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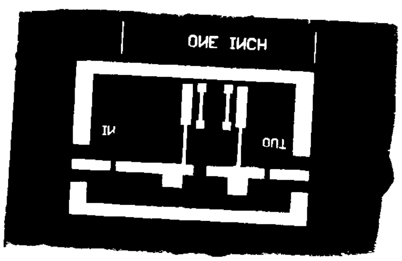


Amateur Radio Equipment Development: A Historical Perspective

By Joel R. Hallas, W1ZR

The year 2014 marks the 100th year of the American Radio Relay League (ARRL, the national association for Amateur Radio). While there was Amateur Radio before there was an ARRL, it is clear that most of the advancement of radio science occurred during the period. This 100 year period saw radio move from a barely functional, unreliable set of disparate communications operations to a largely integrated system comprising major elements of the lives of everyone.

Radio amateurs have not only benefited from all the technological advances over the period, but also many contributed directly to the development and success of important advances in technology. This section highlights some of the major developmental steps of Amateur Radio that led us to the technology that we enjoy today.

First There Was Spark

Transmitters have what would appear to be a rather straightforward job — generate an ac signal and modify it in some way to carry information. In the early days, it was found that an electrical spark created in a circuit that was connected to an antenna could be received some distance away by various devices connected to another antenna. It didn't take folks very long to figure out that if the sparks were sent in patterns corresponding to Morse or other telegraphy codes, information could be sent over long distances without connecting wires. Wireless communication was born!

Figure 1 shows an advanced amateur spark station from 1907 — functional and on display at the ARRL Headquarters Museum of Amateur Radio (but not hooked to an antenna). On the lower left is the receiving side, a crystal detector with "cat's whisker," variable tuning coils and a tubular variable tuning capacitor. The top deck holds the transmitter, starting with the automotive type spark coil on the right. In the middle is multilayer flat mica capacitor with taps at various points, along with the tuning inductor. Together they defined the operating frequency. On the left of the top shelf is the actual spark gap. The lower right contains a land line sounder and switches to allow traffic to be relayed to offices connected locally.

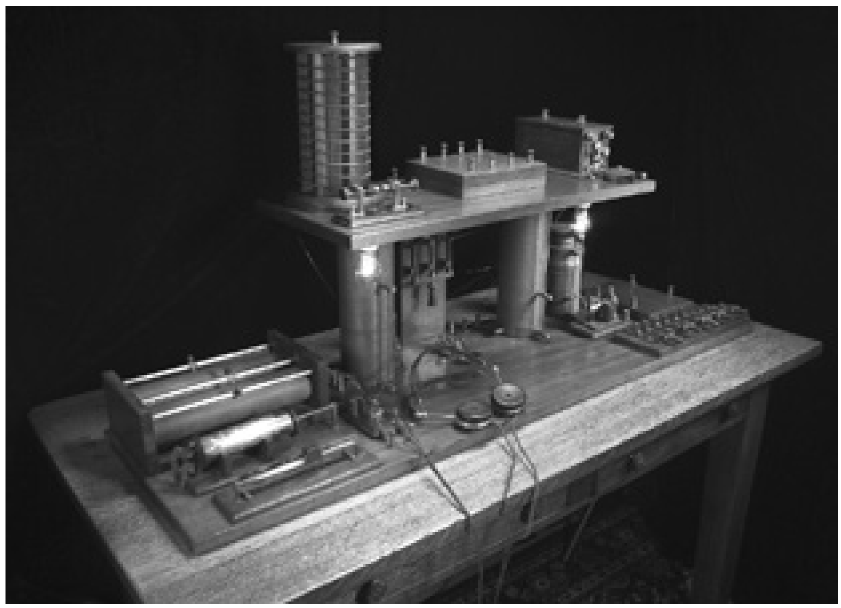


Figure 1 — An advanced amateur spark station from 1907 — functional and on display at the ARRL Headquarters Museum of Amateur Radio. [Joel Hallas, W1ZR, photo]

Technology quickly advanced through a number of variations of spark and arc transmitters and a wide variety of ingenious receiving devices. Eventually wireless telegraphy became a serious business involved with monitoring the safety of ships at sea in ways that hadn't been possible before. See **Figure 2** for a view of an early shipboard wireless station. The commercial systems of the day were more electromechanical than what we would consider electronic today.

The Receiving Detector

The primary function of a receiver takes place in a *detector*, the circuit that extracts the information from the modulated RF signal that arrives from the antenna. As shown in **Figure 3**, the simplest receiver consists of just a detector. The earliest receivers were constructed in just that way, starting with some unusual electromechanical marvels from Marconi's days. The crystal detector, in the form of the crystal set, was the first radio experienced by youngsters from the 1920s to the 1980s and was the mainstay of many commercial users until vacuum tubes became available. While even simpler configurations are possible, the circuit of **Figure 4** is typical. In an early set, a galena crystal and a cat's whisker would be employed instead of the 1N34 germanium diode.

The key performance parameters of this receiver are easy to describe. Its *sensitivity* is the signal level required at the antenna input to create an audible signal as a consequence of the diode acting as a square-law detector. The *selectivity* is determined by the loaded Q of the single-tuned resonant circuit, while the *dynamic range* extends from the sensitivity level to the point that the headphone diaphragms hit their limits or the diode opens due to high current.

While this receiver looks like it would be rather limited in performance (and it is), it does work and was even the basis for most microwave radar receivers from WWII and for some years thereafter. That was the origin of the venerable 1N34 diode. The *diode detector* is still used for full carrier amplitude modulated signals in more complex modern receivers.

Early Amateur Transmitters

Amateur transmitters evolved from those based on automotive type spark coils with

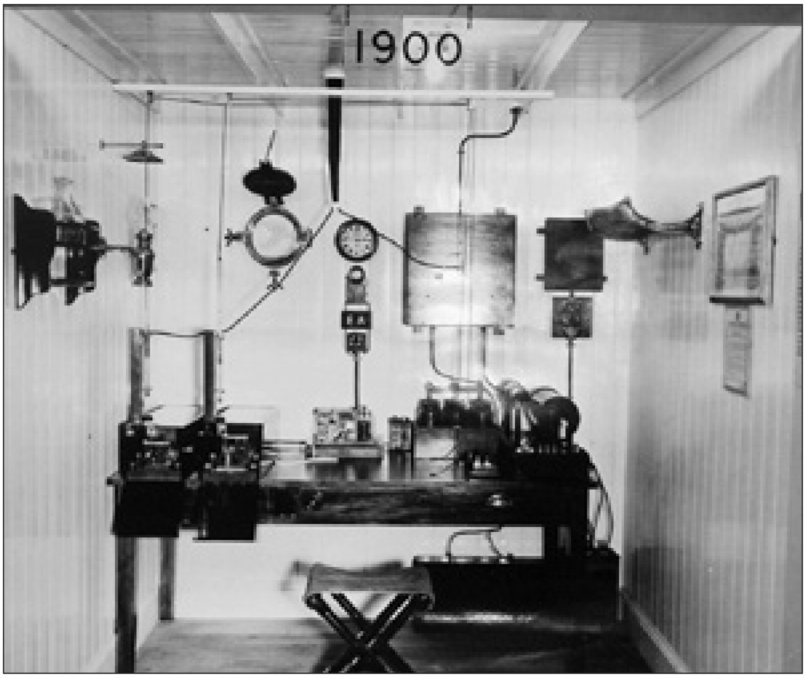


Figure 2 — Marconi shipboard radiotelegraph station from 1900. (Photo courtesy of Marconi Company, PLC)

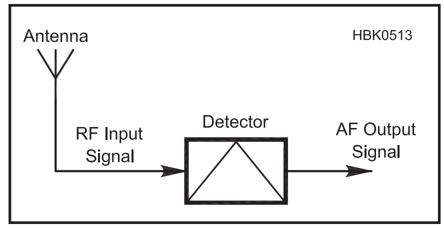


Figure 3 — The minimal receiver is just a detector.

electromagnetic interrupters (think buzzer) to high voltage transformers powered by ac mains and using motor driven *rotary spark gaps*. The rotary gaps produced sparks interrupted at a multiple of the motor speed, resulting in a form of modulation that could be heard using the popular crystal detectors of the day. The key was typically in the ac mains circuit, not a system that would pass a safety inspection today.

A high power amateur station had a transformer that could deliver 1 kW, although it's not clear how much of that power was radiated as light and noise by the gap. While the commercial — especially maritime — radio services moved to more sophisticated and complex arc transmitters, along with a number of different detector technologies, the amateur operators tended to stay with spark transmitting and crystal receiving technology until vacuum tubes became readily available.

Some commercial fixed shore stations used electromechanical *alternators* as the generators of RF for transmission. A very basic alternator might consist of a coil with a core of soft iron (pure molecular iron that will magnetize and demagnetize easily) rotating past two poles of a magnet inside a round iron mounting as in **Figure 5**. As one end of the rotating iron core coil passes the north magnetic pole, a magnetic field is built up in it. As a result of the magnetic field increasing then decreasing in the rotor coil's core, one half of an ac cycle is induced in its coil. As it continues to rotate and passes the south pole, an opposite half cycle of ac is induced in it. To produce 60 Hz ac, the coil would have to rotate 60 times per second, or 3600 revolutions per minute — a very fast rotation. The two ends of the rotating coil are connected to two slip rings on the shaft that rotates the coil. There are two brushes of fixed carbon or other material that make a constant contact with the rotating slip rings. Any ac voltage that is generated is available at these two brushes. By using multiple rings and brushes, multiple cycles of ac will be produced with each rotation.

Better systems were used to generate ac at frequencies in the lower RF range. Ernst Alexanderson, a prolific Swedish/American electrical engineer and inventor, and Canadian inventor Reginald Fessenden first produced a lower power 60 kHz RF alternator in 1906, with which they could transmit telegraphy as well as voice and music. Later the higher powered Alexanderson alternators used frequencies closer to 20 kHz. Other popular alternators of the period were the German Joly-Arco and the Goldschmidt, which was somewhat similar to the Alexanderson.[1]

The final form of the Alexanderson alternator uses a rotary toothed disc, called an *inductor* (not to be confused with a regular electronic coil). This type of inductor is a large round, flat, soft-iron disk with teeth or other regularly spaced features along the edge as shown in **Figure 6**. The many teeth of the inductor are rotated rapidly by a constant speed ac motor between the north (N) and south (S) poles of double-wound field pole electromagnets mounted one tooth-width apart all around the inside of the machine. As an iron tooth passes between the N and S ends of the dc-excited electromagnet field poles, it provides a better path for the N to S magnetic lines of force. As a result the magnetism in the field pole builds up, then collapses as the tooth moves on.

As the next inductor tooth passes the field pole, it develops another magnetic field build up and collapse. These varying magnetic fields induce ac cycles in all of the secondary RF output field-pole coils. Since there are as many field poles around the inside of the machine as teeth on the inductor, ac voltages are developed in the field pole ac pick-up coils all at the same time and are all inductively coupled to a coil/antenna/ground circuit that radiates the RF waves. The speed of the motor rotating the inductor and the number of teeth determines the output frequency.[2,3]

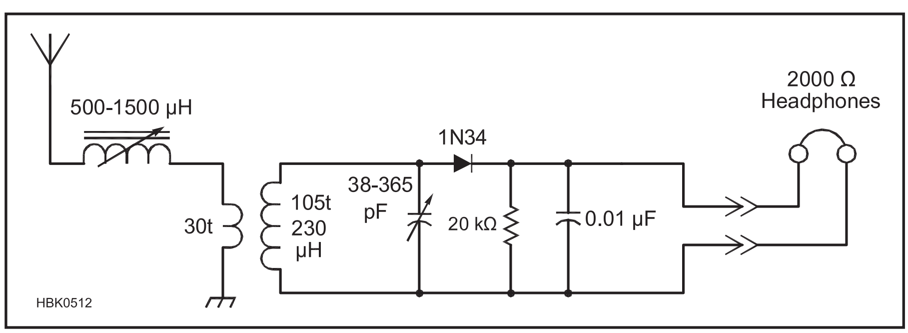


Figure 4 — Circuit of a typical "modern" crystal set with a semiconductor diode as the crystal detector.

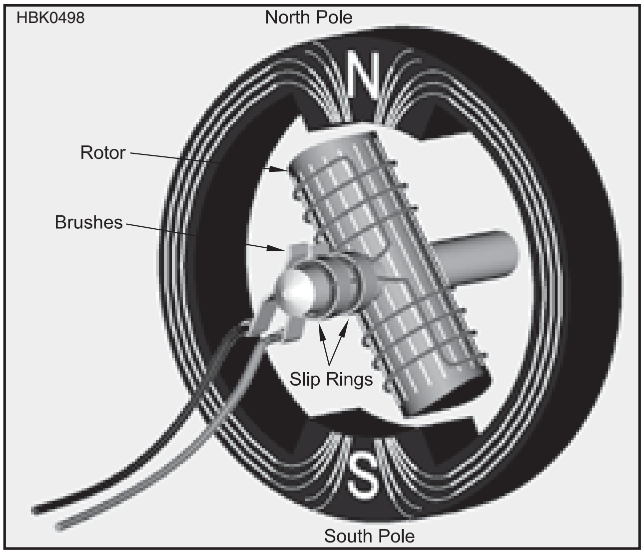


Figure 5 — Basic alternator consisting of a soft-iron cored coil rotating past two poles of a magnet.

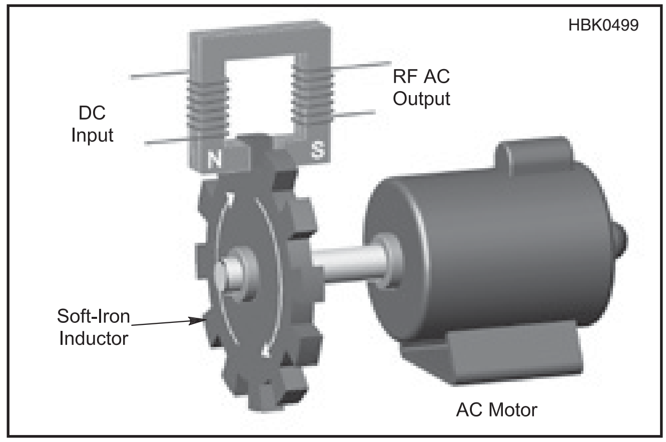


Figure 6 — High frequency alternator using an inductor disc.

Then Came Vacuum Tubes

The Diode Detector

The vacuum tube was a development based on Edison's light bulb and invented in 1904 by John Fleming, a professor of physics at the University of London. Fleming was a consultant to the Marconi Company who had worked on the development of the transmitter for their transatlantic trials. The *Fleming valve* was a diode and, as such, it could work as a receiving detector in place of the mechanically difficult galena crystal and cat's whisker arrangement. This was a great improvement, especially for shipboard stations, since vibration — not to mention recoil from cannon — would result in movement of a cat's whisker and loss of reception.

Early vacuum tubes were quite expensive and out of the reach of most amateurs. With most amateurs being able to avoid the heavy vibration and artillery issues of shipboard operation, the inexpensive but finicky crystal detector continued to serve until tubes became obtainable more reasonably.

The Triode Vacuum Tube Changes Everything

The development of the Audion by Lee DeForest in 1906 moved electronics and radio a large step forward. The Audion was the first three element (*triode*) vacuum tube. It added a grid located between the filament or cathode and the anode or plate of the earlier diode. A small change in grid voltage resulted in a larger change in plate current. This allowed the tube to amplify rather than just rectify. Weak signals applied to the grid would become stronger signals in the plate circuit.

Amplification could do a number of things that hadn't been possible before. Weak signals from a diode detector could be made strong enough to fill a room using loudspeakers. Perhaps more importantly, an amplifier with positive feedback could be made to oscillate and thus generate a sinusoidal signal. If this signal were generated at radio frequencies, it could be coupled to an antenna and keyed in Morse code, making for a very efficient transmitter.

The signal transmitted by a vacuum tube oscillator was a sustained and nearly pure sinusoidal signal referred to as continuous wave or CW. This was quite unlike the wideband chopped and damped waves of the spark transmitter. This was a mixed blessing. On the one hand, the CW signal occupied a much narrower band of frequencies, allowing other spectrum users to operate on frequencies much closer together without interference. On the other hand, the buzzing or chopped sound of the spark signal in a crystal set was replaced by a series of thumps on key-up and key-down when the set received a CW signal.

The Regenerative Receiver

Fortunately, the solution to this problem was provided by the same technology in the form of the oscillating, or *regenerative*, detector. Invented by Edwin Armstrong in 1913, the regenerative, or *autodyne*, detector became the mainstay receiver technology of amateur and commercial operators for many years.[4] The regenerative detector used an amplifying stage that was made to oscillate at a frequency slightly offset from the received CW signal. The mixing of the two signals in the tube resulted in a beat note that was clearly audible in the headphones in the plate circuit. In addition, the oscillation magnified the sharpness of the input tuned circuit, resulting in high gain and the ability to separate signals. This approach is in many ways similar to what is called a *direct-conversion* receiver by modern hams.

The regenerative receiver had a number of advantages, including simplicity and low cost. The early units used a single Audion and provided significant performance with few parts. There were some problems inherent to the design, however. A major one was that the oscillator was coupled to the antenna, making it serve as a low power transmitter while it was receiving. During World War I, allied transport ships suffered at the hands of U-boats that were able to track them by listening for their oscillating detectors. Working with limited Depression-era resources after the war, clever amateurs would actually use this "problem" as a benefit by keying the receiver to make it serve as a short range transmitter.

Amplifiers

The next element added to the detector was the *amplifier*. The first amplifiers provided bandwidth capable of amplifying only audio frequency signals and thus were inserted following the detector. In that position, they offered no improvement in sensitivity, which was still limited by the diode; however, they appeared to improve sensitivity because the amplified audio output of the receiver was now louder. It was now possible to have more than one person listen to a received signal

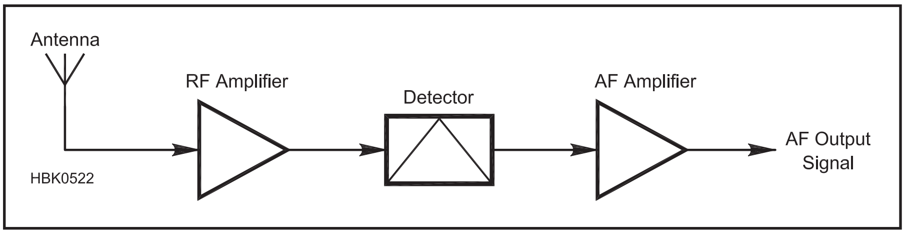


Figure 7 — Detector with RF and AF amplifiers to improve sensitivity, selectivity and power output.

through the use of a loudspeaker.

As shown in **Figure 7**, it wasn't long before vacuum tube and circuit design technology improved to the point that amplifiers could be used at RF as well as AF. The addition of RF amplifiers provided two performance improvements. First, the level of signals into the detector could be increased, providing an improvement in sensitivity. Second, if each amplifier were coupled using tuned resonant circuits, as was the usual practice, the selectivity could be improved significantly.

A receiver with one or more tuned RF amplifier stages was called a *tuned radio frequency*, or *TRF*, receiver. Some had as many as three or four RF stages, initially with separate controls, and later with up to a five-gang variable capacitor for station selection. Needless to say, a certain amount of skill was needed to adjust each stage so it would properly *track* as it was tuned over the frequency range.

There were other challenges with the TRF. With the high gain of the multiple stages all tuned to the same frequency, it was not trivial to avoid oscillation. Another concern was the selectivity provided over the tuning range. If the selectivity of the five tuned circuits could provide a bandwidth of 2% of the tuned frequency, that would result in 30 kHz at the top of the broadcast band (around 1500 kHz) but only 10 kHz at the bottom (500 kHz). Top-end receivers of the day used various methods to maintain similar bandwidth across the range automatically.

Even though the *superheterodyne* receiver became available by the 1930s, the regenerative receiver remained popular with hams through the Great Depression until World War II. The solution to the problem of transmitted signals from the receiver oscillator, described previously, was to insert an RF amplifier stage between the antenna circuit and the detector. This also improved the sensitivity somewhat, and it made the tuning more stable if the antenna moved in the wind. An audio amplifier stage was often used following the detector to permit operation with a loudspeaker. This three tube configuration was manufactured by the National Radio Company from 1931 to 1939 as the SW-3, and was one of the most popular medium and shortwave communications receivers of the era.

Early Vacuum Tube Transmitters

Vacuum tube transmitters developed in parallel with the receiving sets. Initially limited by the cost and availability of tubes that were allocated to the military, receiving type tubes began to become available after World War I. Enterprising amateurs "borrowed" tubes from the family radio and used them as single tube oscillating transmitters at first. Some receiving type audio power amplifier tubes were pressed into service as RF power amplifiers.

While transmitters are composed of many of the same named blocks as those used in receivers, it's important to keep in mind that they may not be the same size. An RF amplifier in a receiver may deal with amplifying picowatts while one in a transmitter may output up to megawatts. While the circuits may even look similar, the size of the components, especially cooling systems and power supplies, may differ significantly in scale. Still, many of the same principles apply.

Continuous Wave Telegraphy

By the early 1920s, large vacuum tubes were developed that could be used in high powered transmitters. Most of the big alternators were replaced by less massive high power tube transmitters. Amateur transmitters also quickly moved from spark to vacuum tube sets. The vacuum tube oscillators generated a single-frequency RF signal directly and continuously, creating the *continuous wave* or CW transmitter.

Although CW transmitters generally operated with less nominal power output than the spark transmitters of the day, they actually delivered more power to the antenna. Signals were of a much narrower bandwidth, allowing more usable channels once receiver capabilities caught up. The early tube transmitters were generally single-stage self-excited oscillators coupled directly to an antenna as shown in **Figures 8** and **9**.

While they were a vast improvement over early technology, such transmitters had their limitations. Note that the antenna was connected directly to the tuned circuit that controlled the frequency. This meant that every time the antenna moved in the wind, the resultant change in loading shifted the transmit frequency. If the cat walked by and was lucky enough not to be electrocuted by the exposed high voltage wiring, the frequency moved even on a calm day. The frequency would often shift with each key closure as the power supply voltage sagged with the additional current drain, resulting in signals with characteristic chirps and whoops.

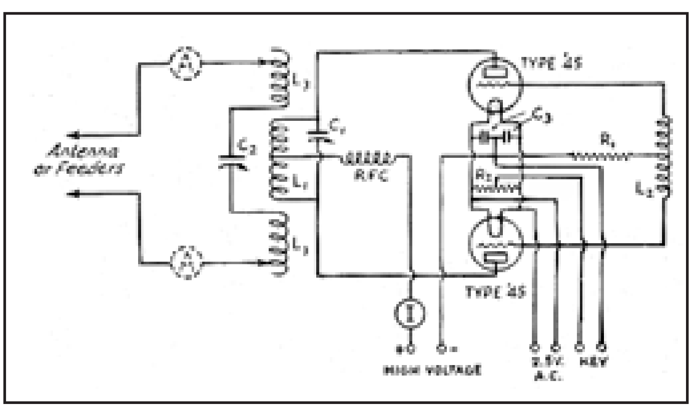


Figure 8 — Transmitter schematic as shown in an early *QST* magazine.

Figure 9 — Photo of a transmitter built from the schematic in Figure 8.

Master Oscillator-Power Amplifier Transmitters

All of these issues were at least partly addressed by adding a power amplifier between the oscillator and the antenna. Morse code transmission was accomplished by using the key to turn the power amplifier stage on and off. If properly designed and adjusted, and powered by a solid power supply, the oscillator could be left on between the transmitted dots and dashes. This stabilized the transmitted signal's frequency and the antenna was well-isolated from the frequency determining components.

This was known as the *master oscillator-power amplifier* (MOPA) configuration and remained popular in WWII military radios and amateur equipment into the 1950s. The popular (for amateurs as war surplus) ARC-5 series military aircraft single band HF transmitters were of MOPA design using a single-triode variable-tuned oscillator and a pair of tetrode amplifier tubes in parallel. While the military ran these at more conservative ratings, amateurs used them to provide up to 100 W output, all from a box the size of a small loaf of bread. Two examples are shown in **Figure 10**.

Crystal-Controlled Transmitters

The transmitters discussed so far operated on frequencies under the control of variable-tuned LC circuits. This provided a certain amount of flexibility, but had the drawback of making the operating frequency uncertain. Stability over time, and with variations in supply voltage and temperature, was not as good as might be desired.

Properties of crystal structures were studied at least as far back as the 18th century, but the first major application of the piezoelectric effect that converts crystal motion to voltage and back occurred during WWI. The French developed a piezoelectric ultrasonic transducer that was used as the transmitter in an acoustical submarine detection system. Between WWI and WWII, the use of resonant wafers of quartz crystal as frequency determining elements in oscillators became feasible and packaged crystals became readily available.

Crystal frequency control was particularly suitable for radios that operated on assigned fixed channels, common to most services other than Amateur Radio. Still, crystal-controlled transmitters were popular in amateur service, with most stations having a large collection of crystals that could be selected to change frequency. For many years, the old entry-level Novice class amateur required crystal control of the transmitter. **Figure 11** shows a selection of popular crystal types.

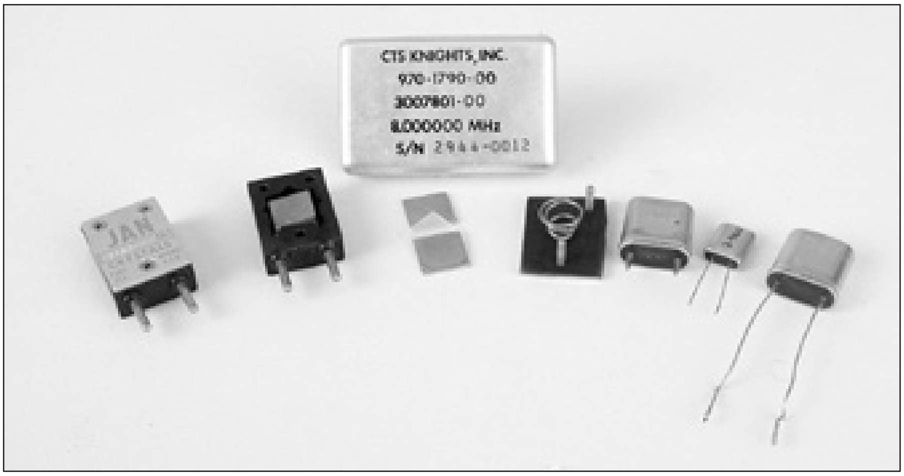


Figure 11 — Sample of typical frequency control crystals for amateur use. The units on the right are modern sealed types. On the far left is a WWII era crystal with a disassembled type FT-243 holder in the center.

Multiband Transmitters

The transmitters discussed so far generally operated over a single band or frequency range. In order to operate on multiple bands, a number of approaches were used. The early amateur bands were selected to be harmonically related in order to avoid interference to other services from spurious harmonic signals. The amateur bands of the early years were 160, 80, 40, 20, 10, 5 and 2½ meters.

A popular approach to early vacuum tube transmitters designed for multiple bands was to use frequency multiplier stages to raise the frequency to a desired harmonic. A frequency multiplier was just an amplifier operated in class C with the output circuit tuned to a multiple of the input frequency. The plate current pulses would cause the output tank circuit to ring at the selected harmonic, filling in the cycles between the pulses. If the Q of the tank circuit were high enough, harmonics through the third or fourth could be generated with acceptable distortion. Additional tuned stages would reinforce the desired harmonic and reject the other signals.

A block diagram of a typical period transmitter for 80, 40 and 20 meters is shown in **Figure 12**. Note that a particular advantage of this configuration is that a single crystal at say 3.505 MHz can also be used at 7.010 and 14.020 MHz. This arrangement was quite common in transmitters designed for frequency modulation in which not only the

Figure 10 — A pair of 7 to 9.1 MHz ARC-5 transmitters from WWII. On the left is the US Army Air Forces version (BC-459), and a Navy equivalent (T-22, ARC-5) is shown on the right. These were available for about $5 in "new in box" condition after WWII.

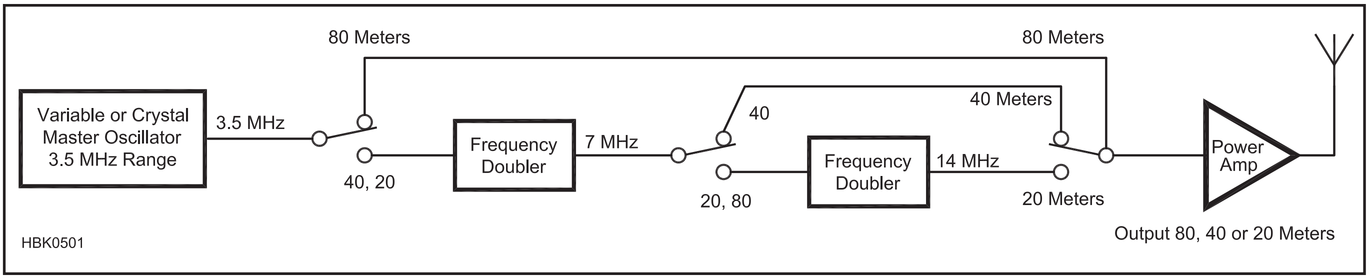


Figure 12 — Block diagram of typical frequency multiplier transmitter.

frequency, but also the deviation, is multiplied at each step.

While the figure shows cascaded doublers, some used circuits switchable between doublers, triplers and quadruplers to avoid the need for additional stages. Note that a limitation of this arrangement is that any drift or instability is also multiplied, making it necessary that particular attention is paid to oscillator stability. The tuning rate of variable oscillators, if used, is also different on each band.

Heterodyne Receivers Solve Many Problems

A receiver design that avoids many of the issues inherent in a TRF receiver is called a *heterodyne*, or often *superheterodyne* (*superhet* for short), receiver. This design was proposed by then Major Edwin Armstrong near the close of World War I for use in direction finding equipment. The superhet combines the input signal with a locally generated signal in a nonlinear device called a *mixer* to result in the sum and difference frequencies as shown in **Figure 13**. The combination of the mixer and oscillator is often called a *frequency converter*. The receiver may be designed so the output signal is anything from dc (a so-called *direct-conversion* receiver) to any frequency above or below either of the two frequencies. The major benefit is that most of the gain, bandwidth setting and processing can then be performed at a single frequency.

Changing the frequency of the local oscillator (ie, turning the VFO control in a modern radio) makes a corresponding change in the input frequency that is translated to the output, along with all its modulated information. In most receivers the mixer output frequency is designed to be an RF signal, either the sum or difference — the other being filtered out at this point. This output frequency is called an *intermediate frequency* or *IF*. The IF amplifier system can be designed to provide the selectivity and other desired characteristics centered at a single fixed frequency — much easier to process

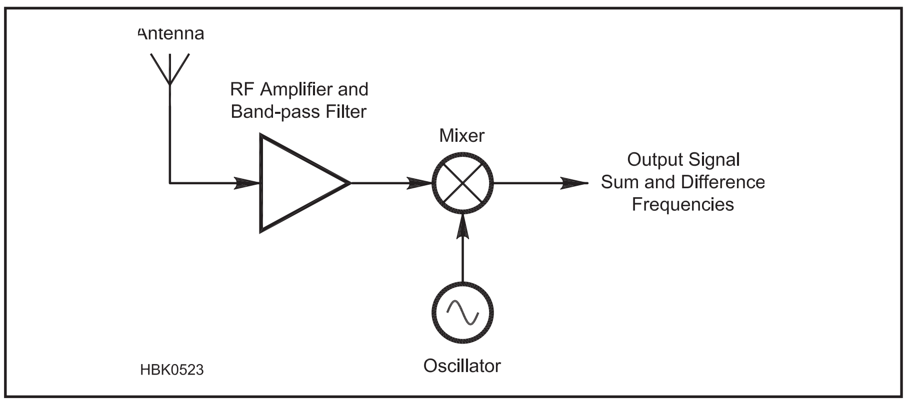


Figure 13 — Basic architecture of a heterodyne conversion stage. The output signal is at a fixed intermediate frequency.

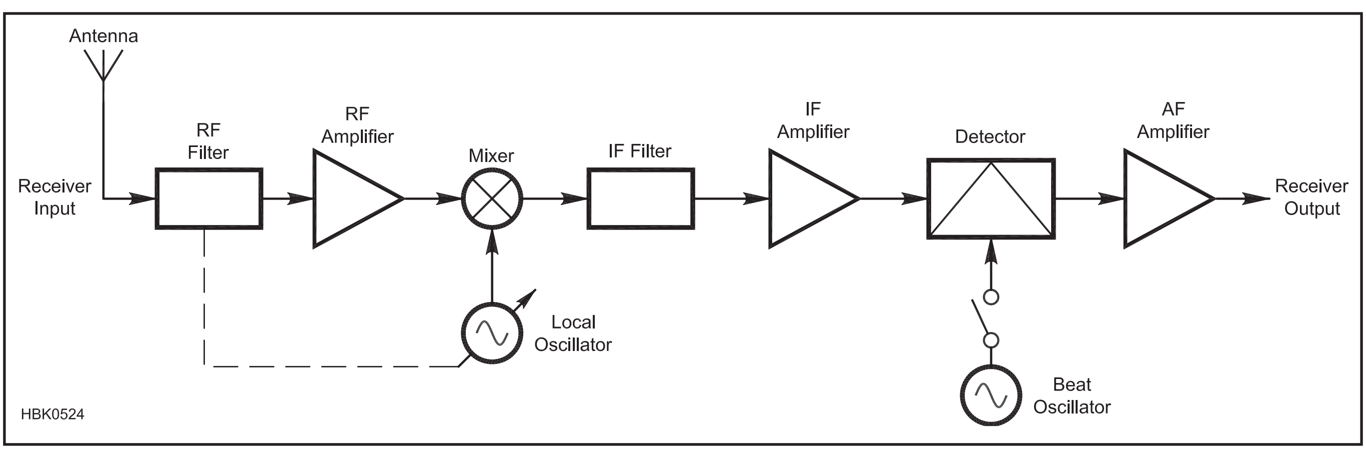


Figure 14 — Elements of a traditional superheterodyne radio receiver.

than the variable arrangement of a TRF set.

A block diagram of a typical superhet is shown in **Figure 14**. In traditional form, the RF filter is used to limit the input frequency range to those frequencies that include only the desired sum or difference but not the other — the so-called *image* frequency. The dotted line represents the fact that in receivers with a wide tuning range, the input filter often tracks along with the local oscillator. While this is similar to one of the issues with TRF receivers, note that generally there are fewer tuned circuits involved in a superhet, making the tracking easier to accomplish. The IF filter is traditionally used to establish operating selectivity — that required by the information bandwidth.

In many instances, it is not possible to achieve all the receiver design goals with a single-conversion receiver, so multiple conversion steps are taken. Traditionally, the first conversion is tasked with removing the RF image signals, while the second allows processing of the IF signal to provide the information-based IF processing. A second mixer is used to detect the IF signal, translating it to audio. The configuration is shown in Figure 14.

In a typical configuration, the local oscillator (LO) and RF amplifier stages are adjusted so that as the LO is changed in frequency, the RF amplifier is also tuned to the appropriate frequency to receive the desired station. An example may help. Let's pick a common IF frequency used in an AM broadcast radio, 455 kHz. Now if we want to listen to a 600 kHz broadcast station, the RF stage should be set to amplify the 600 kHz signal and the LO should be set to 600 + 455 kHz or 1055 kHz.[6] The 600 kHz signal, along with any audio information it contains, is translated to the IF frequency and is amplified. It is then detected, just as if it were in a TRF receiver at 455 kHz.

Note that to detect standard AM signals, the second oscillator, usually called a *beat frequency oscillator* or BFO, is turned off since the AM station provides its own carrier signal over the air. Receivers designed only for standard AM reception, the typical "table or kitchen radio," generally don't have a BFO at all.

It's not clear yet that we've gained anything by doing this; so let's look at another example. If we decide to change from listening to the station at 600 kHz and want to listen to another station at, say, 1560 kHz, we can tune the single dial of our superhet to 1560 kHz. With the appropriate ganged and tracked tuning capacitors, the RF stage is tuned to 1560 kHz, and the LO is set to 1560 + 455 or 2010 kHz and now that station is translated to our 455 kHz IF. Note that the bulk of amplification can take place at the 455 kHz IF frequency, so not as many stages must be tuned each time we change to a new frequency. Note also that with the superheterodyne configuration the selectivity (the ability to separate stations) occurs primarily in the intermediate-frequency (IF) stages, and is thus the same no matter what frequencies we choose to listen to. The superhet design has thus eliminated the major limitations of the TRF at the cost of two additional building blocks.

It should be no surprise, then, that the superhet, in various flavors has become the primary receiver architecture in use today. Just as WWI was coming to a close, the concept was introduced by Major Edwin Armstrong, a US Army artillery officer. The superhet quickly gained popularity and, following the typical patent battles of the times (and hardly unknown today) became the standard of a generation of vacuum-tube broadcast receivers. These were found in virtually every US home from the late 1920s through the 1960s, when they were slowly replaced by transistor sets, but still of superhet design.

Significant development of the superhet to optimize its potential for use by amateurs occurred at the ARRL in the early 1930s. James Lamb, W1CEI, (later W1AL, now deceased) was *QST* Technical Editor from 1929 to 1939, a period of significant activity. While he had a number of inventions, leading to nine patents, perhaps his most significant invention was the single-signal receiver, described in a series of *QST* articles of the period (and available online from the *QST* archive).[5-8]

The single-signal concept made use of a single piezoelectric crystal resonant at the IF frequency. This formed a filter that was part of the IF amplifier. By carefully adjusting the frequency of the BFO and the crystal phasing control, settings could be found that would eliminate the response on the other side of the zero beat, thus reducing by half the apparent number of signals that the receiver could respond to. Lamb's prototype single-signal receiver is on display at the ARRL Headquarters Museum of Amateur Radio.

This technology was a significant part of communications receivers for at least 50 years, although the use of band-pass filters, rather than the single crystal type of Lamb's design, started to take over with the shift of voice operation to single sideband starting in the 1950s. While Lamb's approach was focused on CW operation with its narrow bandwidth, his filter was also useful for AM voice. By use of the phasing control, a sharp notch in the passband could be used to null out an interfering carrier, such as the heterodyne that resulted from the carriers of two AM transmitters on adjacent frequencies.

Voice Enters the Picture

AM Voice — Like the "Big Boys"

In 1920, KDKA in Pittsburgh, Pennsylvania, became the first federally licensed commercial AM broadcast station in the country. KDKA operated on 1020 kHz, initially using a 100 W vacuum tube transmitter. Amateur Radio operators were using voice communication in the same era with similar technology.

The typical AM transmitter used the same RF equipment that was used for CW. To apply the voice modulation, an audio amplifier stage, typically with a transformer coupled output, was inserted in series with the high voltage supply, providing power to the plate of the final stage of the RF deck. The audio amplifier was adjusted so that the plate voltage would swing from zero to twice the normal anode voltage of the RF stage with the audio voltage. This required an amplifier that could deliver 50% of the dc input power of the final RF stage. A block diagram of the setup is shown in **Figure 15**.

In this scheme, the nonlinear (class C) RF amplifier acted like a high level mixer, providing *high-level modulation* by multiplying the RF signal times the audio signal in its plate circuit. Multiplying (in other words, modulating) a carrier with a single tone results in the tone being translated to frequencies at the sum and difference of the two. An analog view of the signals is shown in **Figure 16**. A representative design of an early modulator was described by James Lamb in a 1931 *QST* article.[9]

Thus, if a transmitter were to multiply a 600 Hz tone by a 600 kHz carrier signal, we would generate additional new signals at 599.4 and 600.6 kHz. If instead we were to modulate the 600 kHz carrier signal with a band of frequencies corresponding to human speech of 300 to 3300 Hz (also called *toll quality* — a term carried over from long-distance telephone systems), we would have additional bands of signals carrying the information and extending from 596.7 to 603.3 kHz, as shown in **Figure 17**. These bands are called *sidebands,* and some form of sidebands is present in any AM signal that is carrying information.

Note that the total bandwidth of this AM voice signal is twice the highest frequency transmitted, or 6600 Hz. If we choose to transmit speech and limited music, we might allow modulating frequencies up to 5000 Hz, resulting in a bandwidth of 10,000 Hz or 10 kHz. This is the standard channel spacing that commercial AM broadcasters use in the US. In actual use, stations on adjacent channels are generally separated geographically, so broadcasters can extend some energy into the adjacent channels for improved fidelity. We refer to this as a *narrow-bandwidth* mode.

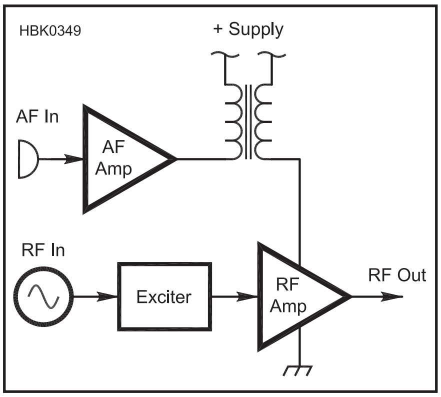


Figure 15 — Elements of an amplitude modulated transmitter using high-level modulation.

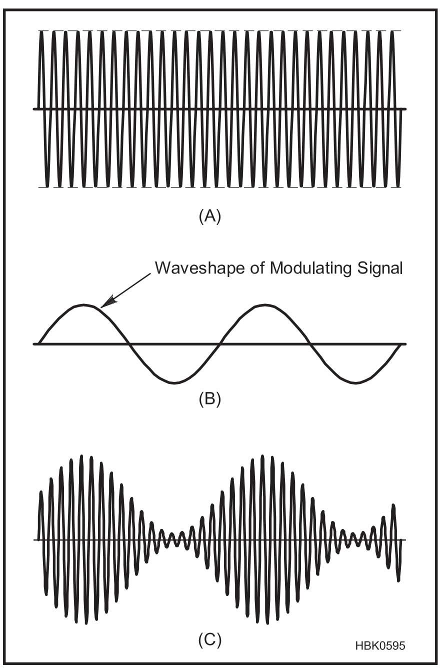


Figure 16 — Graphical representation of the amplitude modulation process that occurs in the system of Figure 15. At (A) the RF signal, at (B) the audio (modulating) waveform and at (C) the RF signal modulated by the audio signal using high-level

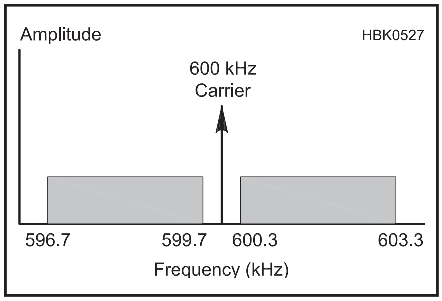


Figure 17 — Graphical representation of spectrum of the amplitude modulation used to send a voice signal on a 600 kHz carrier.

What does this say about the bandwidth needed for a receiver? If we want to receive the full information content transmitted by a US AM broadcast station, then we need to set the bandwidth to at least 10 kHz. What if our receiver has a narrower bandwidth? Well, we will lose the higher frequency components of the transmitted signal — perhaps ending up with a radio suitable for voice but not very good at reproducing music.

On the other hand, what is the impact of having too wide a bandwidth in our receiver? In that case, we will be able to receive the full transmitted spectrum, but we will also receive some of the adjacent channel information. This will sound like interference and reduce the quality of the desired signal we are receiving. If there are no adjacent channel stations, we will receive any additional noise from the additional bandwidth and minimal additional information. The general rule is that the received bandwidth should be matched to the bandwidth of the signal we are trying to receive to maximize *signal-to-noise ratio* (*SNR*) and to minimize interference.

As the receiver bandwidth is reduced, intelligibility suffers, although the SNR is improved. With the carrier centered in the receiver bandwidth, most voices are difficult to understand at bandwidths less than around 4 kHz. In cases of heavy interference, full carrier AM can be received as if it were SSB, as described later, with the carrier inserted at the receiver and the receiver tuned to whichever sideband has the least interference. A view of a typical amateur AM and CW station from the 1950s is shown in **Figure 18**.

Single Sideband Suppressed Carrier

In looking at Figure 17, you might have noticed that both sidebands carry the same information, and are thus redundant. In addition, the carrier itself conveys no information. It is thus possible to transmit a *single sideband* and *no carrier*, as shown in **Figure 19**, relying on the BFO (beat frequency oscillator) in the receiver to provide a signal with which to multiply the sideband in order to provide demodulated audio output.

The implications in the receiver are that the bandwidth can be slightly less than half that required for double sideband AM (DSB). There must be an additional mechanism to carefully replace the missing carrier within the receiver. This is the function of the BFO, which must be at just exactly the right frequency. If the frequency is set improperly, even by a few Hz, a baritone can come out sounding like a soprano and vice-versa!

The standard commercial AM format is very convenient for receiver design, since the carrier needed to demodulate the received sidebands is sent along with the sidebands. Applying the total received signal to a detector or multiplier allows the audio to be recovered without having to worry about any of the finer points we will discuss later. This is very cost-effective in a broadcast environment in which there are many inexpensive receivers and only a relatively few expensive transmitters. For amateur or commercial point-to-point use, there is usually a single receiver associated with the transmitter on any particular link, and it makes sense to divide the complexity more evenly to reduce the total system cost.

For reception of CW, suppressed-carrier single-sideband voice (SSB) or on-off or

Figure 18 — A typical 1950s amateur AM and CW station used a separate transmitter and receiver. Shown here is W1ZR's 1950s-era station with an E. F. Johnson Viking II transmitter (100 W) with separate VFO on the left and a National HRO-60 receiver on the right.

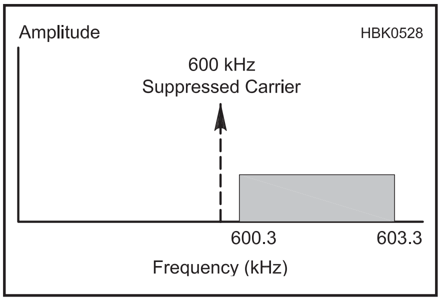


Figure 19 — Graphical representation of the spectrum of a single sideband AM voice signal adjacent to its suppressed 600 kHz carrier.

frequency-shift keyed (FSK) signals, a second *beat frequency oscillator* or BFO is employed to provide an audible voice or an audio tone (or tones) at the output for the operator to receive by ear (or for further processing of digital modes such as FSK). This is the same as a heterodyne mixer with an output centered at dc, although the IF filter is usually designed to remove one of the output products.

This creates a requirement for a much more stable receiver design with a much finer tuning system — a more expensive proposition. An alternate is to transmit a reduced level carrier and have the receiver lock on to the weak carrier, usually called a *pilot* carrier. Note that the pilot carrier need not be of sufficient amplitude to demodulate the signal, just enough to allow a BFO to lock to it. These alternatives are effective, but they tend to make SSB receivers expensive, complex and most appropriate for the case in which a small number of receivers are listening to a single transmitter, as is the case of two-way communication.

Note that the bandwidth required to demodulate an SSB signal effectively is actually less than half that required for an AM signal because frequencies immediately adjacent to the AM carrier need not be received. Thus the toll quality spectrum of 300 to 3300 Hz can be received in a bandwidth of 3000 not 3300 Hz. Early SSB receivers typically used a bandwidth of around 3 kHz, but with the heavy interference frequently found in the amateur bands, it is more common for amateurs to use bandwidths of 1.8 to 2.4 kHz with a corresponding loss of some high and low frequency speech components.

Single-sideband makes it possible to transmit the information with just one of the sidebands and no carrier. In so doing, we use somewhat less than half the bandwidth, a scarce resource, and also consume much less transmitter power by not transmitting the carrier and the second sideband.

Managing Selectivity and Images

After WWII, activity increased dramatically on the amateur bands and across the HF spectrum generally. This generated higher performance requirements for receiver selectivity and for the rejection of *images* — undesired reception of signals at alternate frequencies that also generate mixing products at the receiver's IF. Advances in receiver design during the 1940s were applied and several improvements of the superheterodyne architecture appeared.

Double and Triple Conversion

In the 1950s, a new technique called *double-conversion* was defined to solve the image problem. Rather than having to decide between a high IF frequency for good image rejection, or a low IF frequency for narrow channel selectivity, some bright soul decided to do both! As shown in **Figure 20**, a conversion of the desired signal to a relatively high IF is followed by a second conversion to a lower IF to set the selectivity. This arrangement solved the image problem nicely, while the rest of the receiver could be pretty much kept the same as before. A number of manufacturers offered revised receivers in the fifties that added an additional conversion stage, often just above 10 or 15 MHz, regions that had poor image rejection in the original design. Note that major changes were not required in the receiver, and some looked just like their single-conversion predecessors from the outside. The extra stage was shoehorned in, with just a retuning of the first local oscillator required.

Some manufacturers decided that if two conversions were good, three could be even better. These designs often began with the same basic architecture, but converted the 455 kHz second IF down to a third IF, often in the 50 to 100 kHz range, for even sharper selectivity. This approach lasted until crystal lattice filters that outperformed the low frequency LC circuits became available, eliminating the need for the additional conversion stage.

The Collins System

In the 1940s, a visionary radio pioneer, Arthur Collins, WØCXX, founder of Collins Radio in the 1930s, came up with another approach to double-conversion. One problem with earlier MF and HF receivers was that they used a standard LC (first) oscillator using a variable capacitor of the type used in

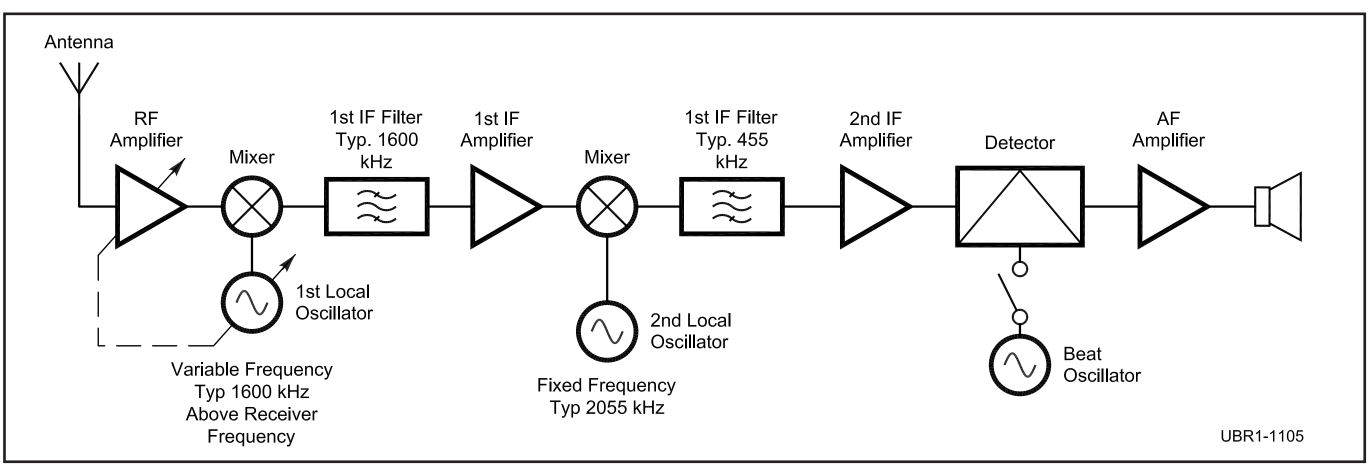


Figure 20 — An early type double-conversion superhet receiver.

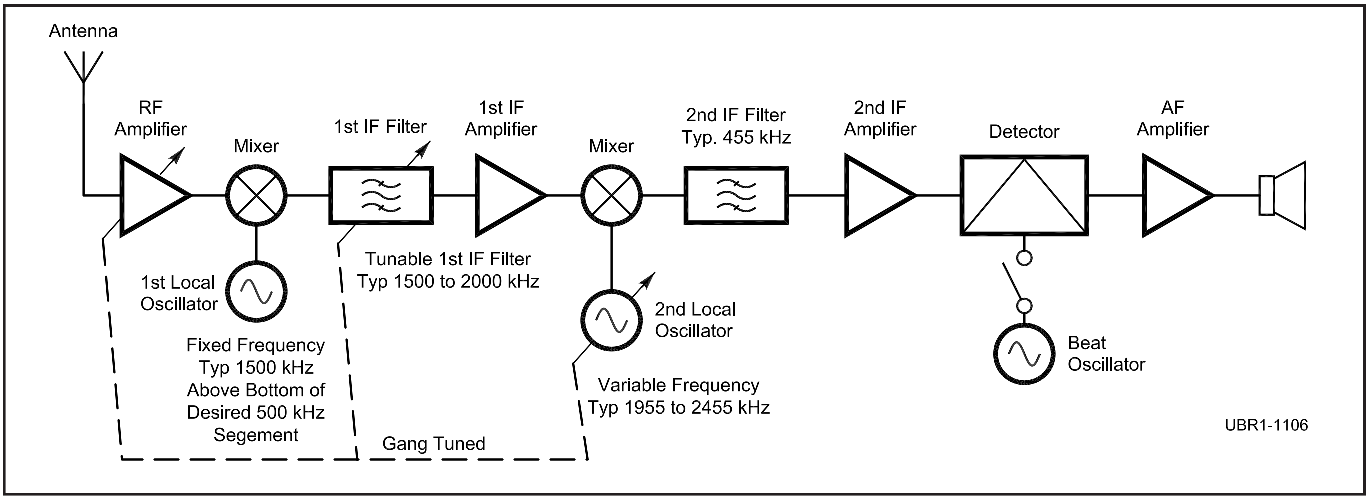


Figure 21 — A double-conversion superhet receiver using the Collins system desribed in the text.

broadcast receivers. These capacitors typically had a 9:1 capacitance range, resulting in a 3:1 frequency range. The typical tuning arrangement for multiband receivers was 0.5 to 1.5, 1.5 to 4.5, 4.5 to 13.5, and 13.5 to 30 MHz. This covered frequencies from the AM broadcast band to the top of the HF range in four bands.[7] With this arrangement, the tuning rate and dial calibration marks became less and less precise as higher bands were selected, making tuning difficult.

The Collins system (see **Figure 21**) switched the variable oscillator to the second mixer and used crystal-controlled oscillators rather than variable oscillators, in the first position. Although many more bands were required (30 for the famous 51J series that covered 0.5 to 30.5 MHz), each tuned with exactly the same tuning rate. Collins went a step further and designed an inductance-tuned oscillator (usually called a *permeability-tuned oscillator* or *PTO*) that was linear throughout its range. These oscillators could be tuned to the nearest 1 kHz from beginning to end, avoiding the tuning uncertainty of other receiver types. For this to work properly, synchronized or gang tuning is required between the oscillator, first IF variable filter and RF stages. This was no problem for the engineers at Collins Radio, but their equipment brought a premium price.

The Pre-Mixed Arrangement

A third approach to double-conversion was a mix of the first two (no pun intended). The pre-mixed arrangement (**Figure 22**) uses a single variable oscillator range, as with the Collins design, but does the mixing outside of the signal path. This avoids the need for variable tuning at the first IF, allowing a tighter roofing filter (first IF filter), but the trade-off is the need for filtering at the output of the pre-mixer (not shown).

High Frequency Crystal Lattice Filters

Just as double-conversion receivers were settling into becoming the "way things were done," the use of piezoelectric crystals as elements of band-pass filters became feasible. While crystal lattice filters at 455 kHz were popular following WWII because of availability of large stocks of surplus crystals at that range, they didn't address the image response issue. They just provided more selective filter responses at the same IF frequency. Starting in the 1950s, crystal lattice filter technology moved higher in the MF region and into HF, improving receiver selectivity dramatically.

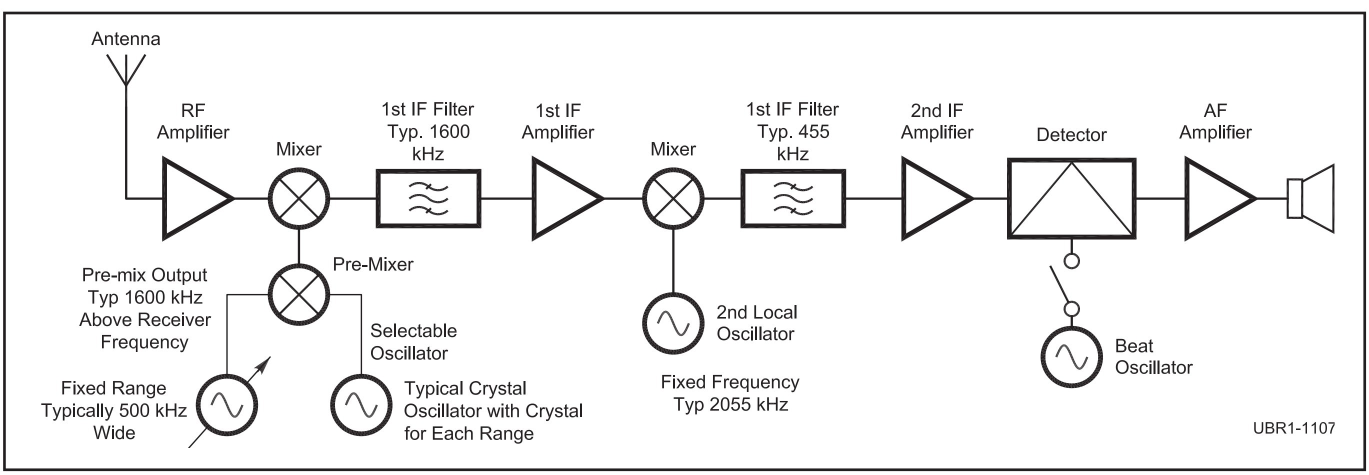


Figure 22 — A double-conversion superhet receiver with the pre-mixed local oscillator configuration.

Transmitting Single Sideband

There are a number of ways of generating a single sideband signal for transmission. While other approaches are possible, the most popular technique used in Amateur Radio, and most other services, is called the *filter method*, in which a selective filter is used to eliminate the undesired sideband from a DSB suppressed carrier signal. The next most frequently encountered technique is called the *phasing method*, which takes advantage of the trigonometric properties of the sinusoidal waves that we're dealing with. We will describe both here. (Both methods are described in more detail in the **Modulation** chapter of *The ARRL Handbook*.)

The Filter Method of SSB Generation

The block diagram of a simple single-sideband suppressed-carrier (SSB) transmitter is shown in **Figure 23**. This transmitter uses a balanced mixer as a *balanced modulator* to generate a double-sideband suppressed-carrier signal without a carrier. (See the **Mixers, Modulators, and Demodulators** chapter of *The ARRL Handbook* for more information on these circuits.) That signal is then sent through a filter designed to pass just one (either one, by agreement with the receiving station) of the sidebands.[10] Depending on whether the selected sideband is above or below the carrier frequency, the signal is called *upper sideband* (*USB*) or *lower sideband* (*LSB*), respectively. The resulting SSB signal is amplified to the desired power level and we have an SSB transmitter.

While a transmitter of the type in Figure 23 with all processing at the desired transmit frequency will work, the configuration is not commonly used. Instead, the carrier oscillator and sideband filter are often placed at an intermediate frequency that is heterodyned to the operating frequency as shown in **Figure 24**. The reason is that the sideband filter is a complex narrow-band filter and most manufacturers would rather not have to supply a new filter design every time a transmitter is ordered for a new frequency. Many SSB transmitters can operate on different bands as well, so this avoids the cost of additional mixers, oscillators and expensive filters.

Note that the block diagram of our SSB transmitter bears a striking resemblance to the diagram of a superheterodyne receiver as shown in Figure 14, except that the signal path is reversed to begin with information and produce an RF signal. The same image rejection requirements for intermediate frequency selection that were design constraints for the superhet receiver apply here as well.

The Phasing Method

Most current transmitters use the method of SSB generation shown in Figure 23 and discussed in the previous section to generate the SSB signal. This is done in two steps — first a balanced modulator is used to generate sidebands and eliminate the carrier, then a filter is used to eliminate the undesired sideband, and often to improve carrier suppression as well.

The *phasing method* of SSB generation uses two balanced modulators and a phase-shift network for both the audio and RF carrier signals to produce the upper sideband signal as shown in **Figure 25**. By a shift in the sign of either of the phase-shift networks, the opposite sideband can be generated. This method trades a few phase-shift networks and an extra balanced modulator for the sharp sideband filter of the filter method. While it looks deceptively simple, a limitation is in the construction of a phase-shift network that will have a constant 90° phase shift over the whole audio range. Errors in phase shift result in less than full carrier and sideband suppression. Nonetheless, there have been some successful examples offered over the years.

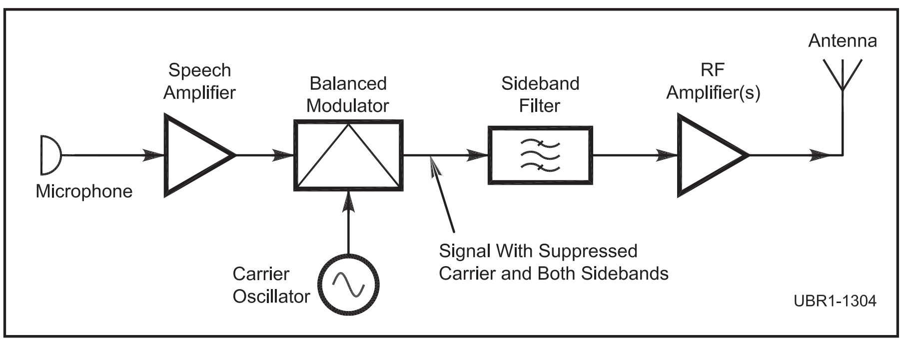


Figure 23 — Block diagram of a single-frequency filter-type single sideband suppressed carrier (SSB) transmitter.

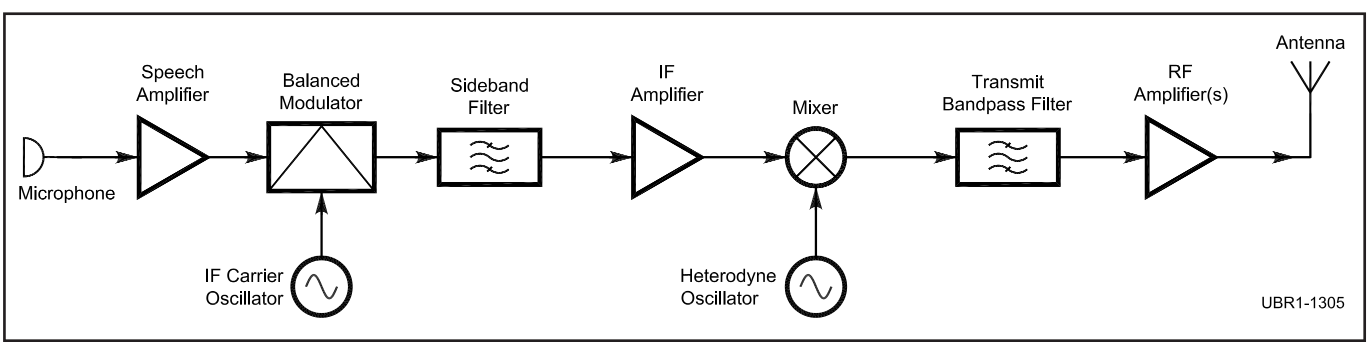


Figure 24 — Block diagram of a heterodyne filter-type SSB transmitter for multiple frequency operation.

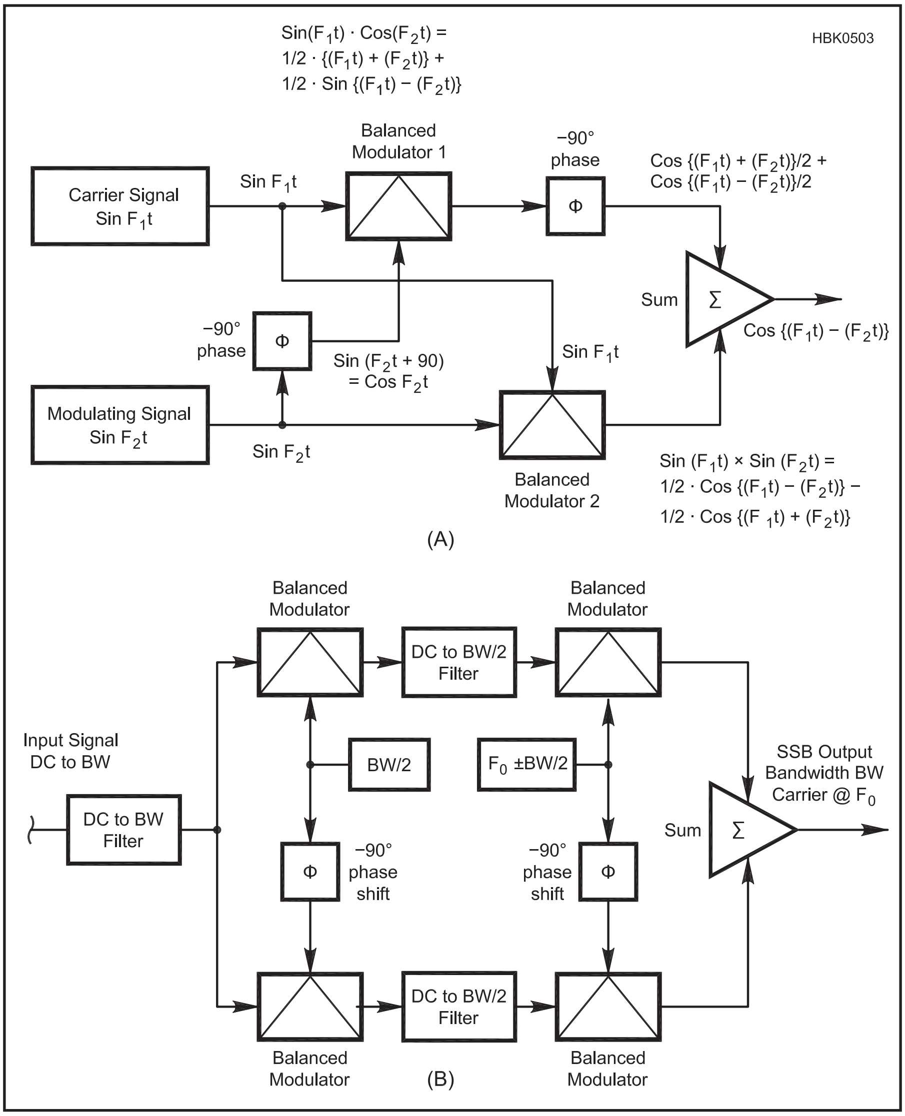


Figure 25 — Block diagrams of phasing type SSB transmitters for single frequency operation.

The SSB Transceiver is Born

A major evolution step of Amateur Radio technology was the migration from separate transmitters and receivers for SSB to a single package that contained both. This was called a *transceiver*, a combination of the words *trans*mitter and re*ceiver*. This was based on the observation that, unlike AM transmitters of the period, most SSB transmitter designs looked a lot like the architecture of a modern receiver, except configured in reverse, as in the form of a heterodyne transmitter as shown in Figure 24.

Figure 26 — An early SSB transceiver, the 1957 vintage Collins KWM-2(A). [Photo courtesy of the Collins Collectors Association]

Some early heterodyne SSB transmitters and transceivers took advantage of the fact that the 9 MHz heterodyne oscillator frequency could actually be used on both 80/75 and 20 meters with the same 5.0 to 5.5 MHz frequency range in the VFO. The difference product at 9 MHz minus the VFO frequency (4 to 3.5 MHz) was used for the lower band, while the sum product at 9 MHz plus the VFO frequency (14 to 14.5 MHz) was used for 20 meters. This was a popular dual-band scheme in the early days of SSB voice. One drawback was that the VFO tuned across the band from low to high frequencies in the opposite direction on each band. For many hams of the day, it was a small price for the simplicity.

The transmit-receive (TR) switching arrangements in transceivers allowed use of the same oscillators and filters for both receiving and transmitting, resulting in the transmitter automatically being tuned to the receive frequency. This was a significant advantage for SSB operation because the two frequencies needed to be close together to provide intelligible speech. One of the earliest successful HF SSB transceivers was the KWM-2 series by Collins Radio, shown in **Figure 26**.

Most SSB transceivers did not include a built in power supply, and with the elimination of the need for the heavy modulation transformers of AM transmitters, the resulting transceiver could be light and compact. Different power supplies could be selected for operation from ac mains for home station operation or from dc systems for mobile operation in vehicles. To say that the HF SSB transceiver revolutionized Amateur Radio would not be an overstatement. Fortunately, all the benefits of the transceiver for SSB operation were also beneficial for CW and data modes.

Narrowband Frequency Modulation

Another popular voice mode is *frequency modulation*, or FM. FM can be found in a number of variations depending on purpose. In Amateur Radio and commercial mobile communication use on the shortwave bands, it is universally *narrowband FM* or *NBFM*. In NBFM, the frequency deviation is limited to around the maximum modulating frequency, typically 3 kHz. The bandwidth requirements at the receiver can be approximated by 2 × (D + M), where D is the deviation and M is the maximum modulating frequency. Thus 3 kHz deviation and a maximum voice frequency of 3 kHz results in a bandwidth of 12 kHz, not far beyond the requirements for broadcast AM. (Additional signal components extend beyond this bandwidth, but are not required for voice communications.)

In contrast, broadcast or *wideband FM* or *WBFM* occupies a channel width of 150 kHz. Originally, transmitted signal bandwidth was extended to provide high-fidelity reproduction of up to 15 kHz audio with an improved signal-to-noise ratio (SNR). However, with multiple channel stereo and sub-channels all in the same allocated bandwidth, the primary channel is today limited to about the maximum audio signal bandwidth.

In the US, FCC amateur rules limit wideband FM use to frequencies above 29 MHz. Some, but not all, HF communication receivers provide for FM reception. For proper FM reception, two changes are required in the receiver architecture as shown within the dashed line in **Figure 27**. The fundamental change is that the detector must recover information from the frequency variations of the input signal. The most common type is called a *discriminator*. The discriminator does not require the BFO, so the BFO is turned off or eliminated in a dedicated FM receiver. Amplitude variations convey no information in FM, so they are generally eliminated by a *limiter*. The limiter is a high-gain IF amplifier stage that clips the positive and negative peaks of signals above a certain threshold. Most noise of natural origins is amplitude modulated, so the limiting process also strips away noise from the signal.

Transmitters using frequency modulation (FM) or phase modulation (PM) are generally grouped into the category of *angle modulation* since the resulting signals are often indistinguishable. An instantaneous change in either frequency or phase can create identical signals, even though the method of modulating the signal is somewhat different. To generate an FM signal, we need an oscillator whose frequency can be changed by the modulating signal.

We can make use of an oscillator whose frequency can be changed by a "tuning voltage." If we apply a voice signal to the tuning voltage connection point, we will change the frequency with the amplitude and frequency

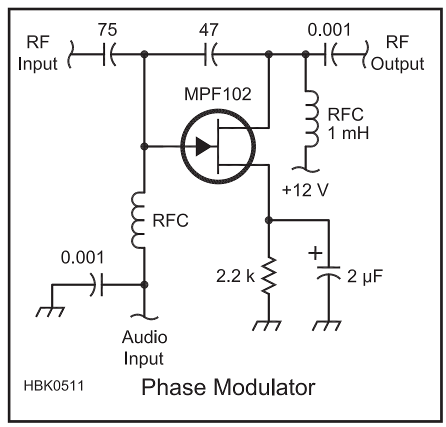


Figure 28 — Simple FET phase modulator circuit.

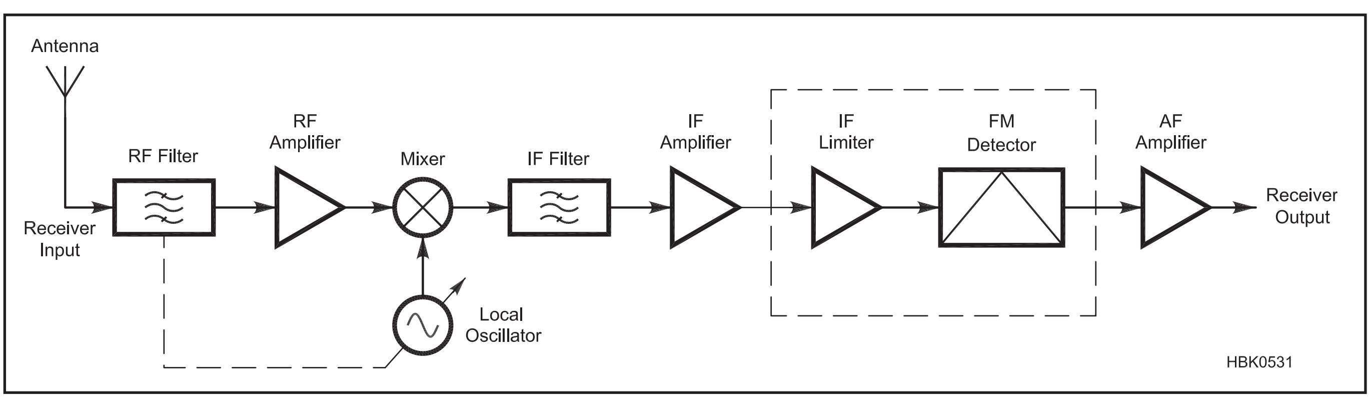


Figure 27 — Block diagram of an FM superhet receiver. Changes from an AM receiver are shown in the dashed box.

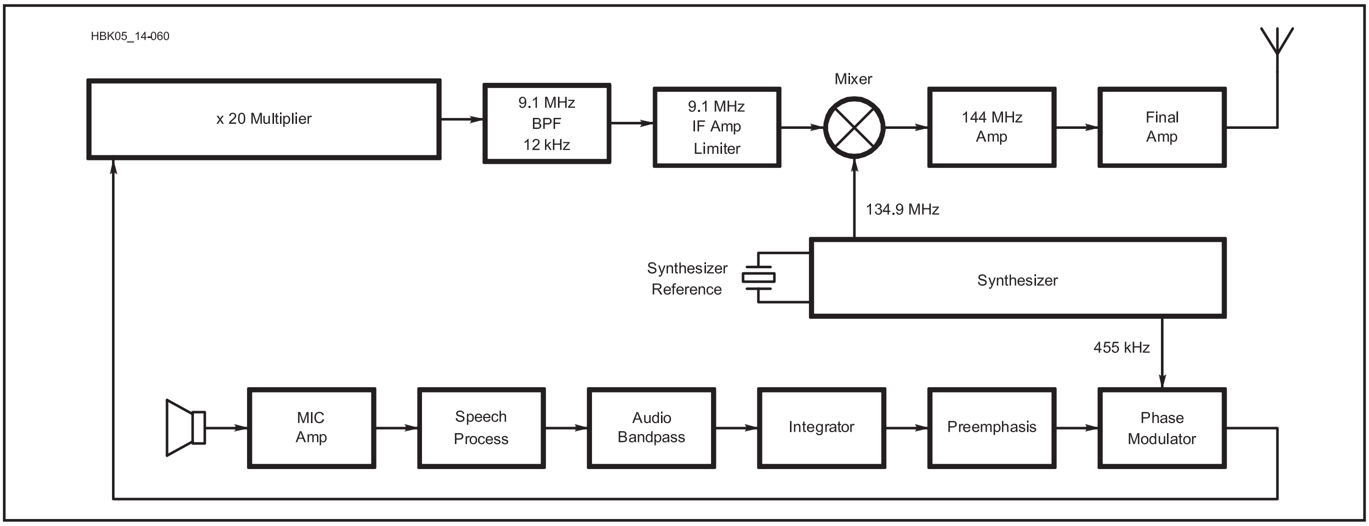


Figure 29 — Block diagram of a VHF/UHF NBFM transmitter using the indirect FM (phase modulation) method.

of the applied modulating signal, resulting in an FM signal.

The phase of a signal can be varied by changing the values of an R-C phase-shift network. One way to accomplish phase modulation is to have an active element shift the phase and generate a PM signal. In **Figure 28**, drain current through the MPF102 field-effect transistor is varied with the applied modulating signal, varying the phase shift at the stage's output. Because the effective load on the stage is changed, the carrier is also amplitude-modulated and must be run through an FM receiver-type limiter in order to remove the amplitude variations. A block diagram of a complete NBFM transmitter using the phasing method is shown in **Figure 29**.

Computers Enter the Scene

Computers have had a strong impact on all facets of life, and Amateur Radio has not escaped the influence of computer technology in a number of areas.

Operation Using Digital Modes

Communication via digital signaling has been popular with radio amateurs since teletype equipment became readily available on the military surplus market following World War II. The availability of personal computers (PCs) with sound cards and specialized software has made many forms of digital transmission easy for amateurs to accomplish without the need for special and often complex hardware. While most new digital modes are more robust and capable than traditional *radioteletype* or RTTY (pronounced "ritty"), it still remains one of the most popular modes bridging the divide between old and new technology.

Radioteletype transmission makes use of the Baudot code constructed with sequences of ON-OFF elements or bits as shown in **Figure 30.**[11] The state of each bit — ON or OFF — is represented by a signal at one of two distinct frequencies: one designated *mark* and one designated *space*. (These states are named after the early Morse tape readers that placed a pen *mark* on a paper tape when the key was down and made a *space* when the key was up.) This is referred to as *frequency shift keying* (*FSK*). The transmitter frequency shifts back and forth with each character's individual elements.

Amateur Radio operators typically use a 170 Hz separation between the mark and space frequencies, depending on the data rate and local convention, although 850 Hz is sometimes used. The minimum bandwidth required to recover the data is somewhat greater than twice the spacing between the tones. Note that the tones can be generated by directly shifting the carrier frequency (*direct FSK*), or by using a pair of 170 Hz spaced audio tones applied to the audio input of an SSB transmitter (*audio FSK* or AFSK), often supplied by a computer sound card. Direct FSK and AFSK sound the same to a receiver.

Note that if the standard audio tones of 2125 Hz (mark) and 2295 Hz (space) are used, they fit within the bandwidth of a voice channel and thus a voice transmitter and receiver can be employed without any additional processing needed outside the radio equipment. Alternately, the receiver can employ detectors for each frequency and provide an output directly to a computer.

Some receivers provide a narrow CW bandwidth filter with the center frequency shifted to midway between the tones (2210 Hz). The

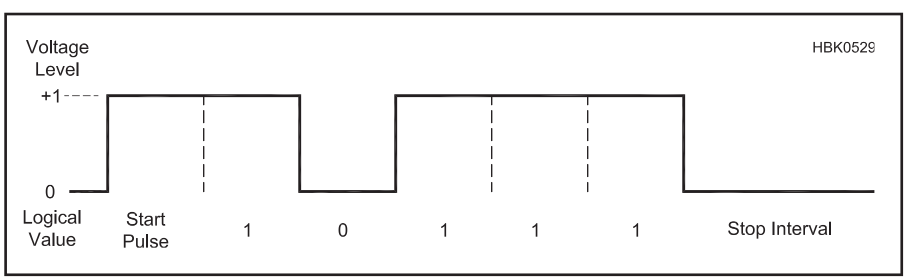


Figure 30 — Voltage pulses corresponding to the letter "A" in Baudot code, including start and stop pulses.

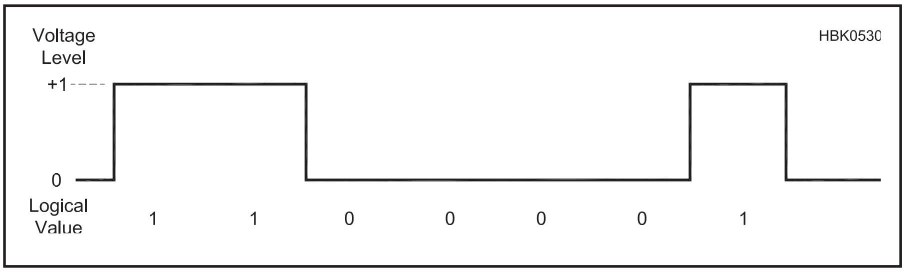


Figure 31 — Voltage pulses corresponding to the lower case letter "a" in ASCII code.

most advanced receivers provide a separate filter for mark and space frequencies, thus maximizing interference rejection and signal-to-noise ratio (SNR). Using a pair of tones for FSK or AFSK results in a maximum data rate of about 1200 bits per second (bps) over a high-quality voice channel.

Phase shift keying (*PSK*) can also be used to transmit bit sequences, requiring good frequency stability to maintain the required time synchronization to detect shifts in phase. If the channel has a high SNR, as is often the case at VHF and higher, telephone network data-modem techniques can be used.

At HF, the signal is subjected to phase and amplitude distortion as it travels. Noise is also substantially higher on the HF bands. Under these conditions, modulation and demodulation techniques designed for "wireline" connections become unusable at bit rates of more than a few hundred bps. As a result, amateurs have begun adopting and developing state-of-the-art digital modulation techniques. These include the use of multiple carriers (MFSK, Clover, PACTOR III, and others), multiple amplitudes and phase shifts (QAM and QPSK techniques), and advanced error detection and correction methods to achieve a data throughput as high as 3600 bps over a voice-bandwidth channel. (Spread-spectrum techniques are also being adopted on the UHF bands, but are beyond the scope of this discussion.)

Another code that is in common use is the ASCII (American Standard Code for Information Interchange) code that was developed for computer-to-computer communication. This is easy to generate on a PC, since it is the code that PCs use to communicate with each other. Unlike the five-unit Baudot code that can send 2^5, or 32, distinct characters and thus uses a special character to toggle between *letters* and *figures*, the seven unit ASCII code (see **Figure 31**) can send 2^7 (128) characters, enough for most printing characters including upper and lower case letters of most languages, punctuation, numerals, special characters and control signals.

The bandwidth required for data communications can be as low as 100 Hz for PSK31 to 1 kHz or more for the faster speeds of PACTOR III and Clover. Beyond having sufficient bandwidth for the data signal, the primary requirements for receivers used for data communications are linear amplitude and phase response over the bandwidth of the data signal. The receiver must also have excellent frequency stability to avoid drift, and excellent frequency resolution to enable the receiver filters to be set on frequency.

Computer Control of Radios

Most modern amateur HF transceivers provide a connection to allow information exchange with PCs. Depending on the internal architecture of the transceiver, it may be possible to completely control the transceiver through the use of the PC. This can permit remote operation using Internet communications between local and distant PCs, allowing radio operation far from the home station. In addition, logging software can make use of the link to receive frequency and mode information from the radio to automatically fill in the log, or to change frequency to find a particular station that has been reported to the PC from a remote network.

Software Defined Radio

Moving beyond processing of digital communications signals and the control of transceivers, computers have taken over many functions within radio transceivers through the use of *digital signal processing* (*DSP*). Early adaptations used PCs to perform digital signal processing on audio signals from traditional receivers, while newer technology involves building the entire radio around specialized high performance DSP integrated circuits.

Modern receivers using DSP for operating bandwidth and information detection often have an additional conversion step to a final IF at a frequency in the tens of kHz. This is established by the maximum sampling rate of a finite frequency response analog-to-digital converter (ADC). Advances in the art have resulted in ADC and processor speeds fast enough that the ADC has been moving closer and closer to the RF frequency — resulting in fewer required conversions, and in some cases, converting the RF signal directly to digital data.

The design choices described above are still found in current receiver architectures, but some recent advances have added a few new twists. While advances in microminiaturization of all circuit elements have made a radical change in the dimensions of communication equipment, perhaps most significant technology impacts on architecture come in two areas — the application of digital signal processing and direct digital synthesis (DDS) frequency generation. Both are discussed in detail in the chapter on **DSP and Software Radio Design** in *The ARRL Handbook*. An overview is presented here.

Digital signal processing provides a level of bandwidth setting filter performance not practical with other technologies. While much better than most low frequency IF LC bandwidth filters, the very good crystal or mechanical bandwidth filters in amateur gear are not very close to the rectangular shaped frequency response of an ideal filter, but rather have skirts with a 6 to 60 dB response of perhaps 1.4 to 1. That means if we select an SSB filter with a nominal (6 dB) bandwidth of 2400 Hz, the width at 60 dB down will typically be 2400 × 1.4, or 3360 Hz. Thus a signal in the next channel that is 60 dB stronger than the signal we are trying to copy (as often happens) will have energy just as strong as our desired signal.

DSP filtering approaches the ideal response. **Figure 32** shows the response of a DSP bandwidth filter with a 6 dB bandwidth of 2400 Hz as measured by the ARRL Lab. Note how rapidly the skirts drop to the noise level. In addition, while analog filtering generally requires a separate filter assembly for each desired bandwidth, DSP filtering is adjustable — often in steps as narrow as 50 Hz — in both bandwidth and center frequency. In addition to bandwidth filtering, the same DSP can often provide digital noise reduction and digital notch filtering to remove interference from fixed-frequency carriers.

Early DSP filters operated in the audio frequency range, based on the frequency

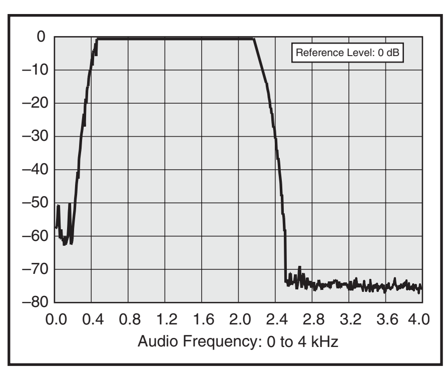


Figure 32 — Frequency response of an aftermarket 2400 Hz bandwidth DSP audio filter as measured in the ARRL Lab.

response limitations of the analog-to-digital converter (ADC) that provides the digitization for further processing. The limitation was based on the Nyquist sampling theorem (AT&T, 1924) that established that a periodic signal could be sampled and then reconstructed from its samples if sampled at least twice during each cycle of its highest frequency component. Early ADCs were limited to sampling rates less than about 50 kHz. Such a DSP filter could be added to the audio signal at the output of any receiver and work its magic on the signals in the audio stream. Unfortunately, at that time ADC and DSP that worked at typical receiver IF frequencies were either unavailable, or prohibitively expensive for amateur use.

Receiver designers quickly resurrected the earlier triple conversion receiver architecture —now with IF frequencies even lower, in the 12-20 kHz range — so that DSP could be employed in the IF rather than audio section of receivers. This allowed the receiver to eliminate interfering signals before the application of automatic gain control, avoiding gain reduction in the presence of off-channel signals. As time and technology have advanced, the clocking speed and input bandwidth of ADCs continues to increase while prices continue to fall, allowing the third IF frequency to move toward the second. Soon the additional conversion step will no longer be required, and we will see high-performance HF receivers that apply signals at the input frequency directly to the ADC.

NOTES

[1]R. Shrader, W6BNB (SK), "When Radio Transmitters Were Machines," *QST*, Jan 2009, pp 36-38.

[2]The only remaining Alexanderson alternator transmitter is the Grimeton VLF transmitter (**http://en.wikipedia.org/wiki/Grimeton_VLF_transmitter**) that operates on 17.2 kHz.

[3]R. Shrader, W6BNB (SK), "Radio Gear of Yesteryear," *QST*, Mar 1994, pp 41-43, 57.

[4]ARRL Staff, "A Short Wave Regenerative Receiver," *QST*, Dec 1916, pp 3-5.

[5]**www.arrl.org/arrl-periodicals-archive-search**

[6]J. Lamb, W1CEI, "What's Wrong With Our C.W. Receivers?," *QST*, Jun 1932, pp 9-17.

[7]J. Lamb, W1CEI, "Short-Wave Receiver Selectivity to Match Present Conditions," *QST*, Aug 1932, pp 9-16.

[8]J. Lamb, W1CEI, "An Intermediate-Frequency and Audio Unit for the Single-Signal Superhet," *QST*, Sep 1932, pp 9-15.

[9]J. Lamb, W1CEI, "High-Power Performance from the Small 'Phone Transmitter," *QST*, Dec 1931, pp 10-23.

[10]Amateur practice is to use USB above 10 MHz and LSB on lower frequencies. The exception is 60 meter channels, on which amateurs are required to use USB.

[11]The Baudot code is still the standard employed in the wire-line teletype (TTY) keyboard terminals used by hearing impaired people.

A BINAURAL I-Q RECEIVER

This little receiver was designed and built by Rick Campbell, KK7B. It was first described in the March 1999 issue of *QST*. It replaces the narrow filters and interference-fighting hardware and software of a conventional radio with a wide-open *binaural I-Q detector*. If you liken a conventional receiver to a high-powered telescope, this receiver is a pair of bright, wide-field binoculars. The receiver's classic junk-box-available-parts construction approach achieves better RF integrity than that of much commercial ham gear. A PC board and parts kit is available for those who prefer to duplicate a proven design.[1] The total construction time was only 17 hours. There are a number of toroids to wind, and performance was not compromised to simplify construction or reduce parts count. **Fig 14.64** is a photo of the front panel built by KK7B.

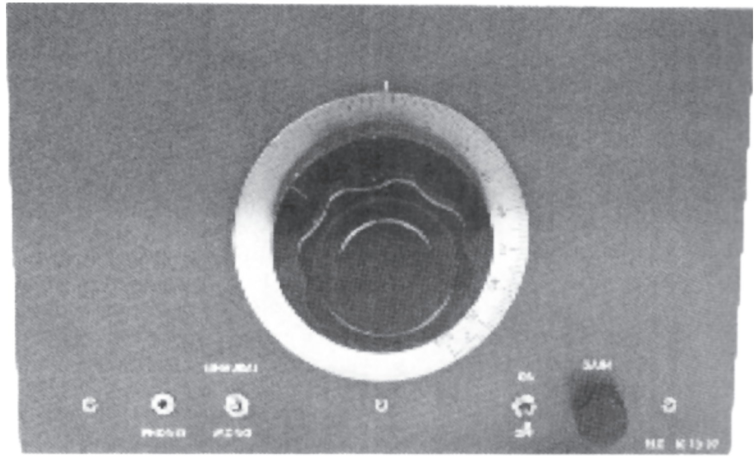


Fig 14.64—A receiver with presence . . . to fully appreciate this receiver, you've got to hear it! "Once my ears got used to the effect, they had to drag me away from this radio. This is one I gotta have!"—*Ed Hare, W1RFI, ARRL Lab Supervisor*

BINAURAL I-Q RECEPTION

Modern receivers use a combination of band-pass filters and digital signal processing (DSP) to select a single signal that is then amplified and sent to the speaker or headphones. When DSP is used, the detector often takes the form shown in **Fig 14.65**. The incoming signal is split into two paths, then mixed with a pair of local oscillators (LOs) with a relative 90º phase shift. This results in two baseband signals: an in-phase, or *I* signal, and a quadrature, or *Q* signal. Each of the two baseband signals contains all of the information in the upper and lower sidebands. The baseband pair also contains all of the information needed to determine whether a signal is on the upper or lower sideband before multi-

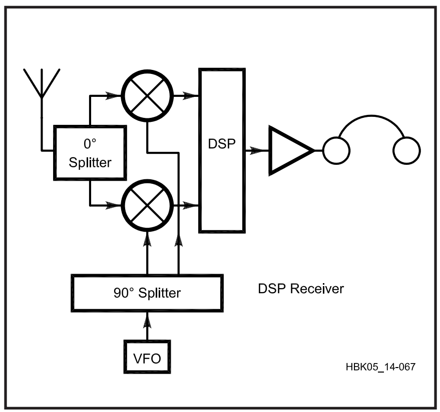


Fig 14.65—The simplified block diagram of a receiver using a DSP detector; see text.

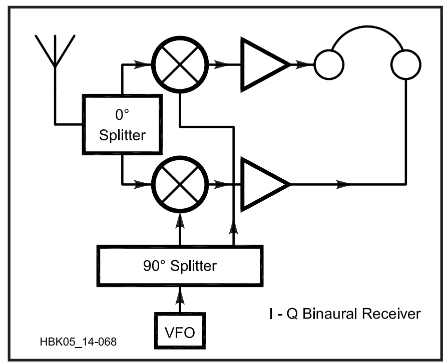


Fig 14.66—The block diagram of a binaural I-Q receiver that allows the ear/brain combination to process the detector output, resulting in stereo-like reception.

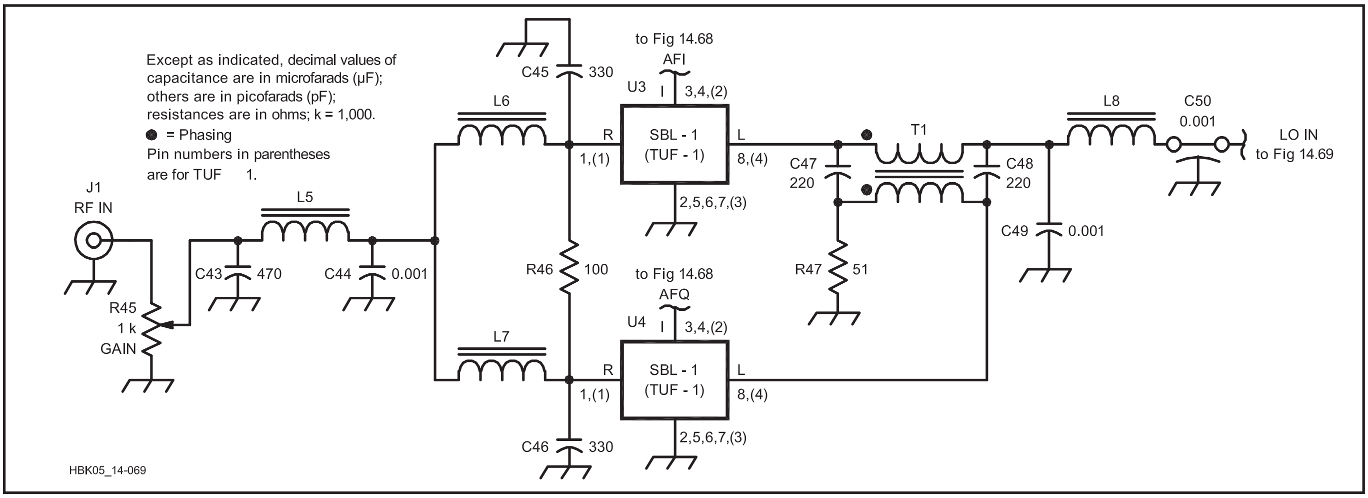


Fig 14.67—This diagram shows the front end and *I* and *Q* demodulators of the Binaural Weekender receiver. Unless otherwise specified, resistors are 1/4 W, 5% tolerance carbon-composition or film units. Equivalent parts can be substituted. Pin connections for the SBL-1 and TUF-1 mixers at U3 and U4 are shown; the TUF-1 pin numbers are in parentheses. A kit is available (see Note 1). Parts are available from several distributors including Digi-Key Corp, Mouser Electronics, and Newark Electronics.

C43—470 pF disc ceramic.
C44, C49—0.001 µF metal polyester.
C45, C46—330 pF disc ceramic.
C47, C48—220 pF disc ceramic.
C50—0.001 µF feed-through capacitor.
J1—Chassis-mount female BNC connector.
L5—1.6 µH, 24 turns #28 enameled wire on T-30-6 powdered-iron core.
L6, L7—1.3 µH, 21 turns #28 enameled wire on T-30-6 powdered-iron core.
L8—350 nH, 11 turns #28 enameled wire on T-30-6 powdered-iron core.
R45—1 kΩ panel-mount pot.
T1—17 bifilar turns #28 enameled wire on T-30-6 powdered-iron core.
U3, U4—Mini-Circuits SBL-1 or TUF-1 mixer.

plication. An analog signal processor consisting of a pair of audio phase-shift networks and a summer could be used to reject one sideband. In a DSP receiver, the *I* and *Q* baseband signals are digitized and the resulting sets of numbers are phase-shifted and added.

The human brain is a good processor for information presented in pairs. We have two eyes and two ears. Generally speaking, we prefer to observe with both eyes open, and listen with both ears. This gives us depth of field and three-dimensional hearing that allows us to sort out the environment around us. The ear/brain combination can be used to process the output of the I-Q detectors as shown in **Fig 14.66**.

The sound of CW signals on a binaural I-Q receiver is like listening to a stereo recording made with two identical microphones spaced about six inches apart. The same information is present on each channel, but the *relative phase* provides a stereo effect that is perceived as three-dimensional space. Signals on different sidebands—and at different frequencies—appear to originate at different points in space. Because SSB signals are composed of many audio frequencies, they sound a little spread in the perceived three-dimensional sound space. This spreading also occurs with most sounds encountered in nature, and is pleasant to hear.

To keep the receiver as simple as possible, a single-band direct-conversion (D-C) approach is used. A crystal-controlled converter can be added for operation on other bands, changing the receiver to a single-conversion superhet. Alternatively, the binaural I-Q detector can be used in a conventional superhet, with a tunable first converter and fixed-frequency BFO. If proper receiver design rules are followed, there is no advantage to either design over the other.

THE RECEIVER

Figs 14.67, **14.68** and **14.69** show the complete receiver schematic. In Fig 14.67, signals from the antenna are connected directly to a 1-kΩ GAIN pot on the front panel. J1 is a BNC antenna connector, popular with QRP builders. Adjusting the gain before splitting the signal path avoids the need for a two-gang volume control, and eliminates having to use separate RF and AF-gain adjustments. This volume-control arrangement leaves the "stereo background noise" constant and varies the signal-to-noise ratio. The overall gain is selected so that the volume is all the way up when the band is quiet. Resistor values R9 and R31 may be changed to modify the overall gain if required. After the volume control, the signal is split with a Wilkinson divider and connected to two SBL-1 diode-ring mixers. (The TUF-1 is a better mixer choice, but I had more SBL-1s in my junk box.) The VFO signal is fed to the two mixers through a quadrature hybrid, described by Reed Fisher.[2] All of the circuitry under the chassis is broadband, and there are *no* tuning adjustments.

The audio-amplifier design of Fig 14.68 is derived from that used in the R1 High-Performance Direct-Conversion Receiver,[3] with appropriate simplifications. The R1 high-power audio output is not needed to drive headphones, the low-pass filter is eliminated, and the diplexer has fewer components. Distortion performance is not compromised—well over 60 dB of in-band two-tone dynamic range is available. The original article, and the additional notes in Technical Correspondence for February 1996,[4] describe the audio-amplifier chain in detail.

THE VFO

Fig 14.69 is the schematic of the receiver VFO, a JFET Hartley oscillator with a JFET buffer amplifier. Components for the VFO tuned circuit are chosen for linear tuning from 7.0 to 7.3 MHz with the available junk-box variable capacitor. Setting up the VFO is best done with a frequency counter, receiver and oscilloscope. The frequency counter makes it easy to select the parallel NP0 capacitors and

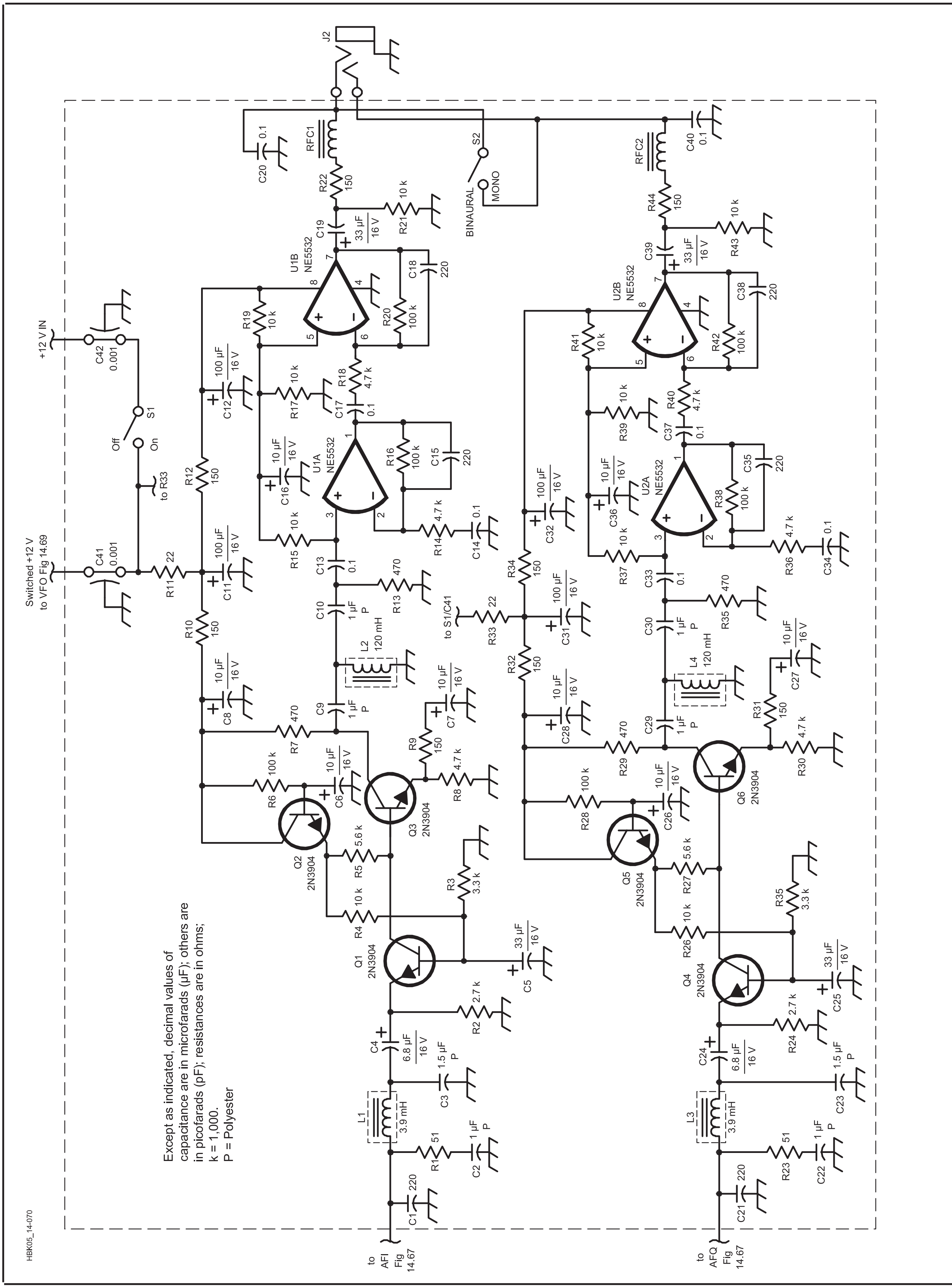

Except as indicated, decimal values of capacitance are in microfarads (µF); others are in picofarads (pF); resistances are in ohms; k = 1,000. P = Polyester
HBK05_14-070
to AFI Fig 14.67
to AFQ Fig 14.67
Switched +12 V to VFO Fig 14.69
+12 V IN
to R33
to S1/C41
Off
On
S1
S2
BINAURAL
MONO
J2
U1A NE5532
U1B NE5532
U2A NE5532
U2B NE5532
Q1 2N3904
Q2 2N3904
Q3 2N3904
Q4 2N3904
Q5 2N3904
Q6 2N3904
RFC1
RFC2

Fig 14.68—This diagram shows the receiver audio-amplifier design.
C1, C15, C18, C21, C35, C38—220 pF disc ceramic.
C2, C9, C10, C22, C29, C30—1 µF metal polyester (Panasonic ECQ-E(F) series).
C3, C23—1.5 µF metal polyester (Panasonic ECQ-E(F) series).
C4, C24—6.8 µF, 16 V electrolytic (Panasonic KA series).
C5, C19, C25, C39—33 µF, 16 V electrolytic (Panasonic KA series).
C6, C7, C8, C16, C26, C27, C28, C36—10 µF, 16 V electrolytic (Panasonic KA series).
C11, C12, C31, C32—100 µF, 16 V electrolytic (Panasonic KA series).
C13, C14, C17, C20, C33, C34, C37, C40—0.1 µF metal polyester (Panasonic V series).
C41, C42, C50—0.001 µF feed-through capacitor.
J2—1/8-inch stereo phone jack.
L1, L3—3.9 mH Toko 10RB shielded inductor.
L2, L4—120 mH Toko 10RB shielded inductor.
Q1 through Q6—2N3904.
RFC1, RFC2—10 turns #28 enameled wire on Amidon ferrite bead FB 43-2401 (six-hole bead).
S1, S2—SPST toggle switch.
U1, U2—NE5532 dual low-noise high-output op amp.

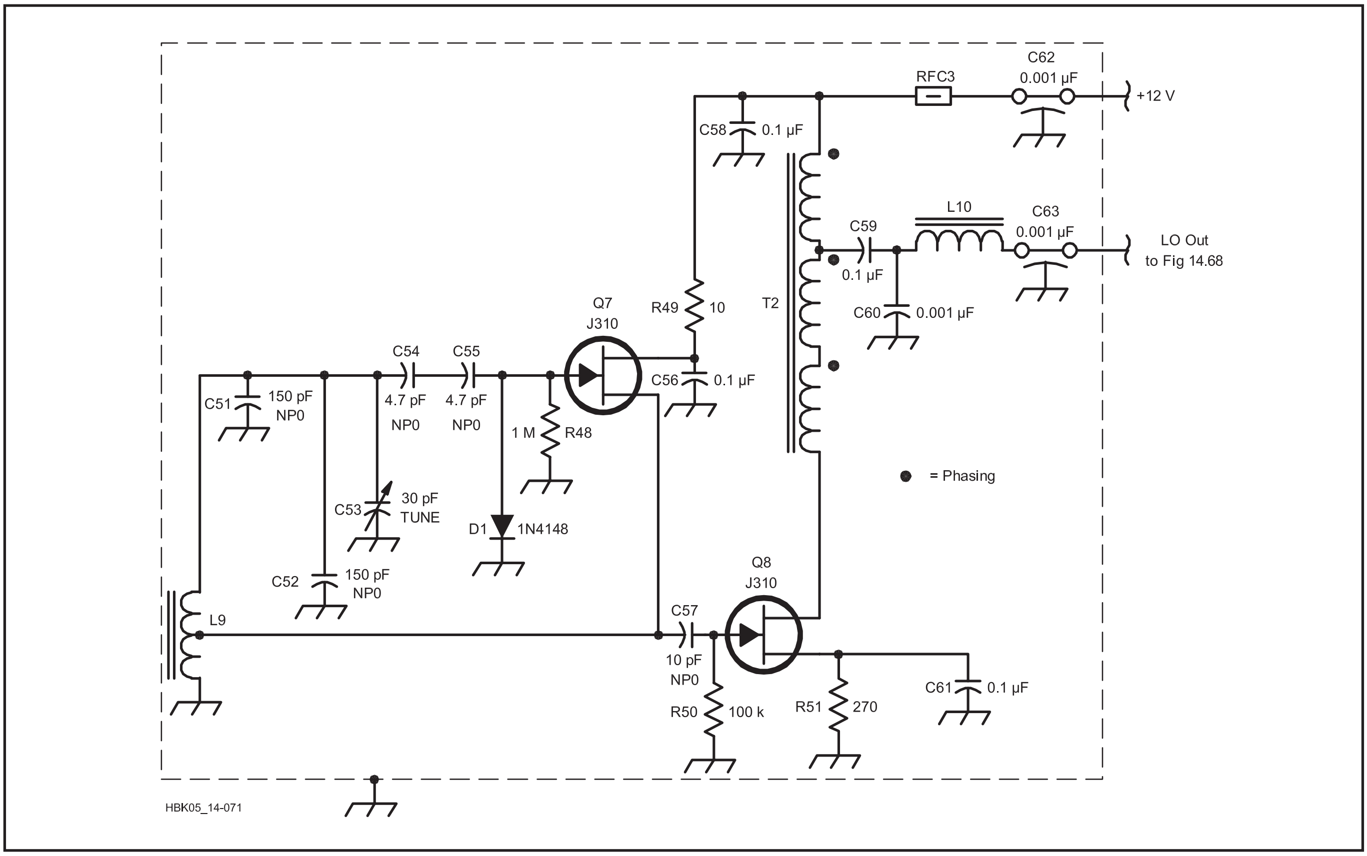


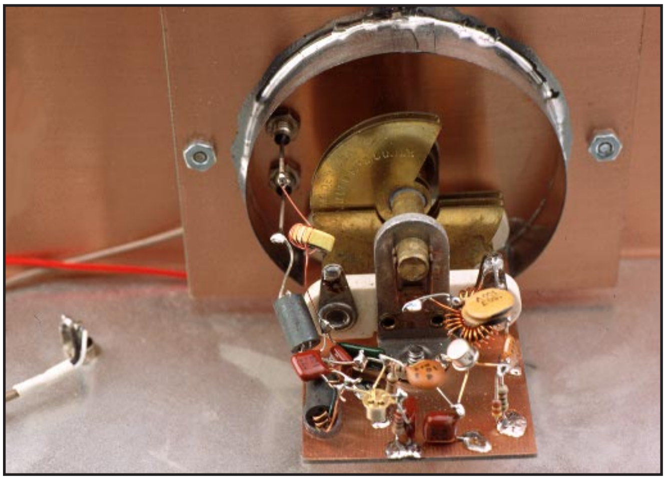
Fig 14.69—The diagram shows the prototype binaural receiver's VFO. The LO output is +10 dBm. This simple VFO works exceptionally well, but must be completely shielded for good D-C receiver performance. A receiver with an open PC-board VFO will work better if the variable oscillator is not running on the received frequency. As noted elsewhere, the kit version of the receiver uses a different VFO.

C51, C52—150 pF, NP0 disc ceramic.
C53—30 pF air-dielectric variable.
C54, C55—4.7 pF NP0 disc ceramic.
C56, C57, C59, C61—0.1 µF metal polyester (Panasonic V series).
C57—10 pF NP0 disc ceramic.
C60—0.001 pF metal polyester.
C62, C63—0.001 µF feedthrough capacitor.
D1—1N4148.
L9—1.5 µH, 22 turns #22 enameled wire on T-37-6 powdered-iron core; tap 5 turns from ground end.
L10—350 nH, 11 turns #28 on T-30-6 powdered-iron core.
Q7, Q8—J310 (U310 used in prototype).
RFC3—10 turns #28 enameled wire on Amidon ferrite bead FB 43-2401 (six-hole bead used in prototype).
T2—10 trifilar turns #28 enameled wire on Amidon ferrite bead FB 43-2401 (six-hole bead used in prototype).

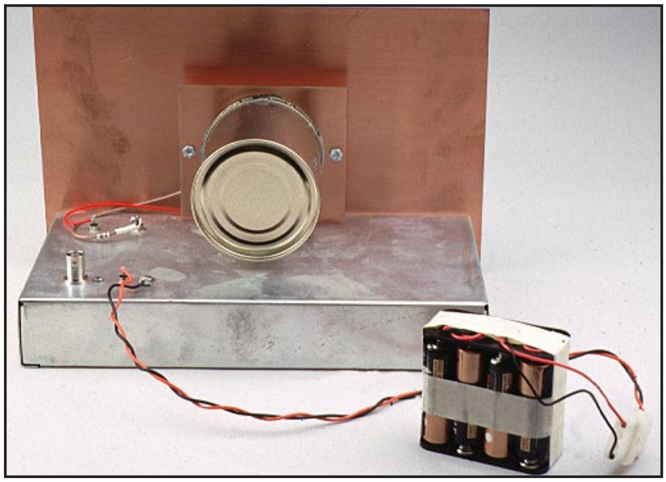
squeezing and spreading the wire turns on L1 achieves the desired tuning range. After the tuning range is set, listen to the VFO signal with a receiver to make sure the VFO tunes smoothly and has a good note. Interrupt the power to hear its start-up chirp. The signal may sound ratty with the frequency counter on, so turn it off. The VFO is one area where craftsmanship pays off. Solid construction, a self-aligning variable-capacitor mounting, complete RF and air shielding and good capacitor bearings all contribute to a receiver that is a joy to tune.

Both connections to the VFO compartment are made with feed-through capacitors. The power supply connection is self-explanatory, but passing RF through a feed-through capacitor (at LO Out) may seem a bit unusual. Electrically, the capacitor is one element of a low-pass pi network. Using feed-through capacitors keeps local VHF signals (high-powered FM broadcast and TV signals near my location) out of the VFO compartment. A second pi network feeds the VFO signal to the detector circuit below the chassis. The use of VHF construction techniques in a 40-meter receiver may seem like overkill, but the present KK7B location is line-of-sight to broadcast towers serving the Portland, Oregon area. Using commercial HF gear with conventional bypassing under these circumstances provided disappointing results.

Fig 14.64 shows the prototype receiver front panel. Receiver controls are simple and intuitive. The ear/brain adjusts so naturally to binaural listening that I added a BINAURAL/MONO switch to provide a quick reminder of how signals sound on a conventional receiver. The switch acts



(A)



(B)

Fig 14.70—A shows a close-up of the VFO. The simple VFO used in the prototype works exceptionally well, but must be completely shielded for good D-C receiver performance. B shows how an empty mushroom can can live again as a VFO shield in the prototype receiver.

What Do You Hear?

Even the earliest solid-state direct-conversion (D-C) receivers had a *presence* or *clarity* that is rarely duplicated in more elaborate receivers. Many of us remember the first time we heard this crispness in a “homebrewed” D-C receiver. As we try to “enhance” our rigs through the addition of IF filters and other “features,” we still hope that the result will be as clean as that first D-C receiver.

This binaural D-C receiver is such an experience—but even better. The binaural processing supplies the ears with additional information without compromising what was already there, enhancing the presence.

As you tune through a CW signal on a quiet band (best done with your eyes closed while sitting in a solid chair), a centered signal enters, but moves to the left background, undergoes circular motions at the back of your head as you tune through zero beat, repeats the previous gyrations on the right side, fades to the right background, and finally drops away in the center. Multiple signals within the receiver passband are distributed throughout this perceived space. With training, concentration on one signal allows it to be copied among the many. An SSB signal seems to occupy parts of the space, left and right, with clarity when properly tuned, leaving others vacant. Static crashes and white noise appear distributed throughout the entire space without well defined position. Receiver noise, although present, has no perceived position.

It’s vital that this receiver include a front-panel switch to shift between binaural and monaural output. Although useful during the learning process, it becomes indispensable for the demonstrations that you will want to do. I used the switch to set up my son, Roger, KA7EXM, for the experience. We entered the shack and I handed him the headphones. He put one phone to just one ear, but I told him that he had to use both, that it would not work with just one. He put the phones on his head, casually tuned the receiver through the 40-meter CW band, removed the phones and commented, “Well, it sounds just like a direct-conversion receiver: A good one, but still just a direct-conversion receiver.” I smiled and asked him to put the headphones on again. As I flipped the switch to the binaural position his hand reached out, seeking the support of the workbench. His facial expression became more serious. He eased into the chair and began tuning the receiver, very slowly at first. After a minute he took the headphones off, but remained speechless for a while—an unusual condition for Roger. Finally, he commented, “Wow! The appliance guys have never heard that!”

A builder of the Binaural Weekender should prepare for some truly unusual experiences.—*Wes Hayward, W7ZOI* much like the STEREO-MONO switch on an FM broadcast receiver—given the choice, it always ends up in the STEREO position!

The author uses a pair of Koss SG-65 headphones with his receiver. They are not necessary, but have some useful features. First, at about $32, they are relatively inexpensive. Second, they have relatively high-impedance drivers, (90 Ω) so they can be driven at reasonable volume directly from an op amp. Finally, they make an attempt at low distortion. Other headphones in the same price bracket are acceptable, but some have much lower impedance and won't provide a very loud audio signal using the component values given in the schematic. Those $2.95 bubble-packed, throw-away headphones are not a good choice! Audiophile headphones are fine, but don't really belong on an experimenter's bench. A stray clip-lead brushing across the wrong wire in the circuit can instantly burn out a driver and seriously ruin your day.

BUILDING A BINAURAL WEEKENDER

A few construction details are generally important, while others were determined by the components that happened to be in my junk box. The big reduction drive is delightful to use, but doesn't contribute to electrical performance. I purchased it at a radio flea market. The steel chassis provides a significant reduction in magnetic hum pickup, something that can be a problem if the receiver is operated near a power transformer. (Steel chassis are available from parts houses that cater to audio experimenters.) The VFO mounting and mushroom-can shield shown in **Fig 14.70** are a simple way to eliminate mechanical backlash, keep radiated VFO energy off the antenna, prevent hand capacitance from shifting the tuning, and reduce VFO drift caused by air currents.

Experienced builders can duplicate this receiver simply using the schematic and construction techniques described here. Unlike a phasing receiver, there is no need to precisely duplicate the exact amplitudes and phases between the two channels. The ear/brain combination is the ultimate adaptive processor, and it quickly learns to focus on a desired signal and ignore interference. Small errors in phase and amplitude balance are heard as slight shifts in a signal's position. Standard-tolerance components may be used throughout.

One note about the kit version: A very good VFO can be built on an open PC board if the variable oscillator is not running on the desired output frequency. The Kanga kit VFO runs at one-half the desired frequency, and is followed by a balanced frequency doubler and driver amplifier.

OTHER EXPERIMENTS

My earliest experiments with binaural detectors feeding stereo audio amplifiers were done in 1979, using two antennas. The technique works very well, but requires two antennas either physically spaced some distance apart, or of different polarization. Listening to the OSCAR 13 satellite on a binaural receiver with cross-polarized Yagis was an unsettling experience. The need for two antennas is a liability—these days most of us struggle to put up one. A number of experiments have also been done with binaural independent sideband (ISB) reception. These are profoundly interesting for AM broadcast reception, and could be used for amateur AM or DSB reception using a Costas Loop for carrier recovery. Binaural ISB detection of shortwave AM broadcasting can be analyzed as a form of spread spectrum with the ear/brain combination serving the despreading function, or as a form of frequency diversity, with the ear/brain as an optimal combiner.

The binaural techniques described here are analogous to binocular vision: They present the same information to each ear, but from a slightly different angle. This provides a very natural sound environment that the brain interprets as three-dimensional space. There are other "binaural" techniques that involve the use of different filter responses for the right and left ears. My experiments with different filter responses for the left and right ears have not been particularly interesting, and I have not pursued them.

SUMMARY

This little receiver is a joy to tune around the band. It is a serious *listening* receiver, and allows digging for weak signals in a whole new way. Digging for weak signals in a three-dimensional sound field is sometimes referred to as the "cocktail party effect." It is difficult to quantify the performance of a binaural receiver, because the final signal processing occurs in the brain of the listener—you. The experimental literature of psycho-acoustics suggests that the ear/brain combination provides a signal-to-noise advantage of approximately 3 dB when listening to speech or a single tone in the presence of uncorrelated binaural noise. The amount of additional noise in the opposite sideband is also 3 dB, so it appears that the binaural I-Q detector breaks even. In some applications, such as UHF weak-signal work, the binaural I-Q detector may have an advantage, as it permits listening to a larger slice of the band without a noise penalty. In other situations, such as CW sweepstakes, the "cocktail party" may get entirely out of hand. Binoculars and telescopes both have their place.

Notes

1 The complete kit version, available from Kanga US, uses a different VFO circuit than the one shown here. The kit VFO runs at one-half the desired output frequency, and is followed by a balanced frequency doubler and driver amplifier.

Steel chassis such as the Hammond 1441-12 (2 × 7 × 5 inches [HWD]) with 1431-12 bottom plate and the Hammond 1441-14 (2 × 9 × 5 inches [HWD]) with 143-14 bottom plate are suitable enclosures. These chassis and bottom plates are not available in single quantities directly from Hammond, but are available from Allied Electronics and Newark Electronics.

2 Reed Fisher, W2CQH, "Twisted-Wire Quadrature Hybrid Directional Couplers," *QST*, Jan 1978, pp 21-23. See also IEEE Transactions MTT, Vol MTT-21, No. 5, May 1973, pp 355-357.

3 Rick Campbell, KK7B, "High-Performance Direct-Conversion Receivers," *QST*, Aug 1992, pp 19-28.

4 Rick Campbell, KK7B, "High-Performance, Single-Signal Direct-Conversion Receivers," *QST*, Jan 1993, pp 32-40. See also Feedback, *QST*, Apr 1993, p 75.

References

Campbell, Rick, KK7B, "Direct Conversion Receiver Noise Figure," Technical Correspondence, *QST*, Feb 1996, pp 82-85.

Campbell, Richard L., "Adaptive Array with Binaural Processor," *Proceedings of the IEEE Antennas and Propagation Society International Symposium,* Philadelphia, PA, June 1986, pp 953-956.

Campbell, Rick, KK7B, "Binaural Presentation of SSB and CW Signals Received on a Pair of Antennas," *Proceedings of the 18th Annual Conference of the Central States* VHF *Society,* Cedar Rapids, IA, July 1984, pp 27-33.

A Dual Band Low Noise Amplifier for 2 Meters and 70 Centimeters

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This article describes constructing a dual band low noise amplifier (LNA) for use on 2 meters and 70 centimeters. Bypass capability is included to permit transmitting on either band while listening to the downlink signal on the opposite band of a VHF/UHF satellite.

LNAs are needed when you have a long run of coax between the antenna and receiver. LNAs must be mounted as close to the antenna as is practical because, once you've lost the signal in your coax, there is no way to get it back. LNAs mounted at the antenna amplify the weak signals before they are attenuated in coax.

Total project cost is around $172, depending on what you can rummage from your or your friends' junk box and your hamfest/eBay shopping prowess. The design includes a weatherproof enclosure. The noise figure is around .5 dB on 2 meters and .8 dB on 70 centimeters with 20-25 dB gain. The power handling capability is limited by the relays and the coax. The relay manufacturer rates its units at 50 W up to 1 GHz. The coax used here withstood 50 W from a satellite transceiver.

A low-pass filter on the 2-meter LNA rejects 70-centimeter signals from the transmitter and a high-pass filter on the 70-centimeter LNA rejects 2-meter signals from the transmitter. This prevents overloading the LNAs while listening on one band and transmitting on the other (i.e., when listening to your uplink signal on the satellite downlink).

Performance

While I have not yet measured the noise figure on either band, I perceived a slight improvement with the LNA on SSB but a very noticeable improvement on FM. I performed testing for desense/overload on each band by listening to a very weak signal on a dipole on one band while transmitting 50 W on the opposite band with a dipole. The antennas were eight feet apart aligned parallel to each other. There was no degradation in signal.

The measured filter losses were:

- 2 meter low pass: -.1 dB @ 144 MHz/-70 dB @ 436 MHz
- 70 centimeter high pass: .4 dB @ 436 MHz/- 52 dB @ 144 MHz.

Parts

While at a yard sale, I noticed a home satellite TV multi-dish switch box for 50 cents (Figure 1). I had to have it but with no idea how I would use it. While my satellite TV multi-dish switch box came from a yard sale, they also are available on eBay. I bought my second one at a hamfest for $2. See the pictures showing various stages of disassembly.

The LNAs are available on eBay for $18.95 each ($2.45 shipping) from Chuck Steer, WA3IAC; you will need two. Download the datasheet for the low noise IC (a MiniCircuits PGA-103) to see the lead layout. You can also examine the impressive specs for yourself at **www.minicircuits.com**.

The four relays are available on eBay for $30 or less each. Make sure you get the type you need, latching or non-latching (see discussion below under "Issues and Modifications"). Latched relays need only a short pulse of voltage to switch position and the pole is always connected to one port or the other; non-latched relays must have voltage applied full time to one coil or the other.

The high and low pass filters use silver mica capacitors and air wound coils. Check hamfests and your ham friends' junk boxes for the capacitors and wire.

The N connectors are the hardest part to find. The thread that screws into the chassis is the same as that found on a type F connector. I found two in my junk box but obtained three more from old satellite TV downconverters and surplus LNA/LNBs. They are from the "old" C band satellite TV days when nine-foot dishes were the norm. See Figure 2.

I purchased the SMA cables on eBay for $1.45 each. You will need eight. They are 6" (15 cm) long with male connectors (center pin) on both ends. I had some doubts about the quality of the units I bought on eBay since they are not made in the U.S., so I tore

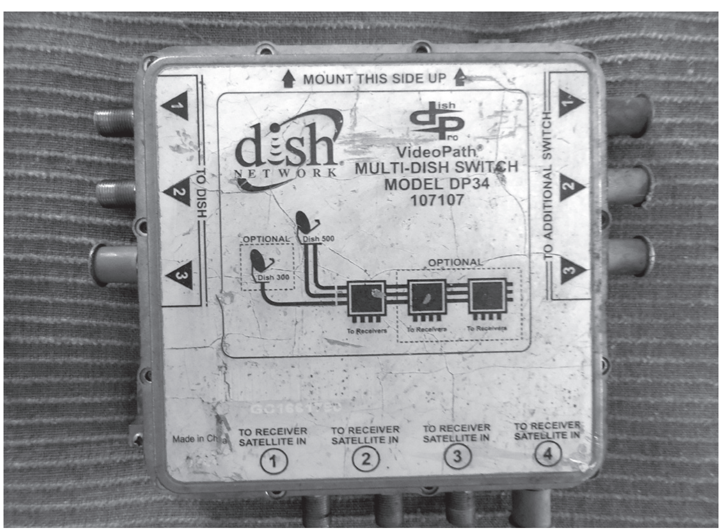


Figure 1 - Original chassis

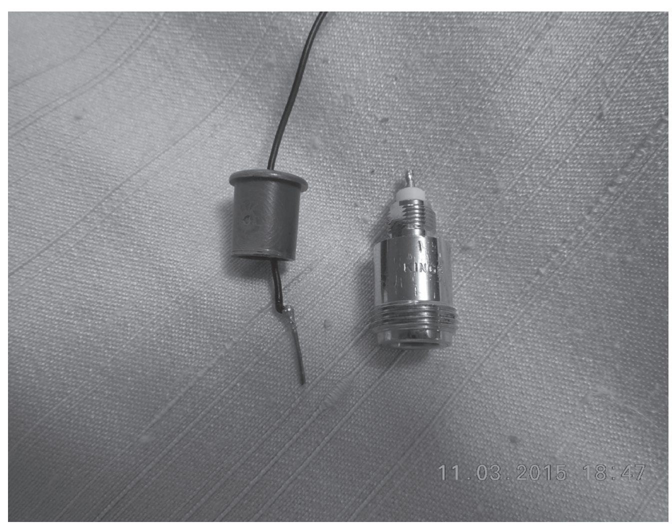


Figure 2 - N connector and plastic cap with wire connection to F connector

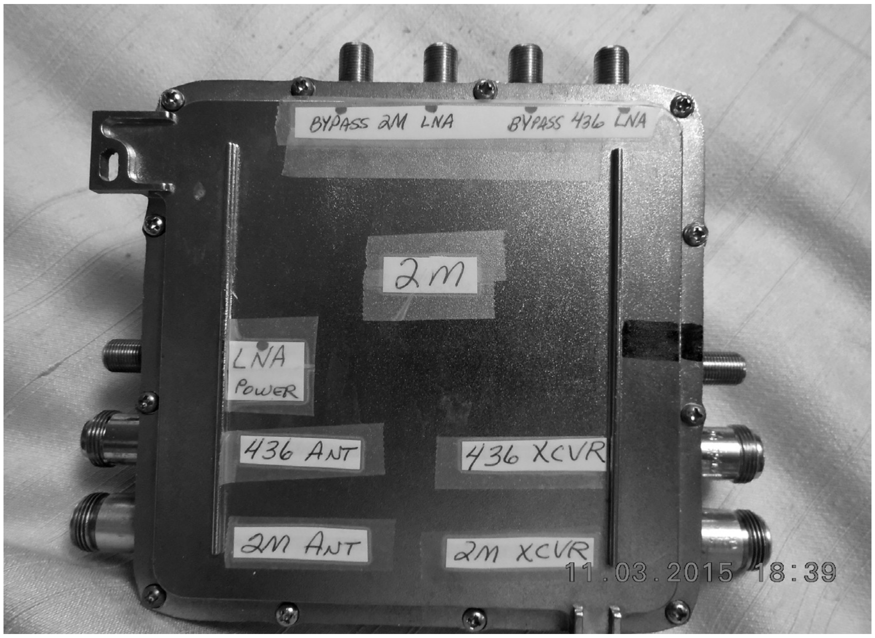


Figure 3 - Labels showing connections

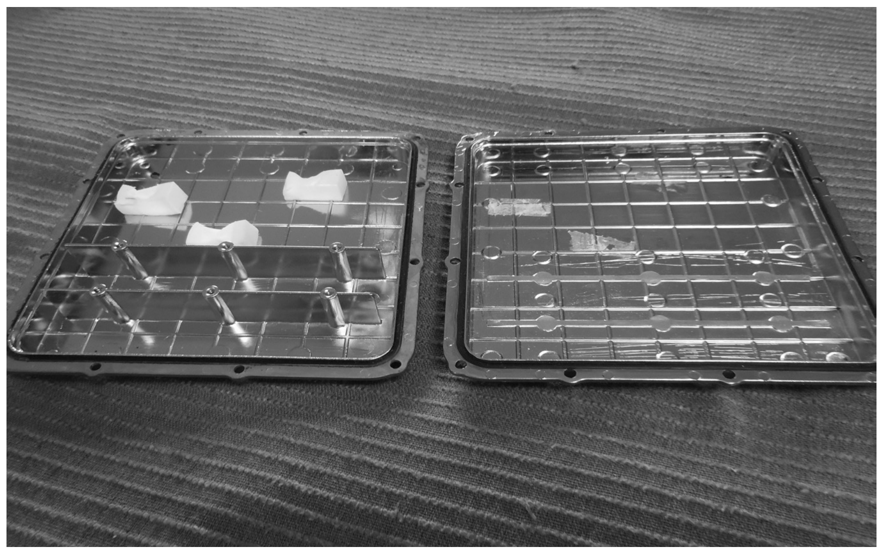


Figure 4 - Metal webs before and after

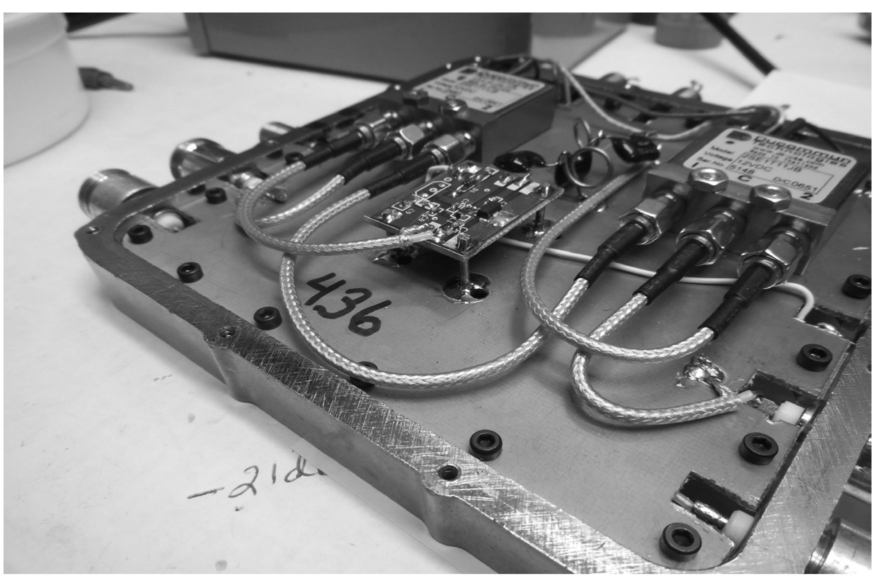


Figure 5 - LNA board mounted on wire standoffs

apart one end to verify they were properly assembled. Only two cables are used as-is. The others need one connector removed, the cable cut shorter, and tinned for soldering to the LNA, filters and N connectors.

Construction

Before removing the screws and covers, mark both covers and the center part of the chassis to show which cover goes on which side. I didn't mark them and spent a lot of time figuring out which cover goes on each side. I suggest a Dremel tool or etcher to make the marks. See Figure 3.

After marking, remove all the screws. The lids will not come free so you will have to pry them loose. Be careful not to damage the gasket around the edge. Once the covers are off, you will see the white blocks of foam that held them on so securely (Figure 4).

Unsolder all the connectors; de-soldering braid works well. Do not damage the connector center pins. One of the covers has metal webs that go deep into the chassis to shield each stage from the others. You will have to remove this webbing. Also, do not bend the webs back and forth thinking you can just break them off. I learned this the hard way; the webs in mine broke off but also made a big hole in the cover (Figure 4).

You will need to remove four of the connectors and replace them with type N connectors. They are threaded in very tightly and glued. To remove them, secure the chassis to a bench with C-clamps and then use a 7/16" socket to remove them. Use a small wire brush to remove the glue left in the hole so you can easily thread in the new N connectors and provide a good ground connection.

The copper plate that holds all parts is double-sided glass epoxy printed circuit board. Start with a piece 5" x 5" x .0625". Then slowly file and curve the corners of the plate until it fits snuggly inside the chassis. You may have to file the straight edges slightly, but take your time and make it fit right up against all four inside edges of the chassis.

Next put the plate inside the chassis and mark all screw holes very accurately. Remove the plate and punch a mark at each location. Drill 1/8" holes and lightly deburr them. I replaced the Phillips-head metal tappingscrews with 6 mm length x 2.5 mm thread Allen Head machine screws.

Notches are needed where each coax connector's center pin extends inside. I used a nibbling tool for this. Alternatively, a file can be used. The plate is located on the same side of the interior rail as the deeper cover so that neither cover touches the relays, coils or other tall parts.

The LNA kits require that you solder SMD parts. Do not mount the SMA connectors because you will be wiring directly to the board input/output tracks. Some of the surface mount parts are very small (see the picture on eBay). Use a low power soldering iron and toothpick to hold them in place. If you don't feel comfortable, ask a friend who works with SMD parts.

Use bright lights and a magnifying lens and take your time when soldering. One slip and the part will either end up on the floor or soldered out of place. (Don't ask me how I know this, but I will admit to ordering a third LNA kit.) I mounted the regulators below the LNA board and bent them toward the board to get a low profile on the top of the board. It's okay if the tab of the regulator touches the copper plate, as the tab is ground.

Make sure you haven't reversed the leads of the regulator! The power to both LNAs is connected to a single existing F connector (Figure 3). The LNA boards mount directly to the plate on pieces of #14 (.064") copper wire that are soldered to both sides of the plate (Figure 5).

Use the bare LNA circuit board as a template to locate the holes for the support wires. At first I tried to mount the LNAs using screws and spacers but could not get the holes aligned. It turns out the #14 wire supports provide a better ground connection than screws and spacers. Check each LNA before mounting it. Each should draw approximately 100 mA. If you have access to test equipment (a signal generator and something to check the input and output levels), make sure the gain is approximately 20-25 dB. Clean them with flux remover after mounting and after attaching the filters and input/output leads.

The high and low pass filters must be constructed with the components oriented as shown. The coils must be at right angles to each other to minimize coupling between them. The capacitors should have their leads cut as short as possible. See Figures 6 & 7.

When attaching the SMA connectors to the relays, attach the center one first. You cannot reach the center connector with a wrench if either of the outside connectors is in place. Use a 5/16" open-end wrench to gently tighten them. All four relays are mounted to the plate using machine screws that go through the plate.

The voltages to switch the relay positions and supply power to the LNAs pass through the remaining F connectors. To provide a waterproof entry inside, use plastic covers for unused F connectors. It is likely that your chassis will have a few plastic covers. Punch a small hole in the plastic cap and pass a wire to a stiff copper wire that makes contact with the connector center connection (Figure 2). Alternatively, you can pass all wires through a hole that is available after you remove one F connector and epoxy the wires in place for a watertight seal.

Issues and Modifications

The noise figure (NF) isn't record setting because a .8 dB NF on 436 MHz isn't great. However, it does improve reception significantly on FM and some on SSB in my satellite radio (an Icom 821, with sensitivity specs equivalent to the newest satellite radios). The total NF is the loss in the filter plus the NF of the LNA itself. I'm working on a filter for 436 MHz with lower loss but, at this point, haven't been able to get it lower than .4 dB. Perhaps with some experimenting/juggling of coil/capacitor values, you can get the NF lower. Even so, these LNAs will help you hear much better if you have a long coax run with significant loss and will definitely help when operating FM.

You may want to use latching relays to reduce the internal heating that would come from non-latching relays. Or you may prefer to use non-latching relays so that the interior always stays warm which keeps the inside dry. When I built the first LNA assembly and saw that it worked fine, I proceeded to order two more relays but accidentally ordered non-latching units, so I have both types in my unit. There are pluses and minuses either way. If moisture is a serious problem where you live, I recommend the non-latching type to keep it warm but dry inside.

continued on page 22 ...

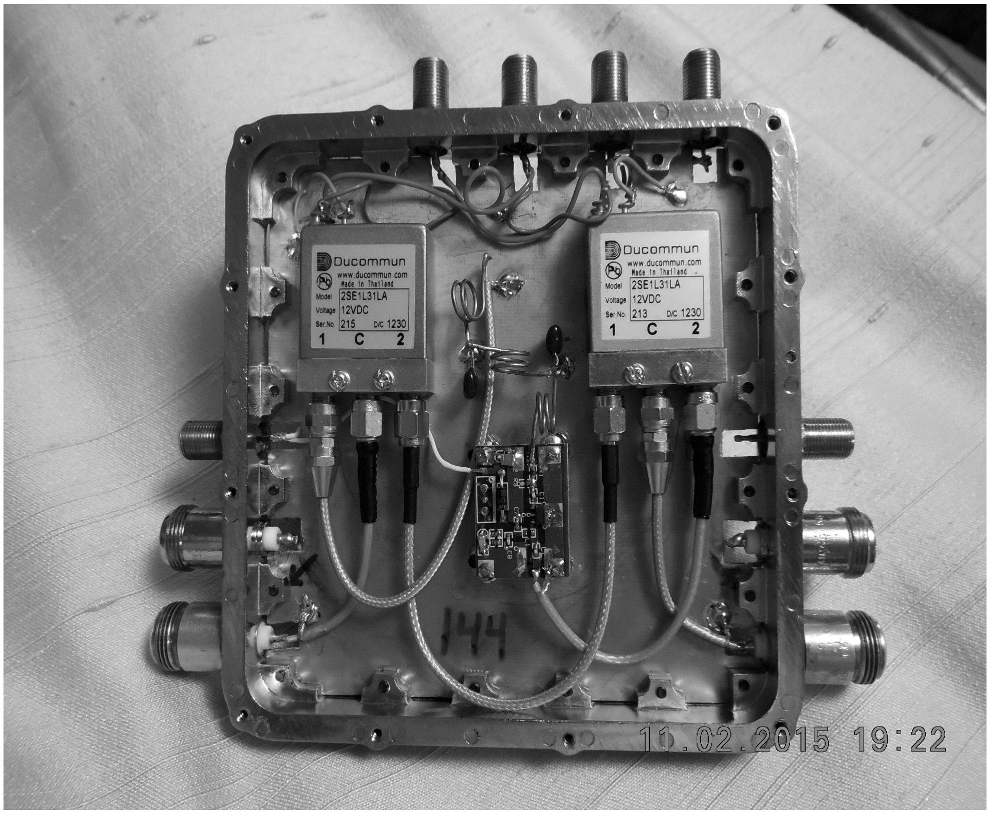


Figure 6 - The 436 MHz LNA complete

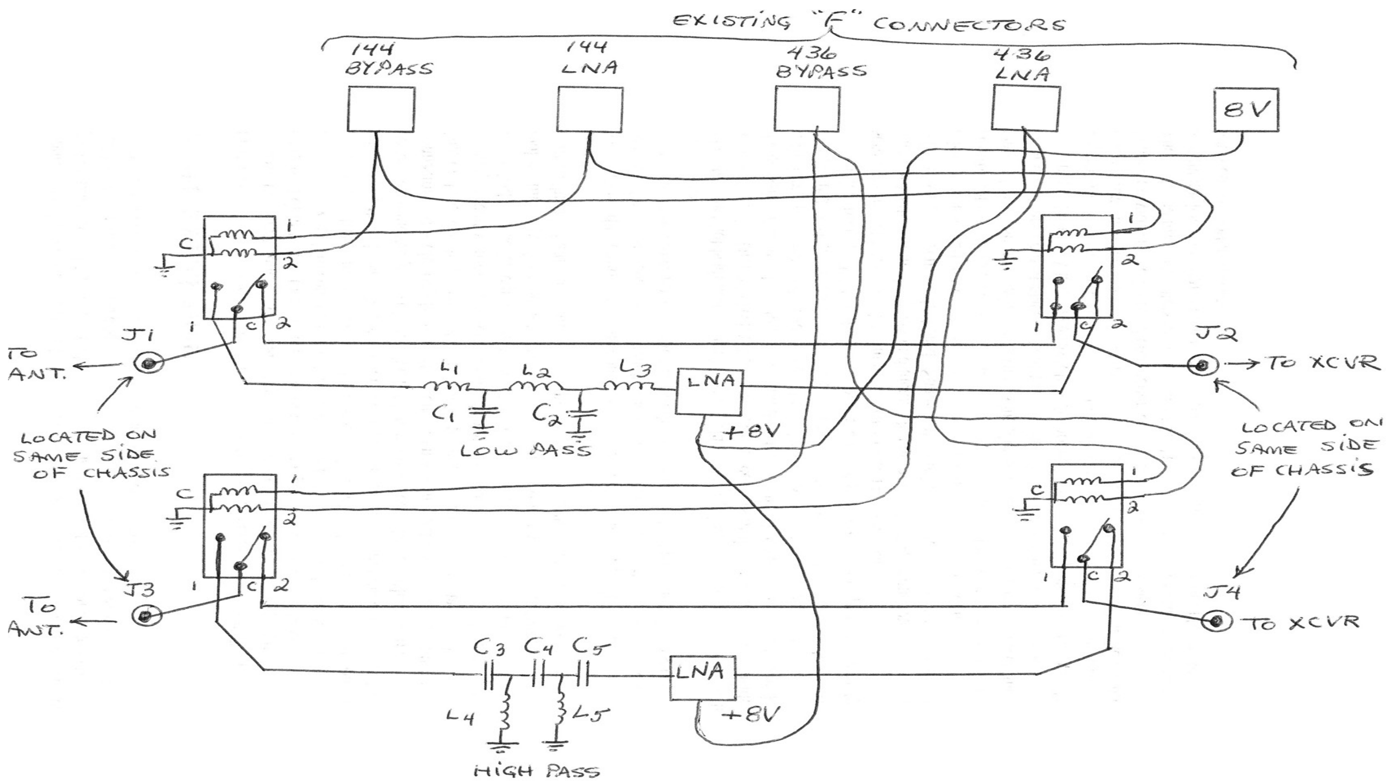


Figure 8 - Dual LNA schematic

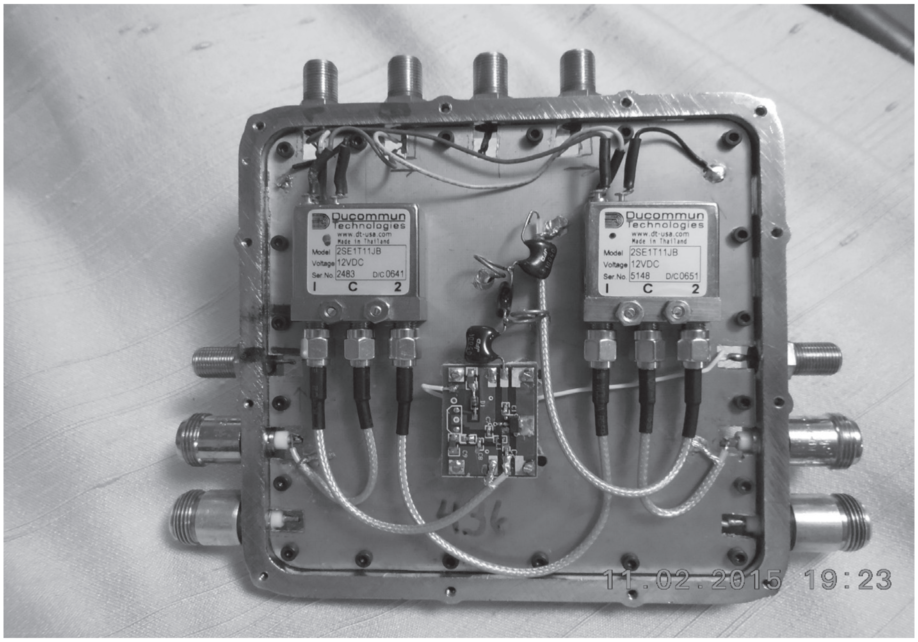


Figure 7 - The 144 MHz LNA complete

Using these LNAs for terrestrial operations requires a sequencer that switches the state of the relays before transmitting. For satellite operation, a simple toggle switch can be used to set one LNA to bypass (uplink) and the other to LNA (downlink).

A suitable filter for experimenters is detailed at **www.zs6wr.co.za/ham-mag/Diplexer.pdf**. It is inexpensive to make, physically small, has reasonable attenuation at the reject frequency and, most importantly, very low attenuation at the pass frequency (.15 dB).

The instructions for the LNA specify 12 V for the LNA, but I found the regulator got too hot to touch because of the current drawn by the regulator and required voltage drop. I suggest using 8-9 V instead.

When wiring the cables to the relays and N connectors, make very sure that you have the output and input to the LNA connected properly. The layout on one side is the mirror image of the other side if flipped over. Both antenna connectors should be on one side and both rig connectors should be on the other side. Check and double-check where the wiring goes before you solder and before you connect the coax cables and relay wiring! See Figures 3 & 8. Label each connector to show which signal goes where.

Parts List

- DISH Network Video Path Multi-Dish Switch Model DP34 107107, eBay, buy two if you see them at a hamfest or yard sale for a low price (spares are good to have just in case….)
- 5" x 5"x .0625" double sided glass epoxy copper clad board
- 4 type N connectors with thread same as F connectors, J1, J2, J3, J4
- Four 12 VDC SMA relays, Duocomm 2SE1T11JB or 2SE1L31LA (latching or non-latching)
- Two 12 pf silver mica capacitors, C1, C2 - Digikey 338-2819-ND
- Three 5 pf silver mica capacitors, C3, C4, C5 - Digikey 338-2818-ND
- 2 LNA kits, on eBay search for "PGA103 LNA" then scroll down to an LNA from seller "chuckwa3iac"
- #14 copper wire, 8 inches long, cut to 2 inch lengths
- 20 metric screws, Allen Head - 6 mm length x 2.5 mm thread
- Four 4-40 x 1-1/8" machine screws, four 4-40 nuts
- Coil dimensions, #18 tinned wire, 12"
- L1, L2, L3 - 2M low pass: 2 turns, 5/16" ID, spaced wire diameter
- L4, L5 - 436 high pass: 1 turn, 5/16" ID, spaced wire diameter

For any questions, please contact the author via email (**wa9pyh@arrl.net**).

Also see Addenda on next page

FoxTelem Version 1.02 Now Available!

Chris E. Thompson • AC2CZ
chrisethompson@gmail.com

FoxTelem Version 1.02 is now available for download. Like the last release, you can patch your installation by downloading the patch file. In this case, it is a single file to replace.

You can download it from **amsat.us/FoxTelem/**.

Everyone should upgrade to 1.02 because this readies FoxTelem for transition to the new Telemetry Server, which will be more reliable.

Additionally, this release fixes a number of issues and adds the ability to download data from the Server to view/analyze in FoxTelem. If you download data, please make sure you save it to a separate directory to your local data, otherwise you will overwrite it. Of course, frequent backups of your data minimize this risk.

The release notes are as follows:

- Fixed bug where opening the Fox-1A spacecraft menu would cause a crash
- Added horizontal and vertical lines to the graphs if button clicked
- Fixed typo on measurements tab
- Fixed a bug where UTC was not displayed for the Diagnostic tables
- Capture the string version of the STP date in ENGLISH for all users, but leave other dates in local language
- Fixed bug where TCA date could be null and a SERIOUS error was reported
- Fixed issue where the tabs were always refreshed when the spacecraft menu closed
- Fixed bug where UTC date was sometimes wrong on the spacecraft T0 panel
- Ready FoxTelem for sending server data via TCP
- Support downloading server data

dedicated efforts to support AMSAT were instrumental in helping Martha to keep the office up and running, ensuring that computers, printers, fax machines, postage meters, modems, credit card machines, and other devices were working as intended, thereby allowing Martha to complete the tasks that she needed to do. Without Bob's efforts, Martha's effectiveness would have been significantly diminished.

Most noteworthy is Bob's dedication to providing that IT and administrative support to AMSAT. Think about it: he essentially went to the AMSAT office one day every week for over 24 years -- an impressive amount of time dedicated to helping AMSAT!

Bill Hook, W3QBC, was another individual who also spent many years helping Martha at the AMSAT office. Indeed, Bob would bring Bill to the AMSAT office when Bill was no longer able to drive. Unfortunately, Bill's health continued to decline to the point where he couldn't come to the office with Bob. Now, with Bob no longer with us, and Bill unable to get to the office, Martha no longer has the two "anchors" who came every Wednesday to help out with various administrative functions.

The lesson from this is two-fold: (1) AMSAT volunteers can make a huge positive impact on our organization no matter how mundane the task -- someone has to do the administrative tasks that keep our organization going; and (2) consistent, methodical volunteerism creates huge impacts over time. Bob and Bill dedicated a couple of hours each week to help at the AMSAT office. Generally, Martha knew when both Bob and Bill would be at the office, and she ensured that they had tasks to do when they arrived. Both individuals did the tasks that allowed Martha to handle other duties. And, while Bill may be the quiet one and Bob was often the "curmudgeon," both individuals exhibited their desire to see AMSAT succeed by dedicating a set amount of time each week to come to the office and help out. Martha appreciated both the work they did as well as their presence in a "one-person office."

I ask that if you live in the Washington Metro area and have some time on your hands, please consider contacting Martha to see what you can do to assist in the work that is done at the office. She needs assistance in a variety of areas, from data entry to stuffing envelopes to taking items to the Post Office. She also could use some IT support from knowledgeable individuals. A couple of hours by AMSAT volunteers at the office will help Martha immensely in terms of both getting tasks accomplished, as well as providing an opportunity for conversation. Bob and Bill were able to dedicate a portion of their free time to helping out at the AMSAT office. We can follow their example and encourage our Washington area members to help out, as they are able.

Fox-1C/1D Launch Update

The AMSAT Engineering team is finalizing the Fox-1C and Fox-1D spacecraft as I write this column in mid-January. Spaceflight, Inc., has informed AMSAT that it now expects our flight to take place sometime between March 1 and May 31, 2016. I expect that AMSAT will be notified in the near future of a specific launch date as the delivery date for these two spacecraft is based upon the launch date. Stay tuned for further developments; we're looking forward to the launch of three Fox-1 class spacecraft in 2016.

Addenda to "Dual Band Low Noise Amplifier for 2 Meters and 70 Centimeters," Nov/Dec 2015 issue:

1. When removing the covers, remove the shallow one first. Remove all the screws on the circuit board since they hold the deep cover in place. If you attempt to pry the deep cover off first, you will bend it.

2. Remove the deep cover second.

3. The two 10 ufd chip caps on the input and output of the 7805 regulator are polarized. The + side of the capacitors has a brown line. If you hold the capacitor so you can read the value, the line will be on the left side. If installed backwards they will eventually fail and smoke.

4. If possible obtain some RF absorbing material. The ham that measured the NF noticed that a lower NF was observed if a 2" x 3" square of this material was installed above the LNA circuit board. I am trying to find a source of more to make available to all who build the project.

5. The final NF are: 144 MHz = .6 dB 436 MHz = .95 dB

SKN 2016 BEST FIST WINNERS

Ray Soifer • W2RS

Thanks to all who participated in AMSAT's Straight Key Night 2016, held in memory of Ben Stevenson, W2BXA.

The following participants each received at least one Best Fist nomination: AA5PK, WA5KBH, WA8SME, W3TMZ, W4CVV, W5PFG. Special kudos go to Glenn Miller, AA5PK, who received three.

Activity was down this year for a variety of reasons, some having to do with availability of suitable satellites and others with changes in amateur radio in general. Since this was AMSAT's 25th annual SKN, it's a good time to consider changes.

While Morse as a license qualification has gone the way of the spark gap, amateur CW activity is as popular as ever. Straight keys and "bugs," however, have found a niche primarily with the "boat anchor" crowd, and AMSAT's insistence on their use in OSCAR SKN is probably limiting participation. Similar considerations have led ARRL to broaden its annual HF event to include all forms of CW, even computer-generated. The idea is to encourage everyone to enjoy CW operation, no matter how they choose to do it.
So, in with the new: AMSAT CW Activity Day on OSCAR.

As with the old SKN, it will be a fun event, not a contest, and will run for 24 hours on January 1. All forms of CW will be welcome. Instead of best fist nominations, all participants will be encouraged to post "Soapbox" comments to AMSAT-BB.

A further announcement will be posted to ANS in December.

Book Your Cruise Early for the Space ymposium!

November 10-14, 2016

See page 23 for details!

Colin Horrabin, G3SBI

17 Denbury Ave, Stockton Heath, Warrington WA4 2BL, England: **colin.horrabin7@ntlworld.com**

A High Performance 45 MHz IF Amplifier for an Up-Conversion HF/LF Receiver

The author describes the design process for a high performance IF Amplifier.

The July/August 2013 edition of *QEX* contained an article about the design of the HF7070 up-conversion HF/LF receiver, which has a 45 MHz first IF. A key building block in this receiver was the IF amplifier, which was a 45 MHz version of the amplifier that uses four J310 junction FETs with source/gate feedback, originally designed by Bill Carver, W7AAZ, for use with down conversion receivers. At 45 MHz, it has a noise figure of 1.3 dB, a gain of 10 dB, a third order intercept point (IP3) output of 40 dBm, and a nominal 50 Ω input and output impedance. In the HF7070 *QEX* article, Table 1 showed the use of this amplifier in the signal path. The gain distribution in the IF strip was such that from a linearity point of view nothing was pushed to the limit.

In the HF7070 there is no preamplifier before the first H-Mode mixer, so it is essential that the first IF amplifier has a low noise figure and a high output IP3 in order to have a sensitive receiver that is also highly linear for close-in signals. It would have simplified the circuitry if an MMIC could have been used in place of the 4 × J310 amplifier, but at the time none were available with the performance of the 4 × J310 amplifier. A low noise figure and a high output IP3 were two conflicting requirements in the available MMICs. However recently Mini-Circuits have introduced the PHA-1+ and the dual matched version the PHA-11+. Unlike most MMICs designed for microwave applications the noise figure and IP3 out are at their best between 40 and 100 MHz, making them suitable for use as IF amplifiers in up-conversion receivers.

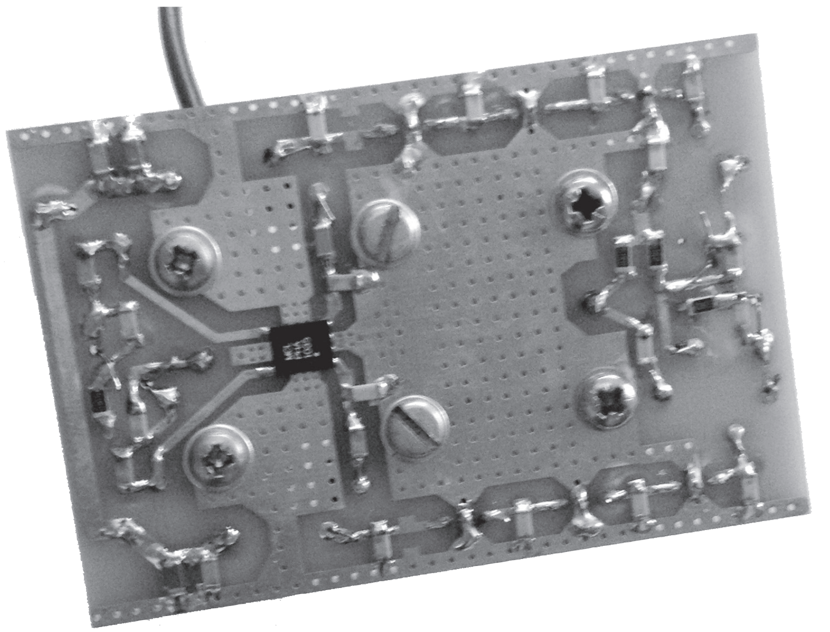


Photo A — Here is the circuit board/wiring side or the completed 45 MHz IF amplifier. The PHA-11 MMIC amplifier IC is to the left of center, near the RF output side of the amplifier. The wire connecting to the top of the board is the 5 V supply.

Unlike the 4 × J310 amplifier, however, the PHA-1+ has too much gain to drop it into the existing HF7070 IF strip. In addition its reverse isolation is not adequate for the output to directly drive a crystal filter like the 4 × J310 amplifier. This deficiency can be overcome by following the PHA-1+ with an 8 dB attenuator before the 4 pole crystal filter to smooth any reactance from the crystal filter and also make the PHA-1 output impedance nearer 50 Ω for the L match to the crystal filter. Fitting the 8 dB attenuator raises the input in-band IP3 of the crystal filter (26 dBm) to 34 dbm. This is still

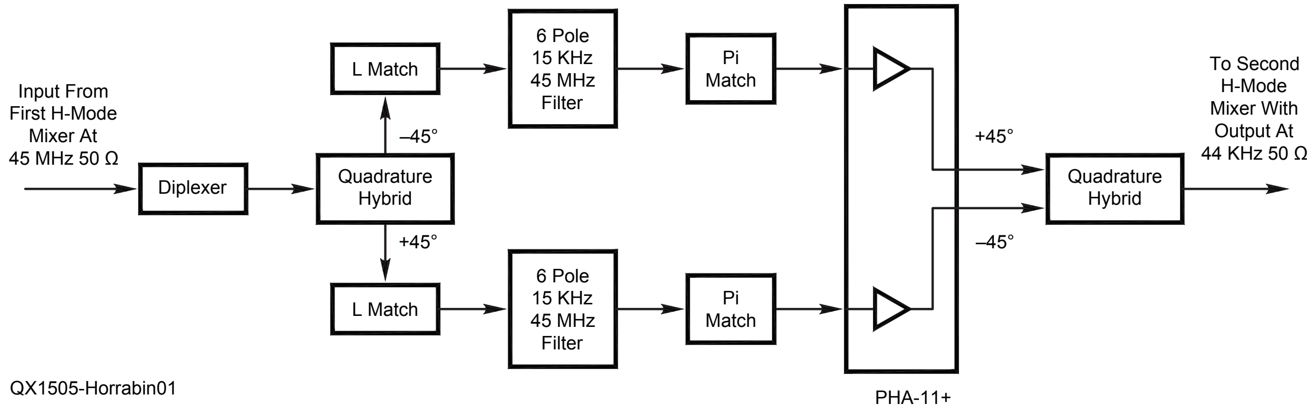


Figure 1 — This block diagram shows the improved architecture for the HF7070 45 MHz IF amplifier, using the PHA-11 MMIC. Note that it is necessary to have at least 110 dB of stop band attenuation at the second mixer image frequency. The 6 pole filter is made from three discrete 2 pole filters.

well below the PHA-1 output IP3 of 42 dBm.

To see if the PHA-1+ would be suitable in a practical design Dave Roberts, G8KBB, first measured its input impedance at 45 MHz using his N2PK vector network analyser (VNA). It was 80 Ω in parallel with 25 pF. Dave then measured the noise figure (NF) at 45 MHz from a 50 Ω source impedance, which was 2.2 dB. In one proposed use of this chip it would see an 80 Ω source impedance, so an L match was fitted to match its 80 Ω plus 25 pF input impedance to 50 Ω. When the chip saw an 80 Ω source impedance its NF was raised from 2.2 to 2.7 dB, which is quite a bit higher than the 1.3 dB of the 4 × J310 amplifier. The PHA-1+ has a gain at 45 MHz of 18 dB, however, and the higher gain could reduce the NF contribution from following stages.

In the present HF7070 45 MHz IF design the optimum gain block at 45 MHz is 20 dB and this is made up of two 4 × J310 amplifiers separated by a 4 pole roofing filter. Also the in-band IP3 of the receiver is ultimately determined by the in-band IP3 of the 4 pole crystal roofing filter that follows the first 4 × J310 amplifier. The 15 kHz bandwidth roofing filters at 45 MHz need to have six poles to get at least 110 dB attenuation at the second H-Mode mixer image frequency. In the existing design there are two poles of quadrature hybrid connected 45 MHz, 15 kHz bandwidth crystal filters after the first mixer. There is then a 4 × J310 amplifier and then a 4 pole filter followed by another 4 × J310 amplifier and then the second H-Mode mixer, to give a 44 kHz output frequency for the 25 bit audio ADC.

Because the PHA-1+ has 18 dB gain, a different IF architecture offered a simplification of the circuit and the possibility of a significant improvement of in-band IP3 and a better out of band IP3 at 20 kHz spacing. See Figure 1. The penalty would be a 2 dB increase in receiver noise figure. If necessary, a preamplifier for use above 7 MHz could be used, based on the push pull MRF 581A amplifier originally designed by Jacob Makhinson, N6NWP, that was presented in *QST*. The MRF 581A is still available, and an 8.8 dB gain with a noise figure of 2.5 dB and an IP3 out of 57 dBm should be possible. The use of this amplifier would not significantly affect the receiver dynamic range, and it would only really be needed to provide extra sensitivity of the receiver for reception of signals above 7 MHz.

In the new IF architecture described using the PHA-11+ MMIC, all six poles of roofing filter are in one place so that careful shielding from input to output is needed. See Figure 2. This has been achieved by machining a 12 mm thick piece of aluminium bar fitted inside a tin box of dimensions 50 × 75 × 25 mm. The aluminium bar also doubles as a heat sink to the PHA-11+ MMIC used in the design, which is the dual matched version of the PHA-1+. One of my neighbors, Alan Heywood — whose brother is a radio

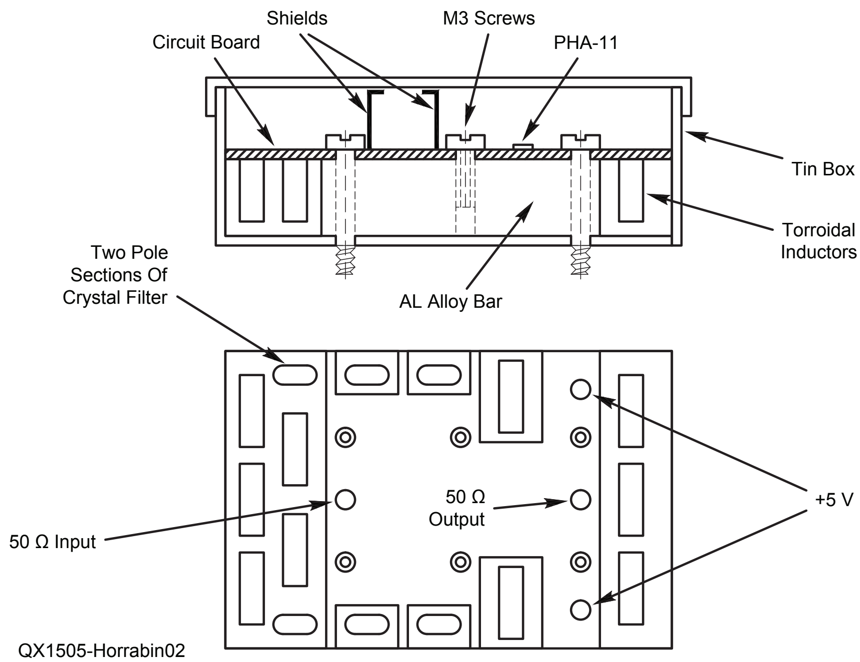


Figure 2 — This diagram shows the mechanical details of the PHA-11 amplifier.

amateur — builds model steam engines for a hobby, and has a good mechanical workshop. He offered to do all the mechanical work. The circuit shown in Figure 3 was constructed on a commercially made, double sided circuit board with plated through holes. Figure 4 shows the circuit board pattern. The circuit board has thermal vias to conduct heat from the chip to the aluminium bar. It was essential to keep the chip temperature down to get the best noise figure and with a dissipation of about 1.25 W, the chip temperature was only 34°C.

All of the inductors in the unit were wound on T37-10 powdered iron torriods because it was convenient to do so for the prototype. In a commercial design, surface mount inductors could be used to replace all the torriods, except for the two that form part of a quadrature hybrid. In the HF7070 radio most of the RF inductors were surface mount shielded chokes manufactured by Vishay.

Construction

I designed the circuit of the 45 MHz amplifier, the circuit board layout, and also drew up the mechanical side of the job. The commercially made printed circuit boards were made by Fischer in Germany. George Fare, G3OGQ, used a software circuit board design package to generate the necessary files for commercial manufacture from my free-hand graph paper layout. Martein Bakker, PA3AKE, arranged the manufacture of the circuit boards with the German company, and also paid for them as his personal contribution to this project.

A few PHA-11+ MMICs were supplied by John-Paul Newbold of Mini-Circuits Europe and two of these chips were reflow soldered onto the commercially made circuit boards in the electronics workshop at Daresbury Laboratory. The mechanical drawing of Figure 2 shows that the circuit board is secured to the aluminium bar by six M3 screws. The outer four screws project from the base of the tin box, so that the completed IF strip could be mounted to a motherboard. Before any of the passive components were soldered to the circuit board, the board was loosely screwed to the machined aluminium bar and this was placed into the tin box to see if there were any tolerance issues. There wasn't, so the six M3 screws were fully tightened, securing the circuit board to the aluminium block. The circuit board and aluminium block were then

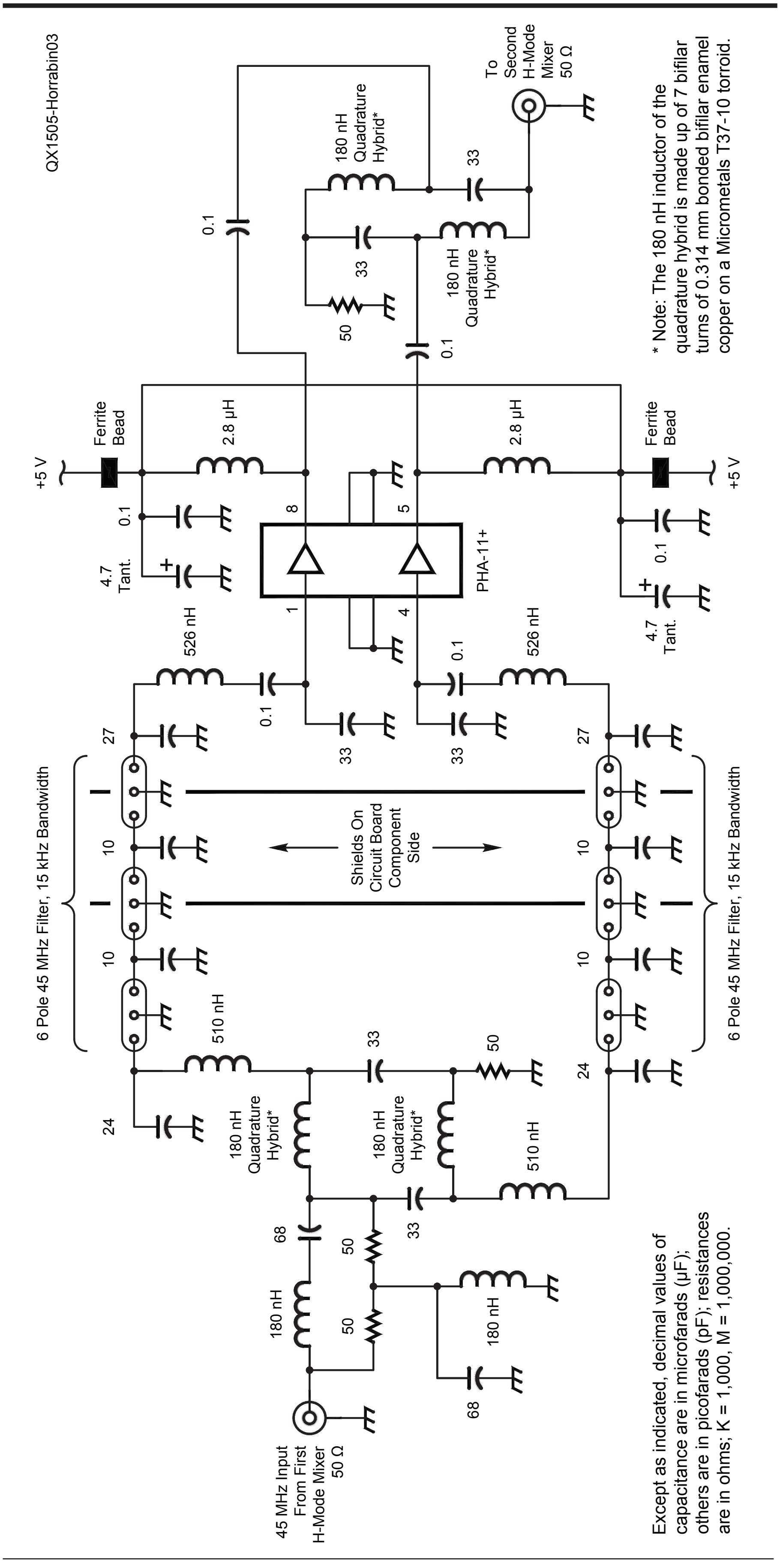


Figure 3 — This schematic diagram shows the 45 MHz IF amplifier circuit. Note that the 180 nH inductor of the quadrature hybrid is made with 7 bifilar turns of 0.314 mm bonded bifilar enamel copper wire on a Micrometals T37-10 powdered iron toroid.

gripped in a small vice so that the surface mount capacitors and 50 Ω resistors could be fitted. Unlike George and Dave, I had never done any serious surface mount assembly before, but by using a special pair of tweezers I found it easy to do (or as we British would say "A piece of cake").

All of the inductors in the design were wound on Micrometals T37–10 toroids. What made the winding of the toroids easy to do was the use of the capacitance and inductance measuring box from Almost All Digital Electronics USA (**www.aade.com**). The result was that the first time the amplifier was switched on it worked straight away and only minor adjustments were needed to remove a 1 dB dip in the passband. After the initial performance tests the edges of the circuit board were soldered to the tin box and two internal shields were fitted to improve the stop band of the unit at 45 MHz minus 90 kHz, the second H-mode mixer image frequency.

Performance Tests

The design team met at Dave Roberts' house one evening with the prototype 45 MHz IF amplifier. Although the unit was designed to mount on a motherboard, two SMA sockets were fitted to the unit for testing, and a shield fitted between them. Using an N2PK vector network analyzer (VNA), its transmission characteristics were plotted. See the graphs of Figure 5. Also the circuit noise figure (NF) and gain were measured using Dave's NF measuring gear. See Figure 6.

The noise figure was measured as 6 dB, and this was what it should be, remembering that the insertion loss of the 6 pole crystal filter forms part of this circuit and should account for about 3.5 dB. Overall circuit gain was 14 dB. What was disappointing was that the stop band was only 80 dB, although at that point the internal shields had not yet been fitted. It was nowhere near the 110 dB desired, however. Two internal shields were then fitted inside the tin box and its transmission characteristics measured again. See Figure 7. The stop band was improved by around 20 dB, but some more work is needed to further improve this.

Next, I visited the RF lab at Daresbury Laboratory to measure the in-band and out of band IP3 of the circuit. Two visits were required due to problems with the first set of measurements. I discussed the first set of results with Martein Bakker, PA3AKE, and set out the next time with a definite strategy in mind. The next IP3 results are shown in Table 1. The big surprise has been the significant improvement in out of band IP3.

The two amplifiers in the PHA-11 have their outputs summed by the second

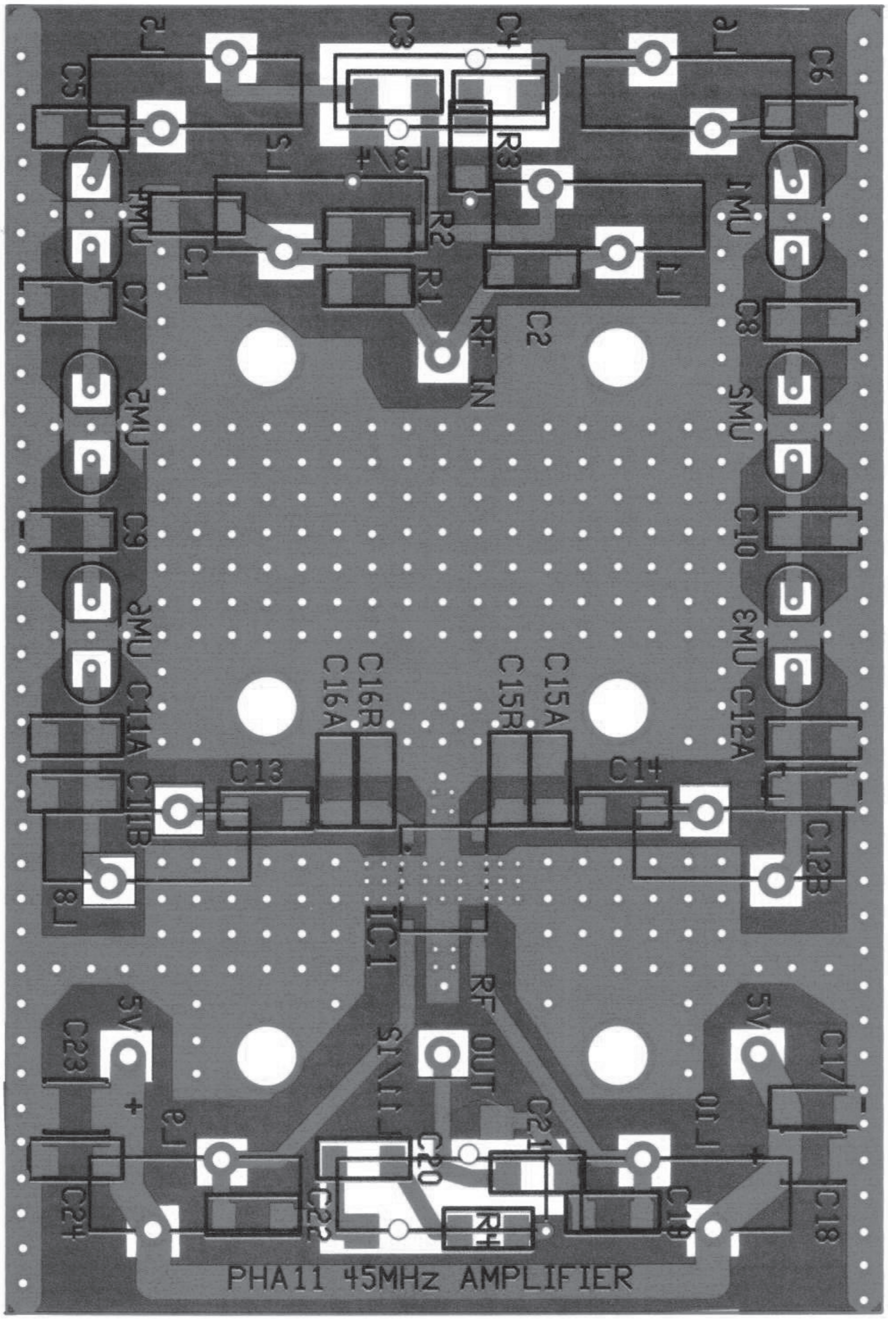
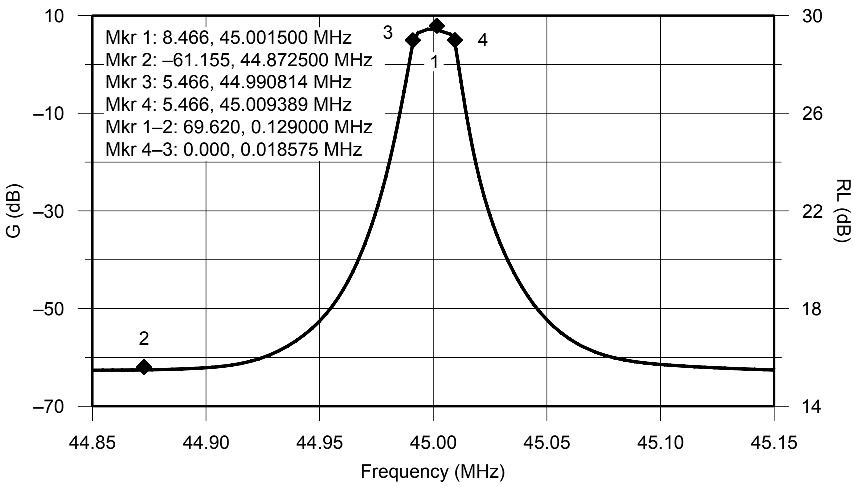


Figure 4 — Here is the circuit board etching pattern for the 45 MHz IF amplifier. This is not to scale. The actual circuit board files are available for download from the ARRL QEX files website. Go to www.arrl.org/qexfiles and look for the file 5x15_Horrabin.zip.

Table 1
Measured Performance of the IF Amplifier

Test Tone (kHz)	*IP3 Product (dBm)*	*Output IP3 (dBm)*	*Input Ip3 (dBm)*	*RX IP3 (dBm)*
2	–75	37.5	23.5	30.5
5	–96	48	34	41
10	–92 (–75)	46 (52.5)	32 (38.5)	39 (45.5)
20	–126 (–89)	63 (59.5)	49 (45.5)	56 (52.5)
40	–130 (–98)	65 (64)	51 (50)	58 (57)
80	–115	57.5	43.5	50.5



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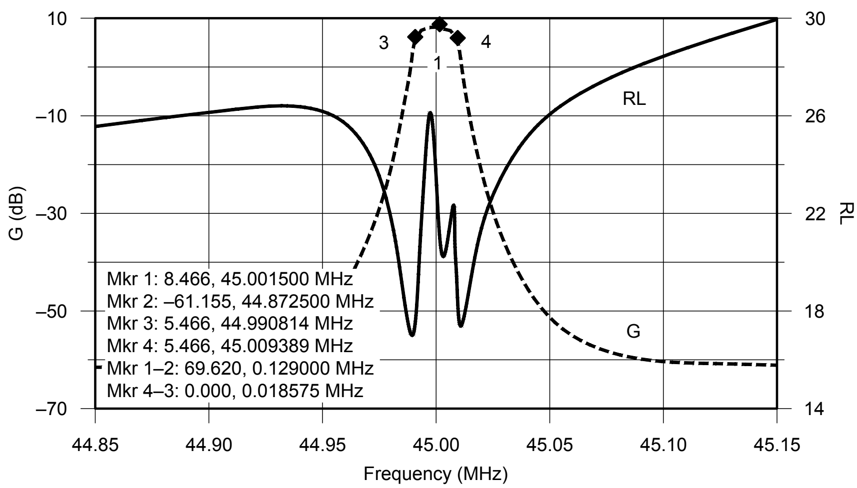
(A)



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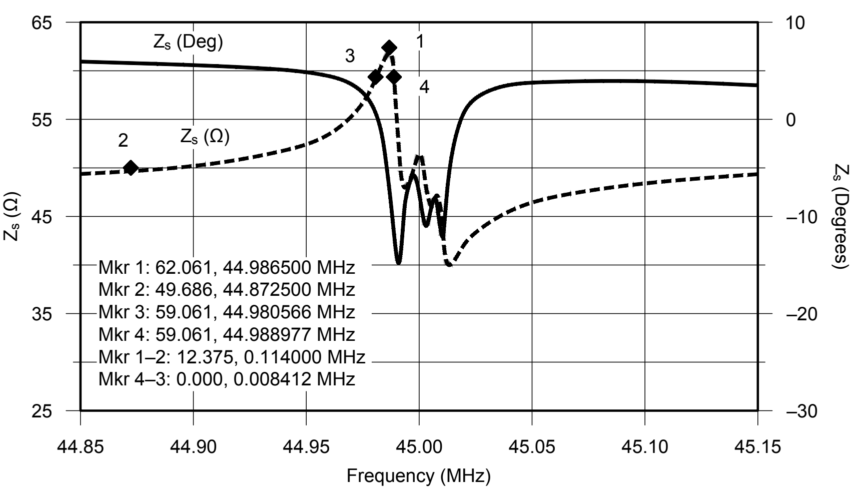
(B)



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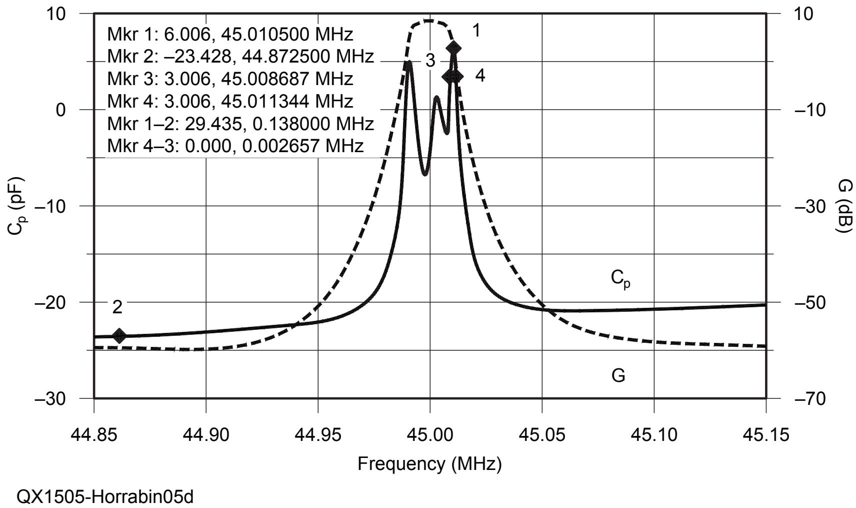
(C)



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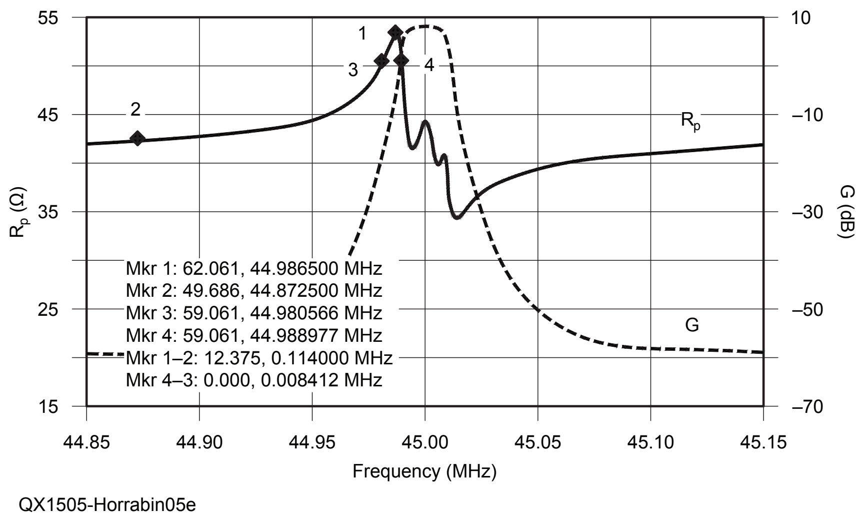

(E)

Figure 5 — These graphs are the performance measurements made by Dave Roberts using his N2PK vector network analyzer.

quadrature hybrid. For an amplifier output level of 0 dBm, each amplifier has an output of –3 dBm. Because it has a gain of 18 dB, the amplifier input level and the 6 pole crystal filter output level is –21 dBm. This gives an input level to the crystal filter of –17 dBm, and an input level to the complete amplifier of –14 dBm. This ties in with the measured performance of the unit as a gain block with a gain of 14 dB and a noise figure of 6 dB.

The main measurements of in band and out of band IP3 were made at an amplifier output level of 0 dBm. A few out of band measurements were made at a 10 dBm output level (shown in parentheses in the table) to give some idea of the linearity of the unit with signal amplitude. The projected IP3 of a receiver using this amplifier assumes a 5 dB mixer loss and 2 dB loss due to its input low pass filter. So effectively, the input intercept of the 45 MHz amplifier module has 7 dB added to its input IP3 to calculate the receiver performance. This presumes that the amplifier is used with the HF7070 front end. It is unlikely that the out of band IP3 of such a receiver will exceed 45 dBm, but Table 1 does show that the IP3 performance of the radio will not be limited by the IP3 of the roofing filters.

Figure 8 is a screen shot from the spectrum analyzer, with the test tones 2 kHz apart but offset by a few hundred hertz. The offset is necessary to identify the third order product, because at this spacing of the signal generators there are spurious peaks from the signal generators on integer boundaries. For the out of band measurements, the signal generator frequencies again had to be offset slightly because there was a low level spurious peak from the signal generator at exactly 45 MHz.

Observations

Using this amplifier to replace the 45 MHz IF amplifier in the HF7070 receiver makes possible a receiver IP3 performance of 50 dBm at 20 kHz tone spacing, for a receiver noise figure of 13.5 dB. In addition, the in-band dynamic range is also significantly improved and that was the real reason for constructing this IF strip using the PHA-11+ MMIC. In fact for in-band signals it is now the IP3 performance of the ADC that is the limiting factor. Improve the ADC and an in-band IP3 of over 30 dBm at 100 Hz spacing is possible for an up-conversion radio. The MMIC has gain at 3 GHz, and fortunately there has been no sign whatsoever of any instability with the circuit during testing. The circuit board layout and the Pi match capacitors close to the input of the chip have probably helped in this regard.

It is very unlikely that John Thorpe will incorporate this prototype amplifier into one of the HF7070 prototypes, to test it as part of a real radio receiver. With John, you never can tell, however!

The PHA-1+ / PHA-11+ MIMC is unsuitable for application in down conversion receivers for the amateur bands but could be used to give state of the art performance in an up-conversion radio with simple circuitry. If you are interested in a state of the art receiver for the amateur bands then PA3AKE's holy grail version of the CDG2000 transceiver is definitely the way to go. By publishing details of this 45 MHz MMIC IF amplifier, perhaps it might be incorporated into some shortwave radio transceivers. In principle the result could be a state of the art up-conversion receiver at a budget price. Mini-Circuits now supply the PHA-22+ MMIC. It looks like a 1.5 GHz version of the PHA-11+ so it should be of lower cost and a drop-in part for this application.

The noise figure of the MMIC chip used in the amplifier adds directly to the receiver noise figure. Mini-Circuits do make chips with 0.5 dB noise figures, but in most of these their performance below 100 MHz is unsuitable. Those that have a very low noise figure below 100 MHz have too much gain. In fact the PHA-1/PHA-11 appears to be the ultimate part made by Mini-Circuits for this sort of application at the present time.

This amplifier has been built into a tin box to get a good stop band and also so that a complete IF module could be mounted onto a circuit board motherboard. It is likely any potential manufacturer of a transceiver would want it to be part of the main circuit board. The 45 MHz monolithic crystal filters used in the HF7070 proto 2 receiver were obtained from the British company Total Frequency Control (TFC). The 6 pole filters used in this prototype 45 MHz amplifier were made up by splitting 4 pole pairs. Total Frequency Control's Japanese supplier does in fact make a 6 pole, 15 kHz bandwidth, 45 MHz monolithic crystal filter in one can. This has a poor stop band, however. They have been asked if they can produce a 6 pole filter with a 110 dB stop band at ±90 kHz. They could probably achieve this by constructing it in a larger can.

One problem with the HF7070 receiver was coupling between the two toriods used as part of a quadrature hybrid. Mini-Circuits makes a shielded quadrature hybrid centered on 45 MHz. So it may be possible to build this circuit on the main circuit board and get a 110 dB stop band at the second mixer image frequency.

I sent a copy of the circuit and the IP3 measurements to several American friends for their comments. Wes Hayward, W7ZOI, was amused that we have spent so much time and effort and our own money on this project because it is only of use to an "appliance" manufacturer and not for normal Amateur Radio projects. The *QEX* article on the HF7070 receiver showed

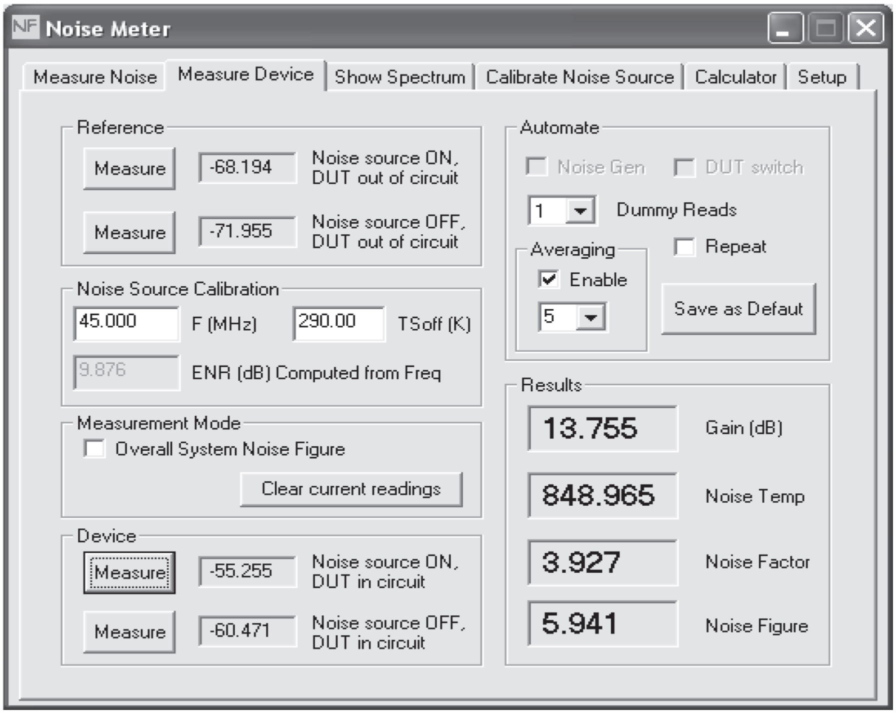

Figure 6 — This screen shot shows the results of the noise figure and gain measurement made by Dave Roberts.

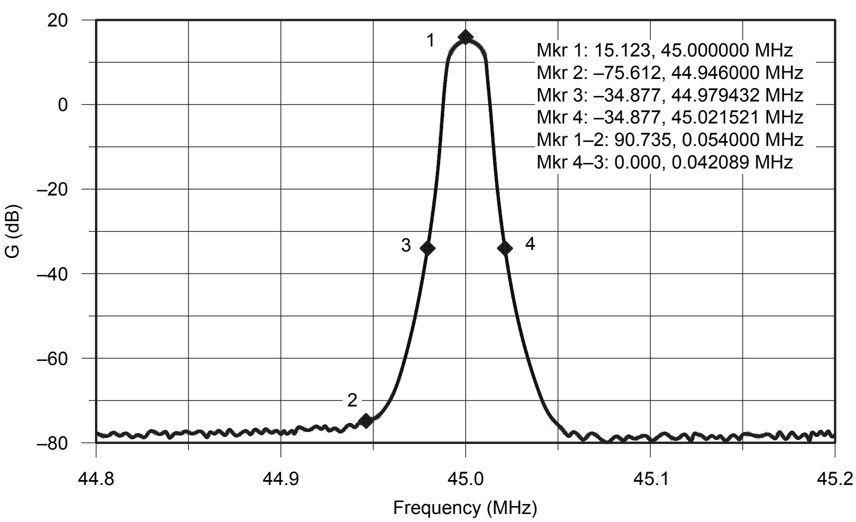

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Figure 7 — This graph is the result of a second measurement of the circuit transmission characteristics made with Dave Roberts' vector network analyzer, after internal shields were installed between sections of the circuit.

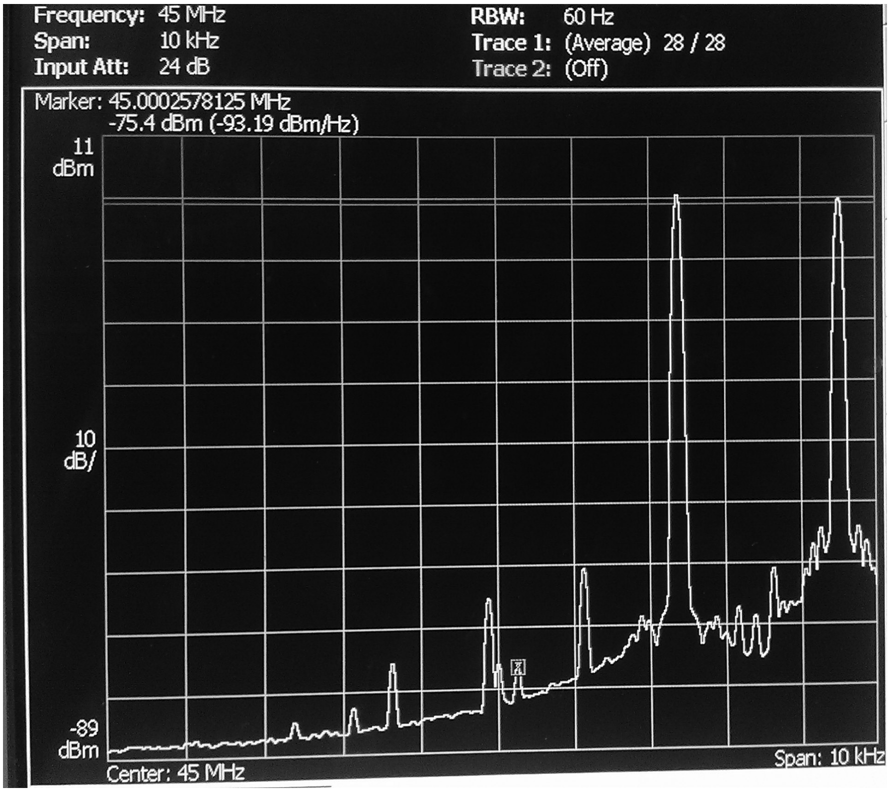


Figure 8 — This screen shot is the spectrum analyzer two-tone test measurements the author made at Daresbury Laboratory. The input tones were 2 kHz apart. The analyzer is set to a center frequency a few hundred hertz offset from the center of the tones.

that it should be possible for main stream "appliance" manufacturers to make a good up-conversion receiver that will outperform most commercial down conversion receivers. The application of the PHA-11+ in a 45 MHz IF amplifier can further improve the IP3 performance of an up-conversion radio like the HF7070, and also simplifies the circuit in that only one chip is required in the 45 MHZ IF amplifier.

Acknowledgements

This project needed the use of machine tools, and I would particularly like to thank my neighbor, Alan Heywood, for doing all the mechanical work.

Our friend Martein Bakker, PA3AKE, ordered and generously paid for the commercial circuit board with its thermal vias, and checked the circuit board files before they were sent to Fischer in Germany for manufacture.

Since Dave Roberts has an N2PK VNA and the capability to measure noise figure, it was only necessary to visit Daresbury Laboratory to measure IP3. As usual, my former colleagues were most helpful even though they were particularly busy at the time. I want to single out the help given by Andy Moss in the RF group, and Nigel Lightbown, who did the reflow soldering of the PHA-11+ chips onto the circuit board in the electronic workshop.

There was a problem in getting samples of the Mini-Circuits PHA-11 + MMIC from the US, because they wanted to sell only the minimum reel size, which came to about $300. John-Paul Newbold of Minicircuits Europe supplied us with some he bought on his own budget, however. He obviously realized the potential of this application of the chip in up-conversion radio receivers when others didn't.

Mark Sumner of MWS Technical Services provided the 45 MHz crystal filters that were made by Hertz Technology in Japan for use in the amplifier. Although these filters are no longer commercially available, those supplied by the British firm TFC Ltd have similar IP3 performance, and they were fitted in the HF7070 proto 2 receiver on which the technical performance measurements were made. Those measurements are presented on Martein Bakker's website.

Finally, although most consumer electronic products these days originate in the far east, it is still American companies that drive fundamental technological advances, of which the Mini-Circuits PHA-11+ is one such example. It remains to be seen how long it will be before it is used in commercial Amateur Radio equipment designs.

Appendix A: Variations on a Theme

In the present circuit, the output of each arm of the six poles of crystal filter went to one of the amplifiers in the PHA-11 and the outputs of the two amplifiers were combined by the second quadrature hybrid.

Two other circuits were considered that used a single device. In both these circuits the outputs of the two arms of the six poles of crystal filter were combined by the second quadrature hybrid before a single amplifying device. The devices considered for this were two Mini-Circuits MMICs: the PHA-1+ and the very low noise PGA-103+.

There were two main reasons for the present architecture. The first being that the quadrature hybrid input network seen by a MMIC is effectively a "floating" circuit. Because the MMICs have gain at 3 GHz we thought that the series inductance and parallel capacitance to ground of a Pi network at the input to the MMIC in the present architecture would give less chance of instability. Also the output IP3 would be increased by 3 dB and the output quadrature hybrid would present a good match to the second H-mode mixer.

If a single PHA-1+ MMIC is used in the alternative architecture at 45 MHz, the noise figure (NF) of the chip would be around

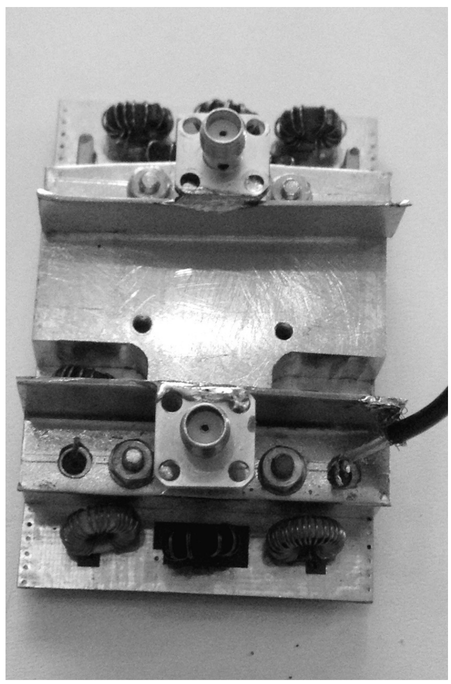


Photo B — This photo shows the component side of the completed 45 MHz IF amplifier circuit board, attached to the aluminum bar shield/heat sink. The SMA connector at the top of the photo is the RF input and the one at the bottom is the RF output. The length of small coaxial cable brings the 5 V supply to the circuit board.

2 dB, with a gain of 18 dB, an IP3 out of 43 dBm, and the output IP3 of the hybrid connected filters would be about 27 dbm for in-band signals. Putting these figures in John Thorpe's spreadsheet gives a receiver NF of 13.6 dB, an IP3 of 26.3 dBm, and a dynamic range in a 2.4 kHz bandwidth of 101.8 dB for close-in signals. The loss because of the six poles of crystal filter and two hybrids is around 4 dB.

The PGA-103+ at 45 MHz has a very low noise figure of 0.5 dB a gain of 26 dB and an output IP3 of 37 dBm. Putting these figures into the spreadsheet gives a receiver performance of NF of 11.6 dB, an IP3 of 20.3 dBm, and a 99 dB dynamic range in a 2.4 kHz bandwidth. This is close to the performance of the HF7070 receiver using the 4 × J310 amplifiers. The other difference would be that the out of band IP3 from 20 kHz outwards would be superior to the present HF7070 receiver.

In both of these circuits the gain of the HF7070 in the spreadsheet for the 44 kHz balanced amplifier was changed so that the total gain from antenna to the 25 bit audio ADC was the same, at 22 dB.

Commercial design can be all about compromise, so the use of the PGA-103+ could be the chosen option to avoid the use of a preamplifier with the receiver above 7 MHz. Where the present architecture will score (and also if a single PHA-1 is used) is that it will have a much better IP3 in the crystal filter transition region for test tones from 5 kHz to 20 kHz than if a PGA-103 + was used. This all presupposes that the MMICs will not become unstable when driven by the output from a quadrature hybrid.

Having demonstrated the use of the PHA-11+ MMIC in this type of 45 MHz IF amplifier, I will leave it to others to try the other options described, to see if these circuits remain stable when driven directly from a quadrature hybrid.

Colin Horrabin, G3SBI, was born in 1941. His father provided him with a World War II BC348 radio receiver for his 12th birthday, followed by a copy of the ARRL Handbook *for Christmas. After years building various projects using government surplus equipment, he obtained his Amateur Radio license in 1963. He has a degree in electrical engineering and a degree equivalent qualification in mechanical engineering. Following an apprenticeship with the British Aircraft Corporation in the early 1960s, he spent over 30 years working at Daresbury Laboratory as an electronic engineer. Colin is interested in small DX antennas for the LF bands, and intends to do some work on small multi turn spiral wound loops that are self resonant, containing ¼ wavelengths of wire, which are suitable for transmitting.*

Reference Material

See the Mini-Circuits website for data on the PHA-1+ / PHA-11+ / PHA-22+ and the PGA-103+ MMICs and other interesting parts: **www.minicircuits.com**.

Reed Fisher, W2CQH, "Twisted-Wire Quadrature Hybrid Directional Couplers," *QST*, Jan 1978, pp 21 – 23. Note that better results at VHF can be achieved by using bonded bifilar wire with the added practical advantage that each wire has a different colored enamel. Such wire is available from the Scientific wire Company (**www.wires.co.uk**).

The "Instruction Manual" for the L/C Meter IIB from Almost all Digital Electronics, USA: **www.aade.com**.

Data on Micrometals T37-10 powdered iron RF toroids is available on the Micrometals website: **www.micrometals.com/rfparts/rftoroid3.html** and various vendor websites.

G R Jessop, G6JP, *Radio Data Reference Book*, Fifth Edition, 1985 (RSGB Publication) page 58, "Pi and L-Pi Network Couplers — Improved Design Methods" (about lower impedance ratio matching circuits).

The N2PK VNA by Paul Kiciak, N2PK, (**http://n2pk.com/**) with modifications by Ivan Makarov, VE3IVM, (**www.makarov.ca/vna.htm**) and a USB interface and *Windows* application software by Dave Roberts, G8KBB (**www.g8kbb.co.uk/**).

Dave Roberts, G8KBB, "The measurement of Noise," *RadCom*, Jan 2007 pp 70 – 80. This is an article about Dave's noise figure measuring system.

ARRL Handbook CD

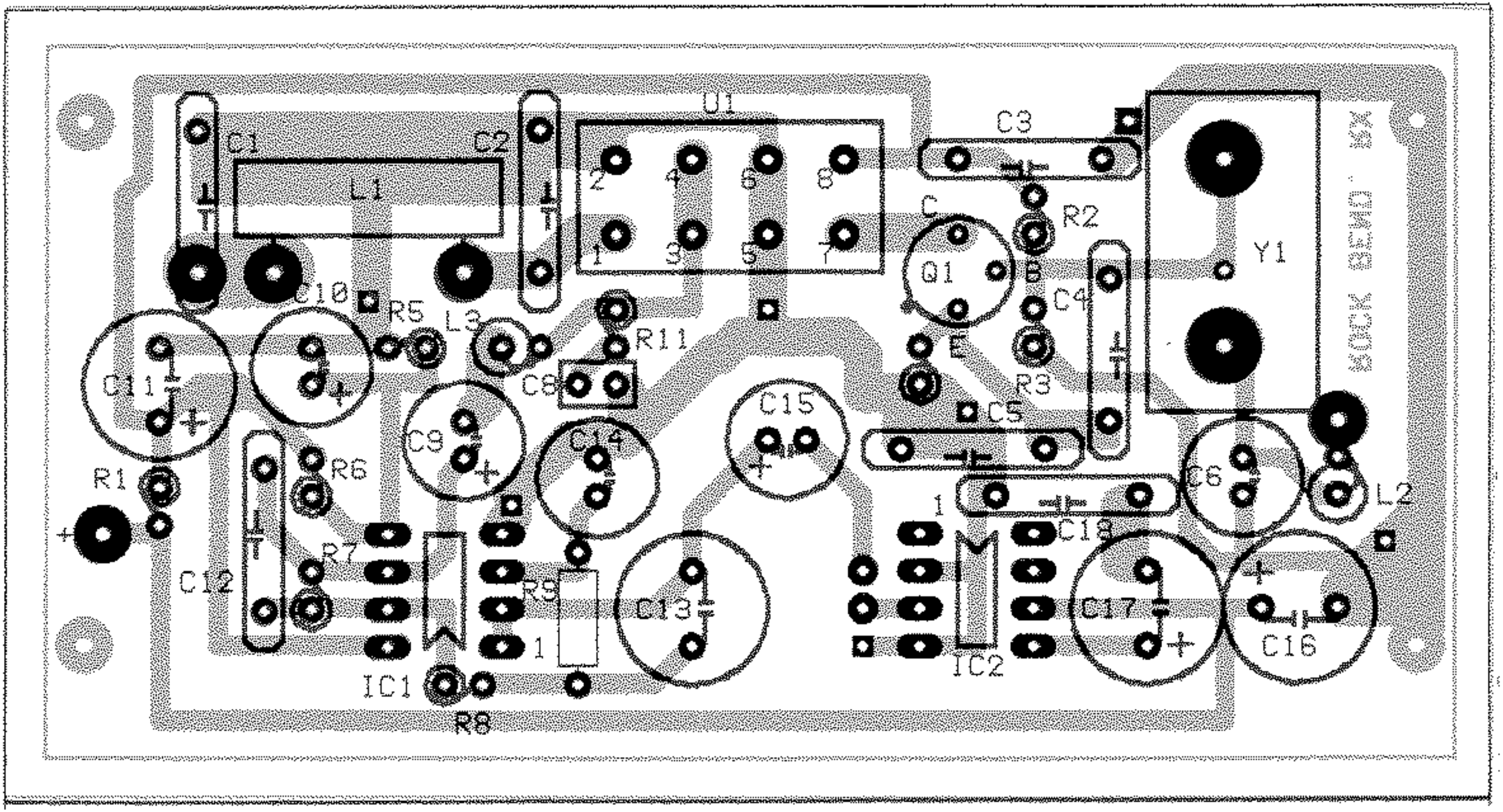
Template File

Title:	Rock Bending Receiver
Chapter:	12
Topic:	A Rock-Bending Receiver for 7 MHz

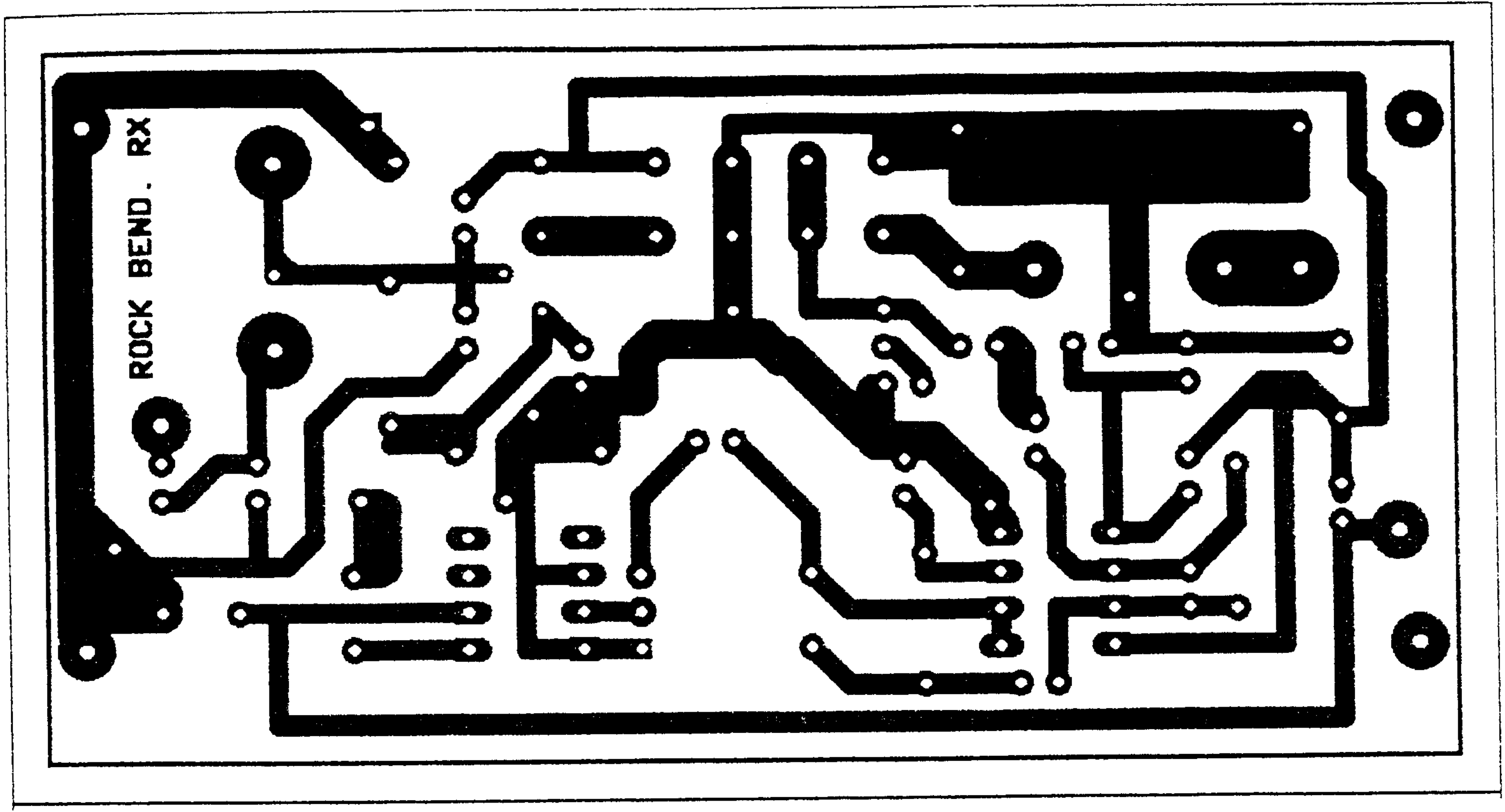
Template contains:

Component placement diagram.

Printed circuit board etching pattern.



R12 (10 Ω) MUST BE ADDED IN SERIES WITH C18


ROCK BEND. RX