6 MHz and generated an on-frequency AM modulated 28.6 MHz signal using my RF generator. The receiver can pick it up from several feet away, even with no antenna on either device. I'm confident the receiver and a complementary transmitter would work well together with matched crystals. I had another crystal in my collection labeled "28.2 MHz." When allowed to oscillate on its own, its natural frequency was 9.4 MHz. Oops. In short, a crystal for this application needs to have its primary frequency in the 10 meter phone band. It can't be the cheaper third-order harmonic crystal, regardless of what the label on the case may say.

The regen oscillator RF preamplifier

Like all regen receivers, this one needs an RF pre-amplifier to reduce the RF radiation from the oscillator and also to boost the signal strength. I used a standard JFET amplifier which immediately worked perfectly with no modifications. It was super easy to tune. I coupled the preamplifier input to the regen crystal oscillator using a 47 pF capacitor. Watching the output from the pre-amp with a scope, it tuned precisely to the crystal frequency. If 2 or 3 closely spaced frequency crystals were used, it shouldn't be necessary to retune the L-C filter. With only one tuned pre-amp L-C circuit, the reception band is fairly wide.

To mix the incoming signal with the regen oscillation, I simply wired them both to the JFET gate. A variable trim capacitor tunes the amplitude of the input. The degree of input coupling is a trade-off. *The more input coupling, the more wideband the receiver input.* I can hear my signal generator 100 KHz off frequency when the input level is high. Conversely, with the regeneration completely squelched using the variable capacitor on the drain, I can't hear anything but a little background noise. Very strong off-frequency 28 MHz signals dominate the regenerative oscillator and the crystal becomes irrelevant.

How does a regenerative detector detect?

When I built the 30 meter/ 40 meter regen in Chapter 7B, I used a huge, iron, multi-Henry audio choke in the circuit. Hand-helds are supposed to be light weight so I was determined to accomplish the same result with tiny RF chokes. Looking at the classic waveform of a modulated AM RF signal, I realized that the waveform envelope wasn't really an audio signal and we weren't really "detecting" or rectifying it. The signal is more like the output from a mixer or a product detector which is a combination of several frequencies. Instead of rectifying the complex waveform, we obtain the audio from a product detector by simply filtering out the RF with a choke and a small capacitor. In a product detector we used an RF choke in series with the output and ... voila! The output only contained the audio component.

I reasoned that an audio frequency speaker transformer should pass RF poorly. This worked beautifully and I was able to use a really tiny speaker transformer salvaged from a small battery powered radio. This speaker transformer was essentially the same as the modulation transformer in the above transmitter prototype, but physically much smaller. However, the transmitter transformer needed to be much larger with higher inductance and more iron. The little transformers worked poorly in the transmitter.

The LM386 audio amplifier chip is "programmed" for high gain (x200) by putting a 10iF capacitor between pins 1 and 8. This is too much gain at high settings and sometimes the chip oscillated making a horrible screech. I added a 10K trim pot in series with the 10uF cap to reduce the gain to prevent oscillation.

Assembling a usable hand-held

To convert the separate regen receiver and transmitter into a compact hand-held, the two circuits should share as many components as possible. The regen oscillator should serve for both the transmitter and receiver. The LM386 amplifier can serve as the microphone amplifier and the headphone/speaker amplifier. The input preamplifier in the regen can serve as the buffer for crystal oscillator in the transmitter. The transmitter final amplifier probably needs to be dedicated to that purpose. Now all we need is a small, ten or twelve-pole, double-throw PTT switch to rearrange all those circuits. Ha! Since the power levels are low, maybe several 4066 CMOS integrated analog switches could do some of that switching? Failing that, we can build both complete units and cram them both into a bigger box. You're the engineer. Figure out a solution!

Using the 10 meter walkie-talkie as an RF generator

The new Maunder Minimum

I doubted I would ever find a use for my 10 meter AM nostalgia walkie-talkie. That's why I didn't package it. I found a use for the transmitter. It's a test signal generator for when the 10 meter band is dead.

Back in the 1950s and 1960s we hams were spoiled by routine high sunspot numbers. The HF propagation all over the world was amazing on 10 meters and even 6 meters. Now the sun seems to be entering another "Maunder Minimum." These were the years between 1645 and 1715 when the sun had very few sunspots for several sunspot cycles. This caused a cold spell here on Earth known as "the little ice age." It also caused lousy HF propagation - not that anyone noticed back then. During the summers of 2019 and 2020, some propagation was available a few hours a day on 30 meters and perhaps an hour or two on 20 meters. To my amazement, 40 and 80 meters seemed dead day and night.

FT-8

The digital transmission mode "FT-8" was invented just in time for this new minimum. FT-8 uses a computer to decode musical tones sent v-e-r-y -- s-l-o-w-l-y. The FT-8 program and computer interface work with commercial HF ham transceivers to transmit and receive very short *standard* messages that can be even 20 dB below the atmospheric noise level. So an HF band that sounds completely dead to the human ear can often communicate around the world. Sadly, the messages it automatically sends and receives are just call letters and the signal strength in dBm. Still, for hams who collect DX contacts, it is way better than a dead band even though there is little for the ham to do himself. The computer simply offers a list of the call letters it is receiving at the moment. The ham clicks on the station he wants, and after a minute he gets a typed out cryptic reply, basically the ham's own call letters and signal strength. It's kinda like looking in a radio mirror - Ah! I have a strong signal in Patagonia today!

Naturally computer gurus immediately thought of using FT-8 to make the entire station automated. The computer "talks" to stations all over the world and, when the ham returns from a two week vacation, he finds that his computer has "worked" thousands of stations on every continent. This was too much high-tech silliness even for the FT-8 inventor. It might be legal, but it isn't human communication.

Bob, N0RN, is a DX hound and the lack of signals during 2020 was frustrating. He finally decided to try FT-8. Because Bob likes to have some understanding of how his equipment works, he built his own FT-8 interface that connects the computer to the HF transceiver. The FT-8 software must be initialized by selecting the HF transceiver model, interface model and the variations in operating modes. The lists of choices were quite extensive. Bob was sure he made all the right choices, but no matter what he selected, it didn't work. After weeks of frustration, he finally bought a commercial interface. The commercial interface didn't work either. Eventually Bob noticed that at the bottom of a long list of interface models there was a choice labeled "other." "Other" worked perfectly and so did his homebrew interface.

The HF signal generator



I have been re-building the single conversion HF receiver described in Chapter 7C. Because there are rarely any signals on the HF bands these days, I needed a continuous, modulated HF signal to listen to. It is hard to test and adjust a receiver, if there are no signals to listen to. The little 10 meter transmitter was ideal. I wired up the proper connectors so I could modulate it with my Walkman radio. I had to set all the audio gain levels near zero, but it works beautifully transmitting into a 47 ohm resistor. Even the music is high-fidelity. Hopefully, nobody outside my basement can hear the rock music I am re-transmitting on 28.6 MHz.

Modern Transceiver AM Generation

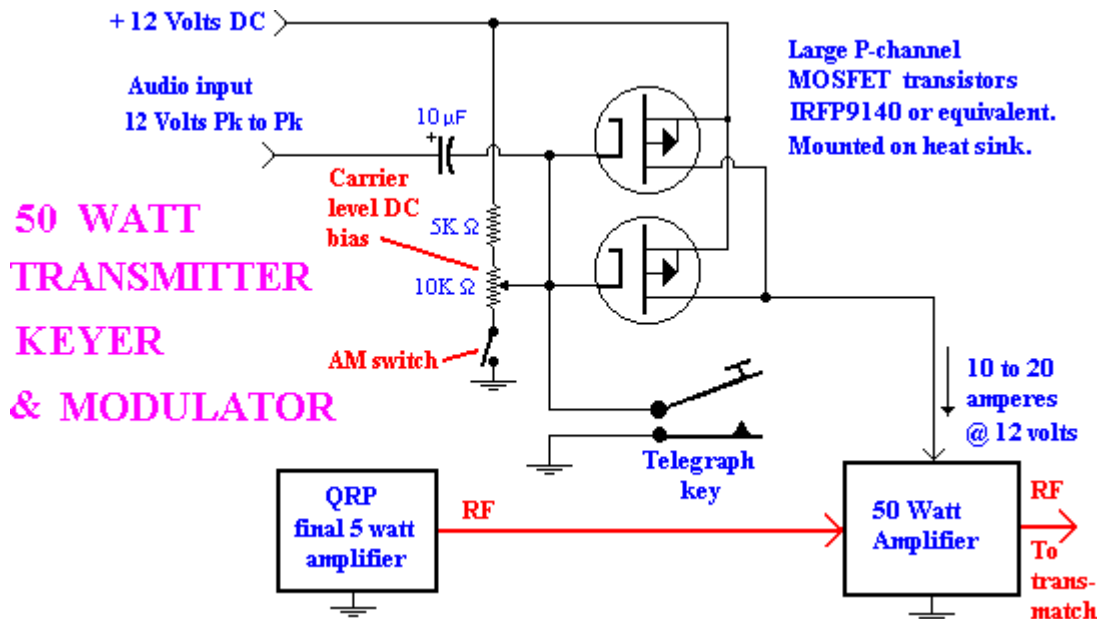
How do modern 100 watt commercial transceivers generate AM? Most modern SSB transceivers have the capability to generate AM modulation. To get into this mode, you read your manual for 20 minutes, bring up menu #26, push button numbers 14, 7 and 12 and you're done. That wasn't hard, I guess. Did you learn anything?

Let's suppose that you're a homebrew fanatic and wish to scratchbuild your own high power AM rig using transistors. Is that hard? Hmmmmm. Well, for one thing, transistors don't have cathodes and screen grids. Emitters are analogous to cathodes but, as explained above, cathode modulation wasn't all that great. Another difference between tubes and transistors is that, for the same power levels, *the final amplifier transistor has DC currents about 50 times larger.* So for DC supply modulation, you must impress high current audio signals onto the 12 volt DC power supply line. As a rule of thumb, the audio power needs to be about 50% of the RF output power.

Modulating a transistorized 50-watt CW transmitter

I have a 25-watt, plate modulator transformer from the 1960s that was designed for use with a transistorized audio amplifier modulating a 50 watt vacuum tube transmitter. In other words, it was designed to modulate a vacuum tube final amplifier, even though the modulator itself was transistorized. In those days high power, high frequency transistors didn't exist. Consequently, transmitters were built with tubes, but audio circuits could be built with power transistors. Since my transformer had low impedance primary windings, I thought I could "run it

backward" and supply enough audio current drive to build an AM "collector modulator." I happen to have an old 10-watt vacuum tube hi-fi amplifier, so I used that to drive the high impedance winding on my modulation transformer. Sure enough, even with music my AM modulation sounded excellent when I broadcast into a dummy load. However, it only modulated about 30% of the carrier amplitude. In other words, I was wasting most of my RF power. I could have built a 25-watt vacuum tube audio amplifier, but I had a more modern idea. Why not use my MOSFET CW keyer as an audio modulator?

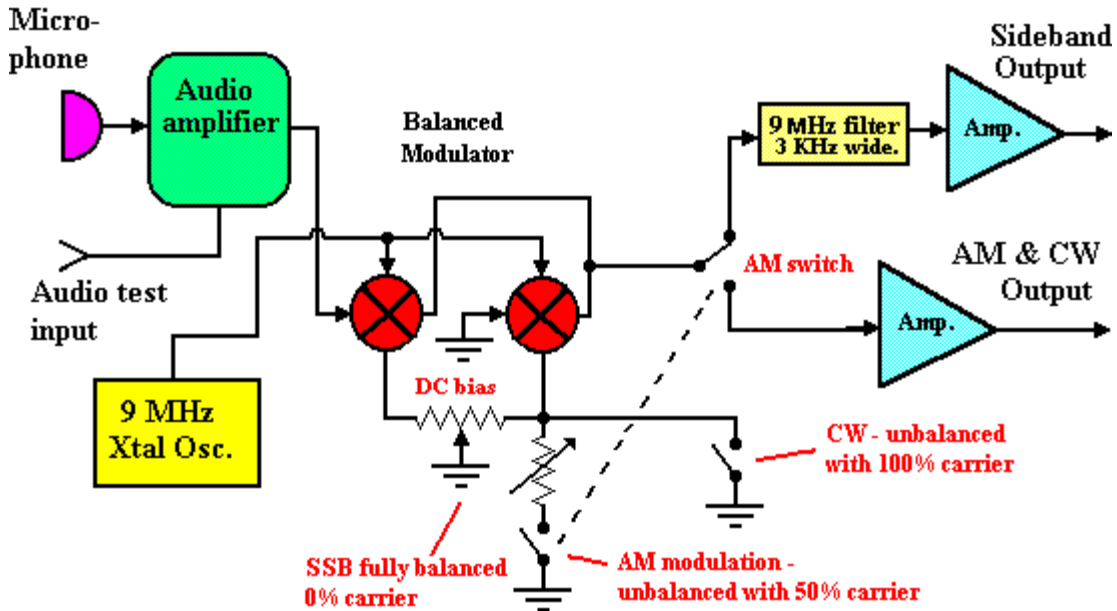


The above keyer was originally designed to turn the DC power to my final on and off with a telegraph key. My AM modulation scheme was to turn the MOSFETs half-on with a simple DC potentiometer, then modulate the gates with a 12 volt P-P audio signal. Because I was driving MOSFETs with a low (audio) frequency signal, hardly any power was needed. This simple scheme worked pretty well, but it was extremely finicky to adjust. It was easy to have too much bias or too little and too much modulation or too little. The difficulty is that the gate voltage versus drain current transfer characteristic is rather non-linear. With feedback and a more sophisticated drive circuit, I believe this method can be made to work well.

The SSB approach to AM

At this point in my R&D, I had not yet succeeded in building a practical SSB transmitter. So rather than invest more time on "obsolete modulation," I went back to work on SSB. I figured that, if I ever got the SSB working, it would be easy to downgrade my SSB generator to AM. This turned out to be true. I tried out several variations of converting SSB to AM. *The method that was simplest and worked the best was bypassing the SSB crystal filter with a switch and unbalancing the balanced modulator circuit.*

BALANCED MODULATOR



AM resembles CW in that a sinewave carrier is generated continuously. However, the same "unbalance" switch used as a SSB/ CW mode switch can't be used for AM. When modulation is applied, the instantaneous power must rise above and below the no-speech carrier level. Ideal AM modulation drives the carrier alternately between zero and 200% of the carrier level. Because there is a limit on the signal amplitude available, *the carrier must be set to 50% of the level used for CW*. This provides a modulation amplitude range from zero to 100% of the CW level. A separate AM mode, double-pole switch bypasses the SSB filter and unbalances the modulator to 50% of the maximum carrier. The AM switch is in series with an adjustable 5K ohm resistor that unbalances the modulator just enough to produce the 50% carrier.

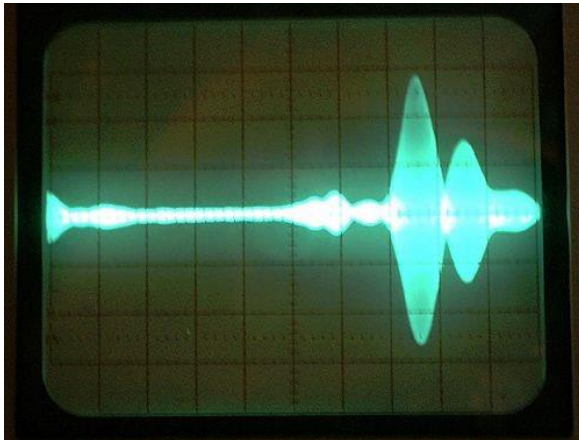
The audio gain pot and your voice level should be adjusted to produce voice peaks twice the carrier level. Compared to SSB, you'll find that AM modulation is quite HI-FI. While testing the generator and transmitter on an 80 meter dummy load, music retransmitted from a Walkman was quite acceptable. In contrast, when using SSB, speech sounds OK, but music is really terrible. The principle difference is that the sideband filter greatly attenuates frequencies below 300 Hz whereas AM preserves the low frequencies. Speech transmitted on SSB can sound like the person's normal voice, but music on SSB is dreadful. It's just as well. The last I heard, music on amateur radio bands is still illegal.

COMPRESSION BY ACCIDENT

Or, sometimes we get lucky

A modern single sideband generator processes the amplified audio from the microphone before the audio is fed into the balanced modulator. This "*compression*" process attempts to

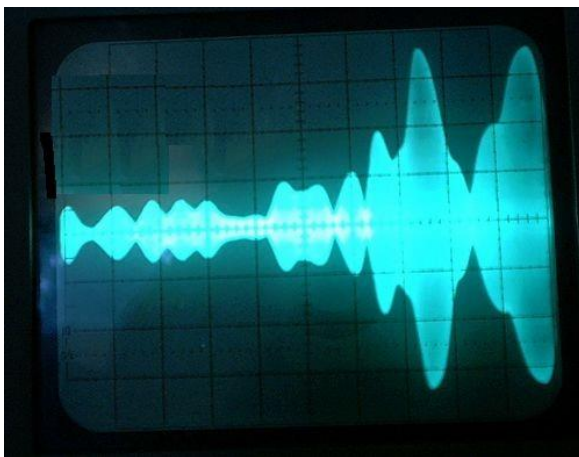
equalize the voice peaks so that as many voice elements as possible are transmitted with full Peak-Envelope-Power. Without this process, most of what you have to say will be transmitted with far less than the nominal peak power. When most of your sentences are reduced to QRP muttering, your intelligibility suffers.



In other words, without compression, the single sideband RF envelope of a spoken word is close to zero most of the time. It would look something like the waveform shown above. A compressor circuit attempts to leave the peaks alone while proportionally amplifying the subtle, low voltage waveform wiggles near the horizontal axis. I guess the latest transceivers use digital processing to accomplish this feat. However, in 1980 a compressor circuit usually performed the following tasks:

1. It amplified the whole audio waveform.
2. It clipped off the highest audio peaks.
- 3.

And finally, it filtered the clipped audio with a 300 Hz to 3KHz bandpass filter.



After compression, the same RF sideband waveform might look something like the above picture. The idea is that all the tiny waves near zero have been expanded. (These waveforms aren't actual before-and-after pix, but they illustrate the principle.) After transmission some modern receivers "re-expand" the waveform to try to restore the original waveform. This entire process is called *companding*. However, for me, building a homebrew SSB that worked at all seemed plenty difficult. Consequently I didn't worry about secondary issues like "companding."

A crystal filter does more than clip the unwanted sideband

In the beginning I was afraid my RF signal might be too wide. So, because it was relatively easy, I built a 3 KHz audio low pass filter. It turned out that I didn't need it. Once I had passed the 9.000 MHz RF double sideband signal through the crystal filter to cleave off the unwanted sideband, I found that the filter had also removed virtually everything above 3 KHz anyway. Also, when I adjusted the original sinewave frequency to get rid of every trace of the carrier, I found the filter had also clipped off the lower 300 Hz of the audio. It's remarkable how normal a voice can sound without the lower 300 Hz. Voices are quite lifelike. In any case the crystal filter accomplished the same frequency filtering that the ARRL Handbook specified for the audio compressor. Interesting!

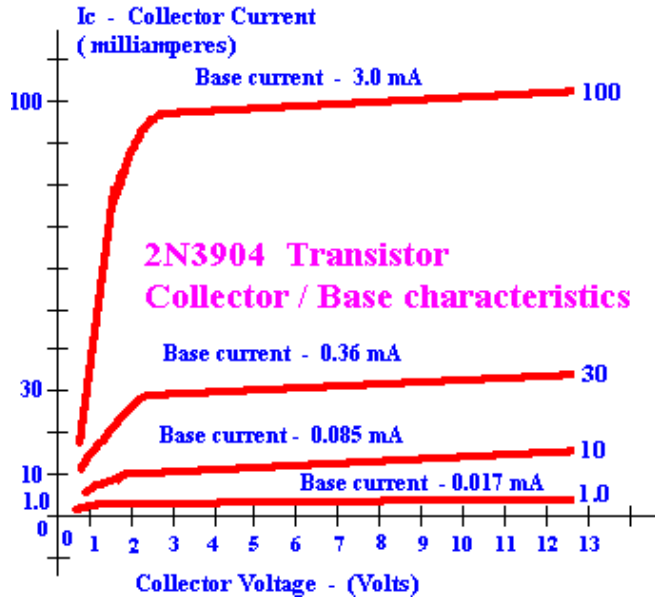
An SSB transmitter has several linear amps in series

After the SSB RF signal has been generated at a milliwatt level, the signal must be amplified and converted to the desired hamband. Including the mixer, this meant that my SSB signal had to pass through 5 stages of amplification to get to 100 watts peak. Each linear stage is forward biased (class A) so that even tiny signals will be amplified. Without this bias, all you would hear are the voice peaks. In other words, an unbiased amplifier (class C) cuts off all the little audio signals a compressor tries to accentuate. I knew that the linearity of all these stages in series couldn't possibly be "perfectly linear." But since voices sounded good, I stopped worrying about linearity.

Where has all the AM modulation gone?

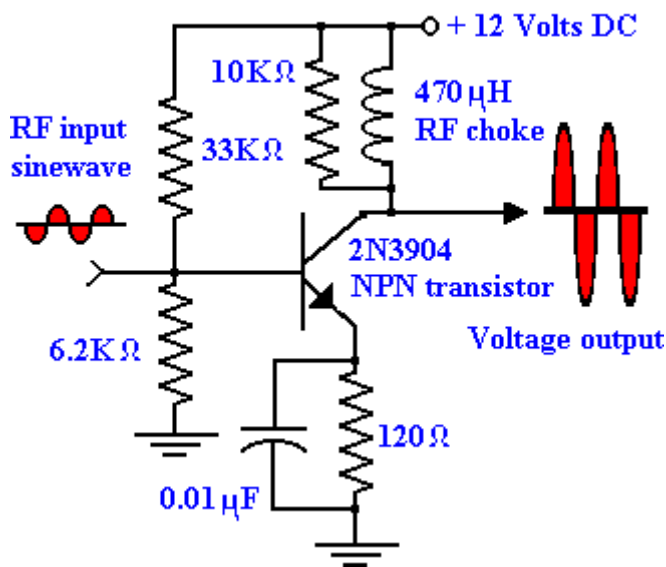
I didn't realize that my RF amplifiers were so non-linear until I added an Amplitude Modulator mode to my SSB generator. I listened to my little 9 MHz AM generator in my all-band shortwave receiver. It sounded great and looked like 100% modulation on the scope. Next I fed the signal from the 9 MHz AM generator into my 80 meter "linear" QRP module which put out about 3 watts on 80 meters. Yes, it worked, but the signal was nearly all carrier. Instead of 100% modulation, on 80 meters I only had about 5% modulation. Where did that huge carrier signal come from? What happened to my modulation?

Transistors aren't linear



"Linear" implies that big signals will be amplified just as much as the small ones. However, if the raw output of the transistor covers most of the collector operating range, then it happens that small signals are amplified more than big ones. I have two 2N3904 transistors in my chain of amplifiers, so the Base/ Collector current characteristics for this transistor are shown above. Notice that one milliampere of collector current requires 0.017 milliamperes base current. But to get 10 milliamperes of collector current takes 0.085 milliamperes. That's 5 times more base current to get 10 times more collector current. But if you want 100 milliamperes of collector current, you need 3.0 milliamperes of base current. That's an additional 35 times more base current. Sure looks non-linear to me. **BEHOLD, A NON-LINEAR COMPRESSOR!**

Transistor Amplifier



The "linear" amplifier above illustrates an accidental compressor circuit. The 33K resistor biases the transistor ON so that even tiny RF signals will be amplified. (By the way, the 10K resistor across the inductor keeps the amplifier from oscillating when there is no input signal.) The main reason for the 120 ohm resistor is to provide negative DC feedback to make the amplifier thermally stable. Without the emitter resistor, the amplifier works, but the transistor runs extremely hot. The emitter resistor also makes the amplifier more linear than the transistor characteristic would suggest because the feedback restricts the transistor to a narrower range of operation. However, 120 ohms feedback makes it a long way from linear. 470 ohms is much better, but still far from perfect.

Oh, well, why fight it? To fix my AM mode, I reduced the imbalance of the balanced modulator to just a few percent of voice peaks. This gives me roughly 50% carrier by the time it arrives at the final amplifier. And as for the SSB, it already works well. Apparently I had a first rate compression system all along and I didn't even know it. Imagine! A happy accident! They sure don't happen often.

A 40 METER PULSE WIDTH MODULATOR

Voice modulation with a CW transmitter

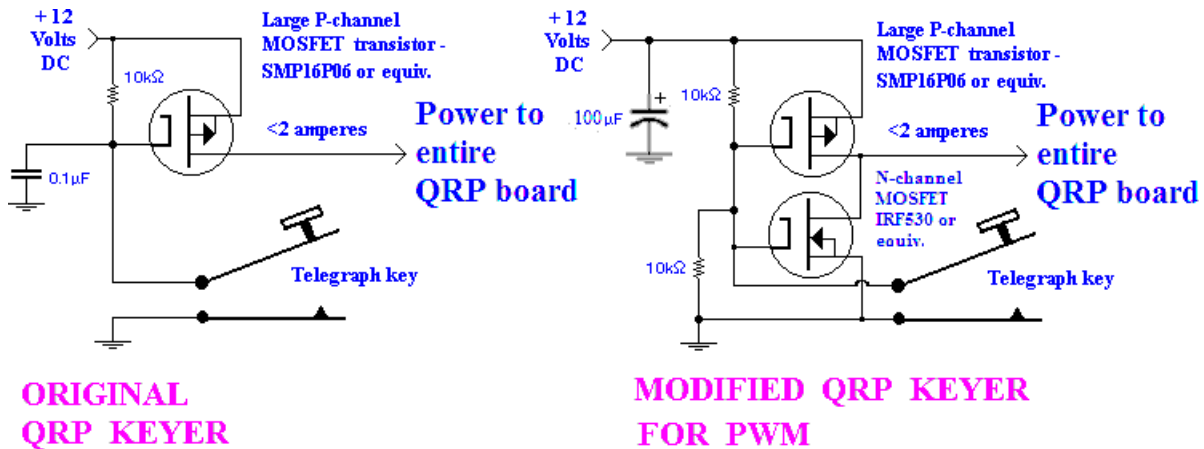
A CW transmitter keys the RF output fully on and off. Can you plug a voice modulator into the key jack of your CW transmitter and transmit good quality audio? Yes, you can! I had used pulse width modulation (PWM) for power supplies and motor control, but I had never seen an example applied to radio. I assumed PWM for RF had been done, but I thought it would be fun to pretend that I was inventing something new. I didn't cheat by looking up actual information. That would have spoiled the challenge!

Because a CW transmitter is always at max power, the signal is relatively immune to noise, much like FM modulation. Instead of coding the audio into changing amplitude or varying the transmitter frequency, *pulse width modulation codes the analog speech signal into varying widths of brief, constant frequency pulses*. Each pulse width represents a sample of the momentary amplitude of the complex audio sinewave. The downside of PWM is that the bandwidth is quite high, probably higher than equivalent commercial AM radio.

The sampling pulses are a fixed frequency, 7 KHz or higher. All that varies is the pulse width. The width of each pulse is proportional to the momentary height of the amplitude. If the sampling rate is fast enough, there will be enough samples to describe the waveform in adequate detail. It is like a picture drawn as a mosaic. The finer the mosaic pieces, the more accurate the reproduction. Long ago in "engine-school" we learned about Professor Nyquist's criterion: "To sample an analog signal, you must **sample at least twice the frequency** of the highest analog frequency you wish to code." In ancient times analog telephones were designed to communicate speech up to 3.3 KHz. Therefore, I needed to key my QRP at 7 KHz minimum for intelligible speech. (Nyquist was right. Lower sample rates produced inferior sound and I could hear the whistle of the sample pulse rate.)

I began the project by using my ancient Heathkit audio frequency square wave generator to drive the key input of my 5 watt 40 meter QRP. I key all my homebrew transistorized

transmitters, even the 100 watt ones, with P-channel power MOSFET transistors that switch the power supply on and off. The circuit I use makes clean CW with no clicks or chirp. Unfortunately, this circuit turns off too slowly to handle 7 KHz "dots." I sped up the turn-off with a complementary N-channel MOSFET as shown below. The faster turn-off produces key clicks when used with ordinary Morse, but isn't too obnoxious. The frequency response is now over 10 KHz. A big electrolytic capacitor on the +12 volt line prevents overshoot on the square wave pulses. 10K ohm resistors prevent oscillation immediately after switching.

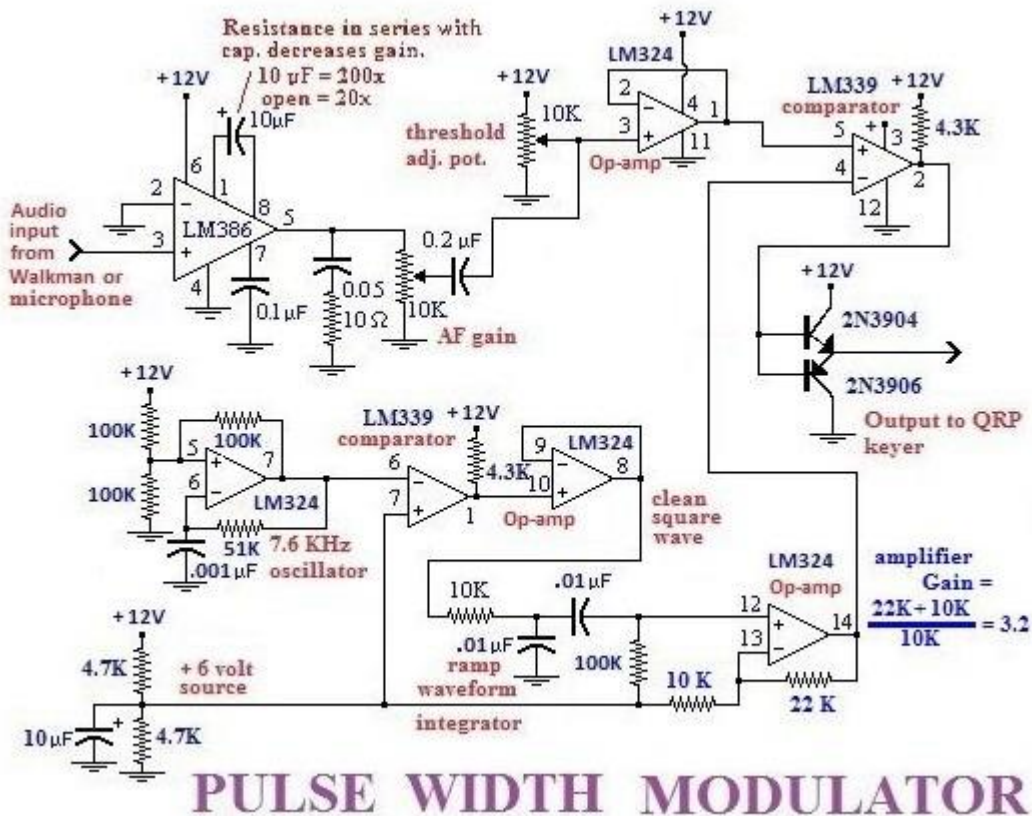


Zero crossing only samples the major frequencies

My first audio sampling experiment was building a zero-crossing pulse-width generator to drive the QRP key input. Whenever the audio sinewave crosses zero volts, a keying pulse begins or ends. This simple system transmits pure tones beautifully. When I put in sinewave audio tones from an audio generator or I whistled a tune into a microphone, the QRP transmitter sent a clear representation to the AM receiver tuned to 40 meters. The audio frequency coming from the receiver speaker is simply the frequency of pulses. Unfortunately, when driven with speech, the result is unintelligible harsh, roaring noises.

Although zero crossing modulators sample the *FREQUENCY*, they capture no information about the amplitude. Speech and music contain multiple, simultaneous frequencies as well as the momentary loudness of the sound. In an audio waveform, all this information is contained in the complicated audio sinewave amplitude, moment by moment.

A constant frequency 7 KHz audio amplitude sampler



The schematic above is my homebrew PWM audio keyer. An oscillator generates a square wave at 7.6 KHz. A comparator (upper right) squares the pulses which drive a capacitive integrator to make ramp-shaped pulses. The small ramp signal is amplified 3.2 times to make the audio threshold and amplitude adjustment easier. The audio signal is compared, moment by moment, with the ramp-shaped pulses. When the ramp amplitude and audio amplitude match, the width of each 7 KHz pulse is terminated. By adjusting the audio threshold, the CW transmitter can be keyed either ON most of the time or OFF most of the time. The best sound seems to occur when the ON to OFF ratio ranges between 10% to 30%.

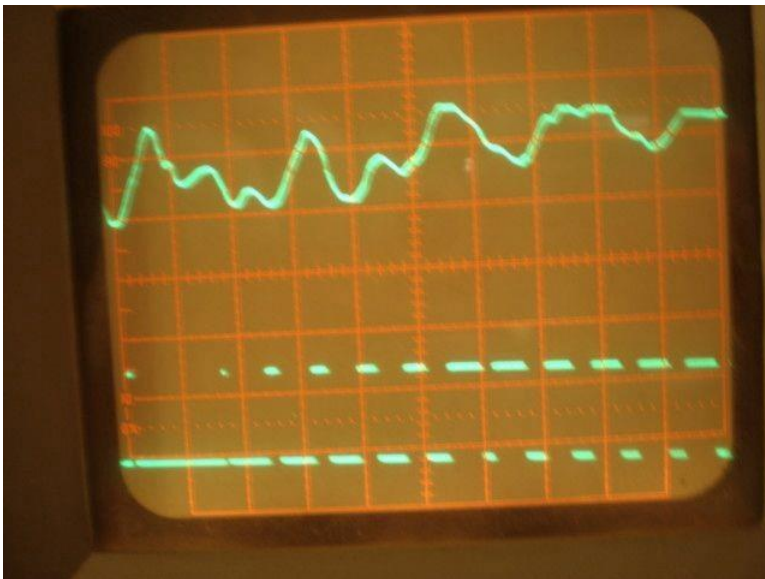
What's a comparator?

I have never mentioned comparators before. They are a special kind of operational amplifier that cannot be stabilized by feedback to produce an output at any level except full ON or full OFF. In other words, the output can never be half ON or other analog value. Op-amps can also be used as comparators, but if the desired output is a square wave, then op-amps are usually too slow for most applications. An op-amp output will always rise and fall more slowly. Like the LM324, the LM339 is a low-scale, linear integrated circuit from the 1970s. However, they are dirt cheap and quite reliable.



The first working prototype

Shown clockwise: PWM breadboard prototype, Walkman audio source, 12 volt DC supply, 40 meter QRP with dummy load.



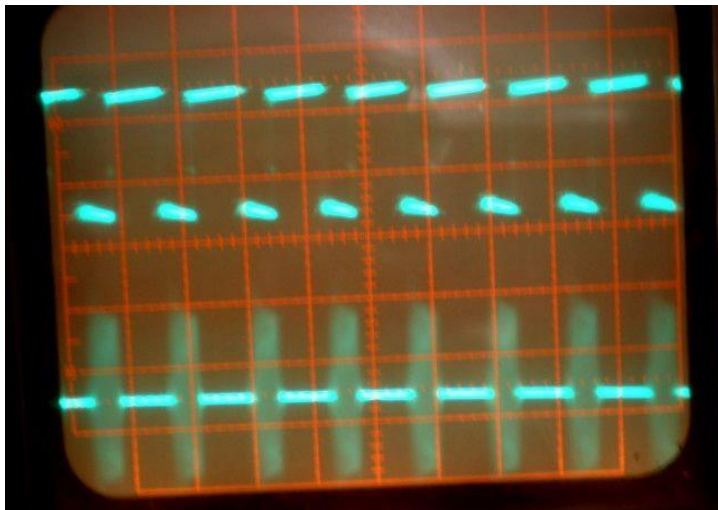
The upper scope trace shows a typical wavy audio signal. The lower trace shows the 7 KHz PWM pulses sampling the audio that immediately followed the upper scan. (My scope can't catch 2 traces at once.)

No special PWM radio receiver is needed.

PWM RF signals are a kind of AM modulation. Wide RF pulses are louder than narrow pulses, so the amplitude variations of speech are faithfully reproduced. The class D audio amplifier in your cellphone works like this. Wide, high current, low voltage pulses are louder than narrow ones. To my amazement, transmitted speech is not only intelligible, music can be quite pleasant. The receiver must be tuned to the exact RF frequency to hear the low bass notes. There is considerable noise outside the +/- 8 KHz bandwidth. When I switch in one or two

crystal filters on the receiver, the narrower bandwidth removes the hash and leaves a quality audio signal. The noise transmitted outside +/- 8 KHz is unnecessary for good reception. Like the crystal filter on the receiver, possibly the transmitter output can be filtered to reduce the bandwidth. I ran my QRP output through a resonant L-C antenna coupler, but this extra filtering made no noticeable improvement. I would have to reduce the bandwidth before I could ever try it on the air.

Full pulse width variation, such as the lower scope scan shown above, produces a much higher audio level for a given RF signal strength. Unfortunately, the switching noise is also higher along with the bandwidth. An odd characteristic of my PWM is that the audio quality is inversely proportional to the degree of pulse width variation. When the audio is best, the width of the RF pulses hardly varies. On the scope, the 7 KHz pulses just shimmy slightly.



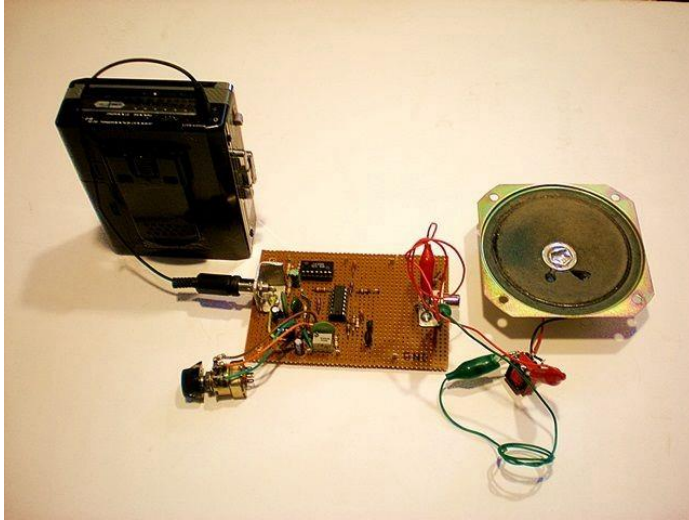
The lower trace is the 40 meter RF voltage across a dummy antenna load. Believe it or not, those equal-looking, blurry, 40 meter HF pulse bursts are transmitting fairly hi-fidelity music. The upper trace shows the binary pulse train that keys them. While transmitting modulation, the bursts move constantly. That's why the waveforms are blurry. When the audio stops, the pulse train stops wiggling.

FM noise advantage with less average transmitter power

Unlike ordinary AM, the pulses are always at full power and well above the noise. It is possible to over-amplify the pulses in the receiver IF, just like an FM discriminator. That way they always saturate and *there will be essentially no atmospheric noise*. The edge of the pulses dithers back and forth creating a change in frequency, causing the wide bandwidth. When I switched in my FM discriminator detector described in Chapter 13B, the audio signal was weaker but similar quality. Detection with a product detector and BFO produces distorted audio, even with the receiver tuned to zero beat.

Testing the PWM with a loudspeaker

After the press-board prototype was working, I built a permanent version of the PWM generator shown below. I used a loudspeaker to demonstrate the generator function directly. The power supply is not shown. In order to hear the sound, the high impedance, square pulses need to pass through a speaker transformer to match the 8 ohm speaker.



The loudspeaker allows us to adjust the input audio, comparator level and the audio output amplifier before modulating a transmitter. The transmitted signal can't be hi-fidelity if the sound isn't equally high quality in this test. Audio quality is best with narrow pulses and low audio input amplitude.

PWM can achieve low average RF transmitter power

By adjusting the audio voltage level threshold, the bursts can be made extremely narrow with reasonable speech audio quality. The advantage of this is that the bursts are short and the transmitter is turned OFF most of the time. That is, the transmitter has a low duty cycle. In contrast, FM transmitters operate at full power whether you are talking or not. Similarly, AM transmitters operate at 50% of peak power average. Although my scope measurement is crude, it appears that the average transmitter power can be as little as 10% of average FM power or 20% of average AM power. In terms of average transmitter power, PWM can be at least as frugal as SSB.

In conclusion, I had the chance to "invent" something cool. If I were back in the year 1935, my PWM would have been competitive with the early FM experimenters. Of course if this modulator were built with the vacuum tubes available in 1935, it would be huge and expensive. On the other hand, surely I can keep the bandwidth under the +/-75 KHz used by modern commercial FM.

Later ... I was telling Ed, WØHB, at a ham club meeting about my PWM. He suggested that there is another way to narrow the wide bandwidth. If the sampling frequency is very high, like 75 KHz, the AM reception will be the same, but the annoying, hissing sidebands will be 75 KHz offset from the center frequency instead of 7 KHz. This wider separation allows antenna tuners to suppress the off-frequency noise more easily. The catch is that the final amplifier must be keyed precisely at 75 kHz. This would be very different from a simple system that can be plugged into the CW key jack.

Narrow Band Amplitude Modulation

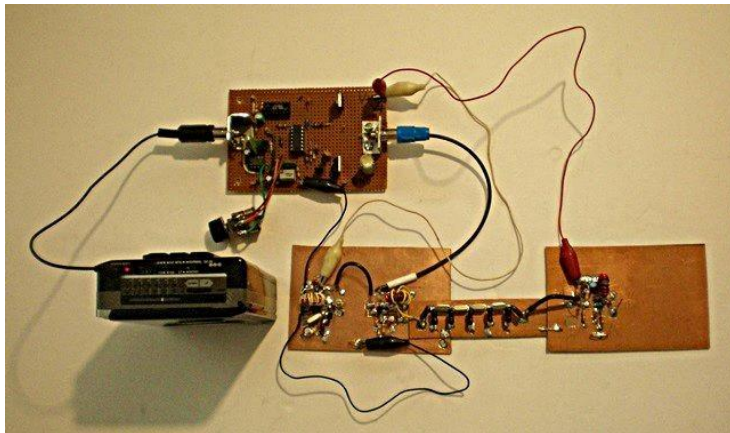
After thinking about the PWM bandwidth problem, it occurred to me that a single-

sideband filter like the ones used in Chapter 15 can eliminate the overly wide bandwidth. A sideband filter can also generate a voice modulation mode I have never heard mentioned, ***Narrow Band AM***.

Narrow Band AM for video

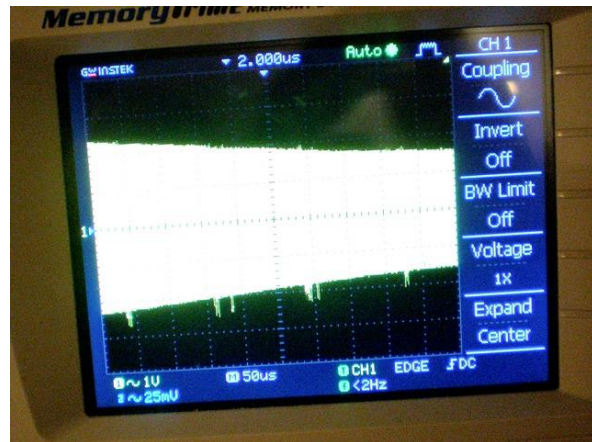
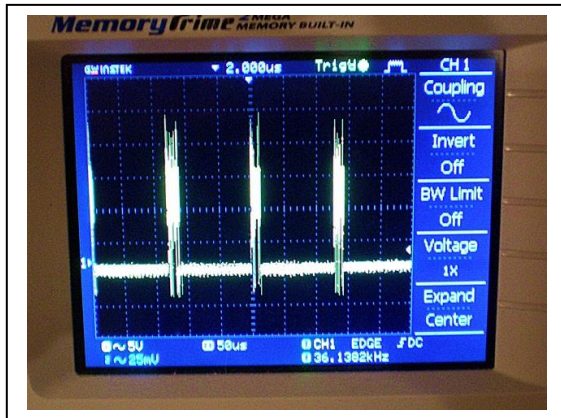
Actually "**NBAM**" was used for many years for analog TV transmissions. The TV audio was FM, but the video transmission was narrow band AM. The old American standard TV channels were 6 MHz wide. However the AM modulated TV signal was 12 MHz wide. If an ordinary AM modulated video signal were transmitted on channel 3, it would also be visible on channel 4. Oops. I discovered this accidentally when I was in high school playing with TV transmission as described in the next article.

The commercial TV stations cut off one of the sidebands using resonant cavity filters. The signal was passed through an empty metal box. The box dimensions are designed to resonate, "echo," at a specific frequency. Any frequency that doesn't resonate with the cavity can't pass through. Think of it as a flute for radio waves. Only the desired "note" passes through the box. The output from the box included the carrier wave, one sideband and a "vestigial" sideband a few KHz wide. The TV receiver can reproduce a first rate picture with one sideband and the constant amplitude carrier wave.



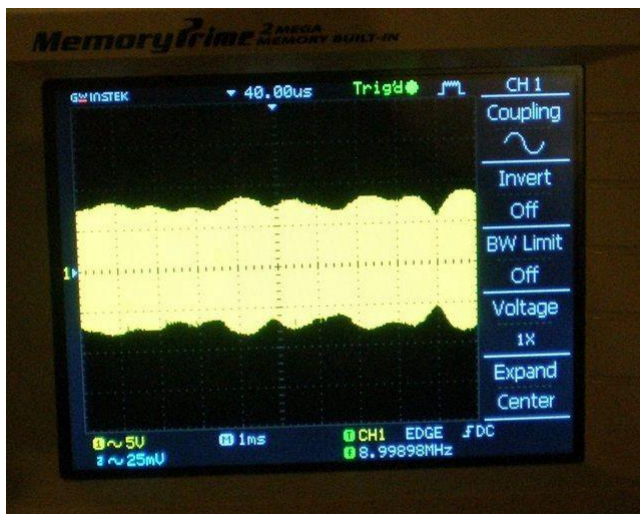
NBAM prototype boards

The Walkman radio on the left provides the speech signal. The PWM board converts the analog audio into an 8 KHz PWM pulse train. The output from the PWM board keys a 9 MHz RF amplifier ON and OFF. The 9 MHz crystal oscillator runs continuously and is on the left of the same amplifier board. The 9 MHz oscillator did not turn ON fast enough to key it along with the amplifier. The variable width 9 MHz bursts are fed into the narrow PC board in the center which is a four crystal sideband filter. This filter is the same as the 9 MHz filter presented in Chapter 15. The output from the crystal filter is then amplified by a second, non-resonant 9 MHz RF amplifier. In a real transmitter, the output from the last amplifier would be passed on to a mixer for conversion to the desired ham band.



Look ma! No pulses!

The output from the unfiltered 9 MHz PWM is shown above on the left. (Yes, my old analog scope display of RF pulses looked quite different and I don't know why. Much of the difference is probably the analog versus digital oscilloscopes.) The picture on the right is looking at the output of the second 9 MHz amplifier. The continuous waveform has no pulses and is the result of the sideband filter. *The crystal filter smoothes out the pulses and produces a continuous waveform.* The pulses are gone except for some slight artifacts propagated through the power supply. The vertical scale in the output picture was dropped from 5V to 1V, but otherwise, the scope settings are the same. Below is the same output with a slower sweep to show the audio modulation.



An AM signal with a large carrier wave

At this scan rate we see a typical AM waveform. The important difference is that the bandwidth is no longer tens of KHz. As heard with my receiver, the signal bandwidth is only about **4 or 5 KHz wide**. If the crystal filter and output were shielded and isolated, the bandwidth would probably be even sharper. The exact bandwidth number depends on how many dB decrease one wishes to define as the edges of a bandwidth.

9 MHz filter output with oscillator frequency shifted for minimum carrier



If I move the 9 MHz oscillator frequency up or down, so that it is 1.5 KHz away from crystal filter center frequency, it now looks like SSB. That is, the signal lobes are centered on zero output instead of a higher average carrier level. Unfortunately, I can't tune in the signal as though it were SSB without serious distortion, so I can't claim that I've generated single sideband. ***I did observe that I can have good, clear AM audio with a very small carrier.***

By now it was obvious that I could have achieved the same results by starting with an ordinary analog AM signal. My R&D has gone in circles. Oh, well. It was fun!

So what's it good for?

Because the bandwidth is so narrow, there's no reason to be embarrassed to use this on the modern phone bands. Notice that double-sideband signals (DSB) are about 7 KHz wide and DSB is considered acceptable. Packaged in a module like the SSB generator in my SSB transmitter, there's no reason I couldn't replace it with an NBAM module and use it on any HF phone band. OK, maybe not on 60 meters - (USB SSB only) - but I was unlikely to do that anyway. No matter what receiver you're using, ***AM is always easy to tune in.***
