# Shortwave Cap

## 0-30 MHz SSB/CW/FM/AM/DRM based on DDS and RISC

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**As a special treat for all radio amateurs we present a general-coverage AM/FM/SSB receiver with a wide range of features, which uses a DDS chip in the VFO section and also has a DRM output that can be fed into a computer. The receiver is controlled by a modern 8-bit Atmel RISC processor. The frequency readout is on a clearly legible 7-segment LED display.**

This receiver can be seen as the successor to the receiver published in January 1999. The experience gained with this predecessor (which, incidentally, has given a lot of listening enjoyment to many constructors), has been used to develop a more advanced RF receiver.

The best parts of the original design have been kept, such as the Intermediate Frequency (IF) and double conversion superheterodyne sections. This new receiver also has some new functions up its sleeve. For example, there is a DRM output that can be fed to a PC for decoding. The tuning resolution has also been improved to provide better fine-tuning, for example for SSB reception. This makes the frequency readout more precise and a BFO is no longer necessary.
Some thought has also gone into the aerial input stage, where an active input circuit makes the need for a long wire aerial superfluous.
We've also used a number of contemporary components, such as a DDS chip that is used as a VFO; the receiver is controlled by a modern 8-bit Atmel RISC processor.
In the design our preference went to 7-segment LED displays for the frequency readout, which look better and are easier to read than a standard LCD.
This receiver has three switchable bandwidths, each of which is optimised for use with one of the possible reception modes (AM, FM, USB, LSB or DRM).
The sensitivity and the ability to deal with large input signals have been improved for the reception bands of this receiver, largely through the use of a diode-ring mixer as the first mixer.
The receiver itself generates very little noise. This can be clearly heard when you aren't tuned into a station and connect an aerial: the noise then increases markedly.
A sensitivity better than 1 µV makes little sense at lower frequencies (roughly below 7 MHz), since it isn't the weak signals that make reception difficult, but rather the strong signals that swamp the others. A higher gain just doesn't make sense under these circumstances. Furthermore, you should find that most signals in that frequency range are strong enough to be picked up, even with a telescopic aerial.
In a nutshell, this is a receiver that is suitable to pick up all broadcast and amateur bands between 0 and 30 MHz.
The ease of operation, its various functions and its performance are guaranteed to provide you with many hours of listening enjoyment.

### Block diagram

The block diagram (**Figure 1**) starts with the aerials. The internal aerial is followed by a high impedance amplifier with adjustable gain. This amplifier isn't required for an external aerial. Directly after the aerial switch is a low-pass filter with a corner frequency of 30MHz. This suppresses any possible image frequencies and other unwanted signals. Following this is the first mixer. Its purpose is to convert the range of 0 to 30 MHz into 45 MHz. For this we require a VFO frequency of 45 to 75 MHz. A first IF of 45 MHz is a good choice because the first image frequencies are 90 MHz away, which make them easy to suppress.
The VFO signal is produced by a DDS generator. More details on its operation can be found in the DDS RF Signal Generator article that was published in the October 2003 issue. The reference frequency for this DDS is obtained by multiplying the 10 MHz crystal oscillator frequency by a factor of three to obtain 30 MHz. Inside the DDS a PLL multiplies this signal a further 6 times, so that the internal reference frequency

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becomes 180 MHz. Normally the maximum output frequency of the DDS should be around 40% of this reference frequency. But if we add a better bandpass filter at the output it is possible to increase this figure somewhat.

The DDS can now produce frequencies in 0.04 Hz steps. The required 100 Hz resolution therefore isn't a problem. The DDS is controlled by a microcontroller that also takes care of driving the frequency readout, the scanning of the keypad and a rotary controller for tuning.

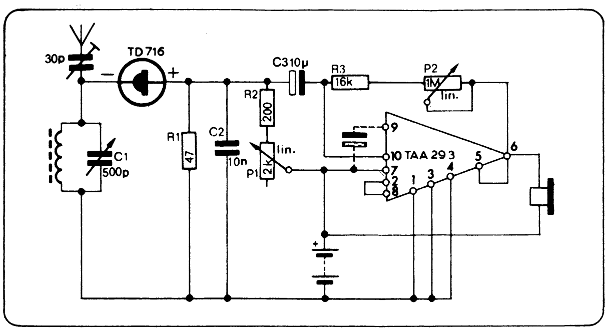
A number of extra I/O lines take care of the selection of the audio bandwidth and reception mode (FM/AM/LSB/USB).

After the first mixer is the first IF filter, which has a bandwidth of 15 kHz. This bandwidth determines the maximum possible bandwidth of the receiver. This filter suppresses the image frequencies of the second mixer. It also removes any other unwanted by-products from the output of the first mixer.

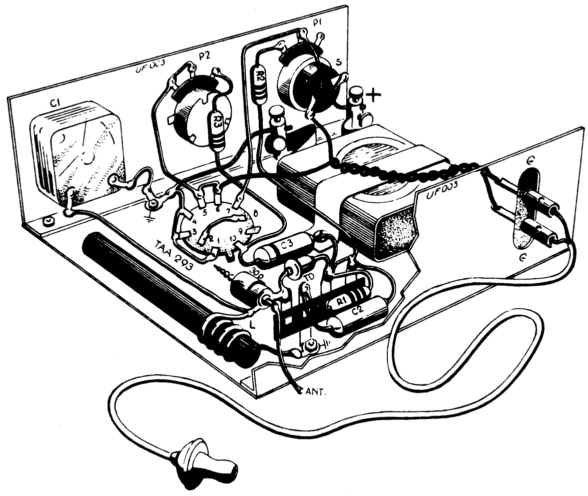
The signal now arrives at the second mixer. This converts the signal into an IF of 455 kHz. It does this using a fixed local oscillator frequency of 44.545 MHz. A frequency of 455 kHz was chosen because this low frequency can be easily amplified, and also because

## Specifications

- Double conversion superheterodyne receiver - first IF 45 MHz, second IF 455 kHz
- Microcontroller control of the DDS generator and other functions
- Tuning range of 0 to 30 MHz in steps of 1 kHz or 100 Hz
- DRM output, suitable for connection to a PC soundcard
- Audio bandwidth of 3, 6 or 15 kHz, dependent on the reception mode
- Keypad with 16 keys for inputting the frequency, mode and bandwidth
- Memory for 64 frequencies, including bandwidth and mode
- Synchronous detector for AM
- Quadrature detector for FM
- Product detector for SSB
- Built-in adjustable input amplifier for a telescopic aerial
- Approximate sensitivity of 1 to 2 $\mu$V (without preamp)
- Supply voltage of 13 to 15 V, max. 650 mA




## Back in time




In the early days of Elektor Electronics many types of receivers were published on a regular basis. Due to the relatively high cost of semiconductors in those days these receivers were usually fairly simple designs. As a comparison we'll show you a shortwave receiver that was published almost 40 years ago in Elektuur magazine. The use of a tunnel diode in this design is of particular interest because it was an unusual component at the time. The construction method was nothing like what we expect these days: some soldering pins and wires were used to connect all the components together because it was quite difficult at the time to produce a PCB.

**(Note: referring article not available in English)**

there are many inexpensive but good-quality filters available for this much-used frequency.
The gain of the second mixer can be adjusted automatically or manually.
To prevent overloading the receiver when very strong SW signals are picked up via a wire aerial we had to add a type of AGC circuit. Manually reducing the gain can improve the clarity of SSB signals in the amateur bands, because this reduces the QRM.

After buffering the signal it reaches three switches that select one of three filters, each with a different bandwidth. These switches are controlled by the microcontroller.
The three bandwidths in question are 3, 6 and 15 KHz. 3 kHz is suitable for SSB, 6 kHz for AM and 15 kHz for FM and DRM.
After another buffer is the IF amplifier. This actually consists of two amplifiers, each with an adjustable gain. This combination can provide a variable gain up to 80 dB, which is sufficient to keep the output signal level of the IF amplifier constant for both weak and strong signals. An AGC circuit is used to adjust this gain automatically. This adjustment follows the signal fairly quickly, for example to suppress the fading of AM signals. But for SSB reception it is made to react more slowly. The fast 'attack' and slow 'decay' of this AGC circuit improves the audio quality of SSB and CW signals.
The DRM detector takes the IF signal before it reaches the IF amplifier. This is because we don't need much amplification of the signal for this output. The DRM transmitters are powerful enough to produce a few hundred $mV_{pp}$ before the IF amplifier. This also avoids any noise from being introduced by the extra amplifiers.
The DRM detector consists of a product detector with a local oscillator of 467 kHz. The difference between this and 455 kHz is 12 kHz, which is filtered before being made available at the DRM output. The 10 kHz wide signal now covers frequencies between 7 and 17 kHz, which is exactly what is required by the DRM program, Dream.
An AGC circuit is not really necessary for DRM because the soundcard and the Dream program can cope with widely varying signal strengths.
Following the IF amplifier are the AM, FM and SSB detectors. The AM demodulator used here is a variation on the well-known synchronous detector. The principle of operation is that the AM signal is multiplied by an unmodu-

## Internal and external aerial

This receiver has the facility to accept two different aerials, each of which is suitable for different frequency bands.

Since even a suspended 20-metre length of wire has a higher impedance at low frequencies, an external aerial will work better than an internal one down to about 500 kHz. For lower frequencies it is better to switch to a telescopic aerial, which also has a high impedance, but is only very lightly loaded by the pre-amplifier.

The external aerial can be a 10 to 20 metre length of wire, suspended between 2 isolators, which in turn should be half a metre away from their mounting points. These isolators can be easily made from a length of PVC tubing. Two pieces around 20 cm long, with holes drilled at the ends are perfectly suitable. There is nothing special about the wire used for the aerial, as long as it can withstand strong winds and heavy rain.

We would advise against connecting an aerial longer than about 1 metre to the pre-amplifier. This can result in distortion and other interference. You would also pick up a lot of noise. A telescopic aerial or a 50 cm length of wire works perfectly well.

lated signal with exactly the same frequency and phase. This signal can be extracted using a PLL with a slow control loop, which keeps the frequency and phase intact should the AM signal momentarily fade. This results in less distortion than would otherwise occur when the signal fades. It is however also possible to extract the mix signal directly from the input signal as long as the modulation components are removed first. This is easily done with the help of a limiter. This limiter provides a mix signal with a constant amplitude. This results in good quality AM reception and provides better resistance to distortion caused by fading (which occurs in diode detectors).
We have used a quadrature detector for the FM demodulator. This type of detector is well known for the good quality audio it produces. The limiter used in the AM section is also used to supply a constant amplitude signal to the FM demodulator. This helps to avoid distortion caused by very strong input signals. It also suppresses AM noise components, which could overload the detector if their amplitude was too large. The overall result is a much better signal to noise ratio.
The actual operation of the quadrature detector is quite straightforward. The FM signal is multiplied by the same signal, but with a phase shift. This phase shift is exactly 90 degrees at the IF, but increases/decreases for an increase/decrease in the input frequency. The multiplier has a corresponding increase/decrease in its output voltage. A balanced multiplier is in fact very similar to an EXOR phase detector.
The SSB detector is a product detector. If we multiply the USB or LSB signal with another signal that differs in frequency by 1.5 kHz we obtain the audio signal. Note that the first mixer creates a mirror image of the spectrum because we've using high-side injection, so USB becomes LSB and vice versa. The local oscillator used here is controlled via the microcontroller to produce either 456.5 or 453.5 kHz. This way the selection can be made via the keypad.
The microcontroller selects the required mode with the help of analogue switches that follow each of the detectors. Since the microcontroller controls both the mode and the bandwidth it can automatically select the appropriate bandwidth for each mode.
After the switches is an adjustable low-pass filter with a steep cut-off. This suppresses whistles and other interference caused by nearby stations. This comes in very useful when receiving SSB and CW signals. The corner frequency can be adjusted from about 500 to 3500 Hz.
To reduce interference from hum and other low-frequency noise a high-pass filter is also included, with a corner frequency of 300 Hz.
The last stage is an audio power amplifier. This has sufficient output power to drive a 2 W loudspeaker.

## Schematic

We should now be familiar with the general design of the receiver. This makes it a lot easier to interpret the circuit diagrams in **Figures 2, 3 and 4**, and not be alarmed by the large number of components. The blocks shown in Figure 1 can be found in the diagrams quite easily.
We start with descriptions of the principal parts in the receiver section of **Figure 2**. At the top-left is the pre-amplifier for the telescopic aerial. This consists of a two-stage configuration using low-noise DG MOSFETs (T1/T2). The first stage provides a high-impedance input and some gain, the second stage functions as a 50 Ω driver. The gain can be adjusted from about +6 to –20 dB.

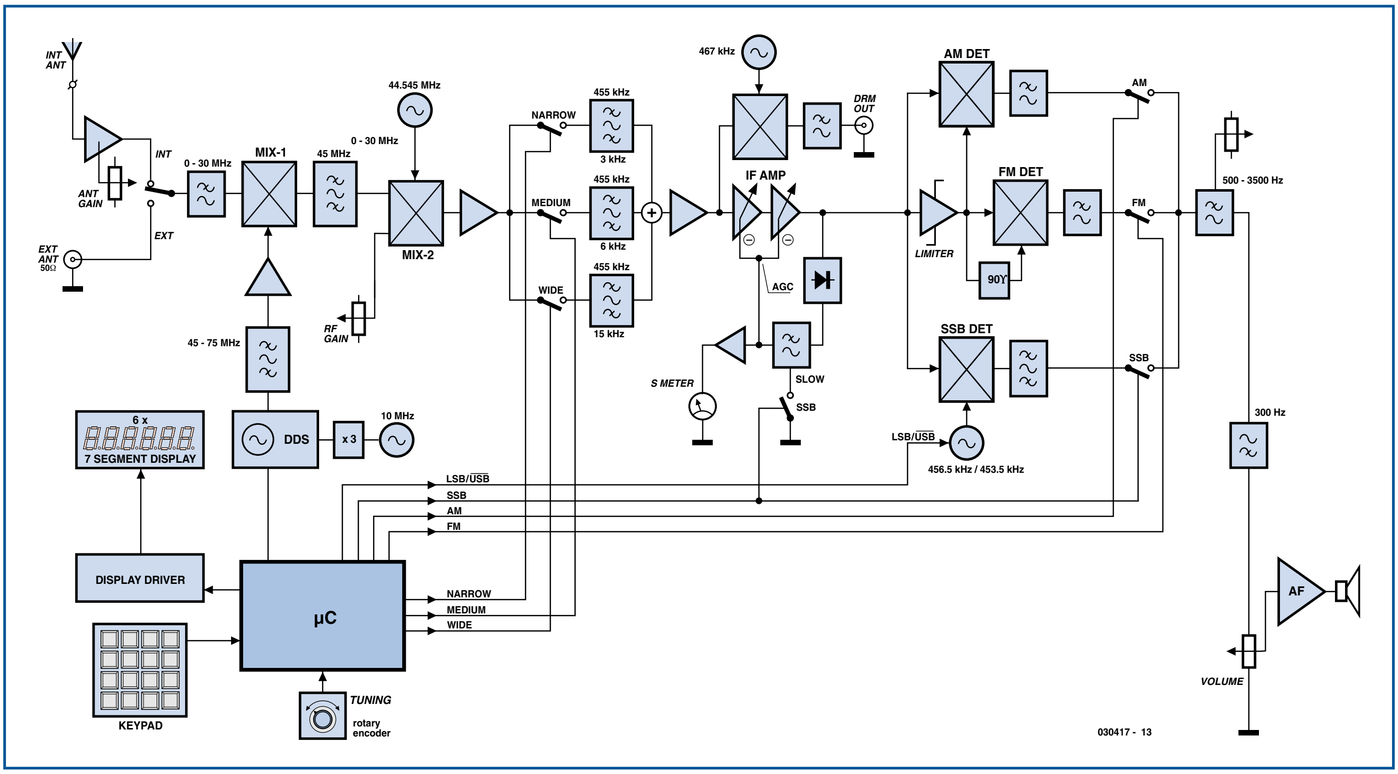
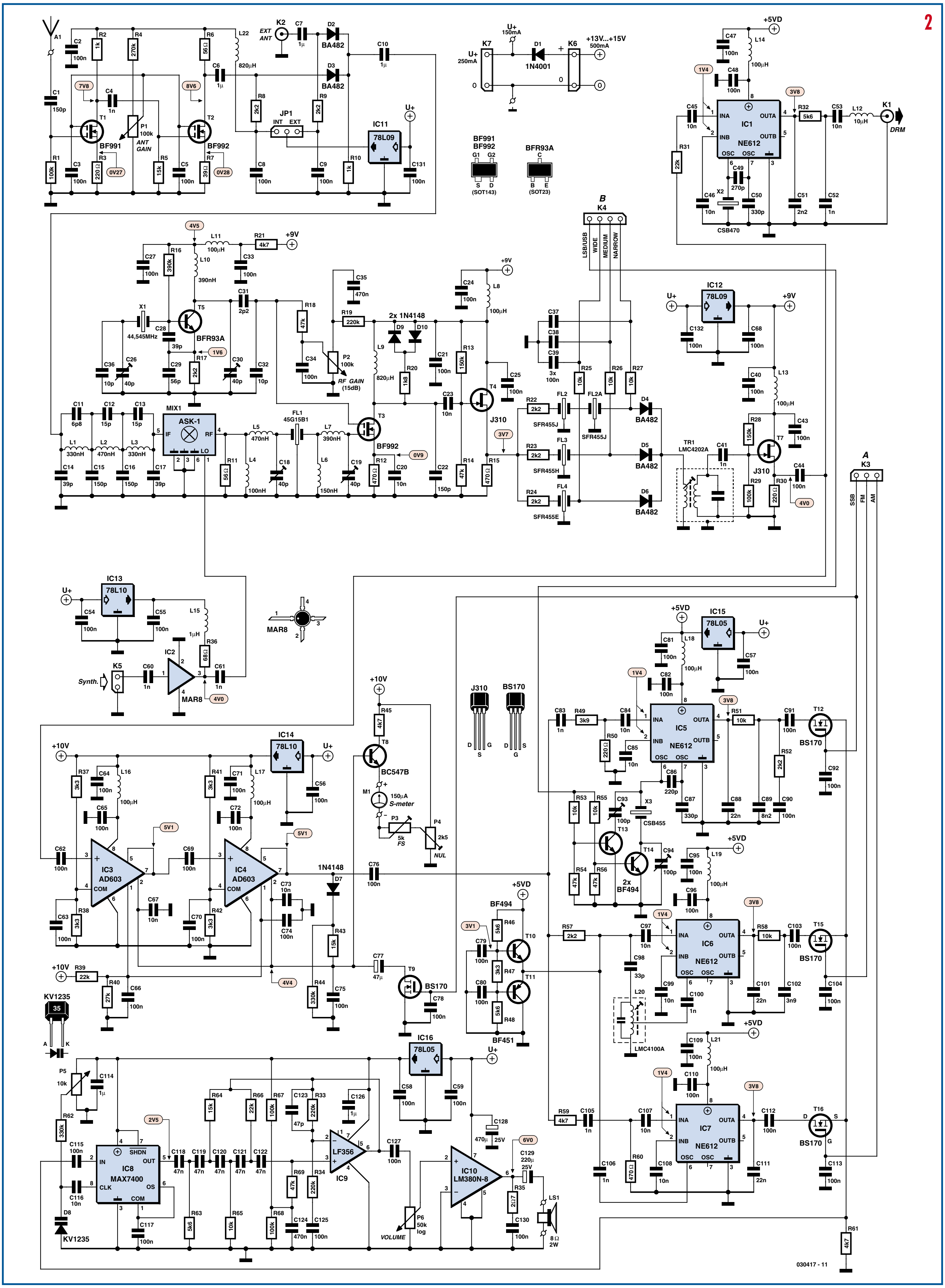


**Figure 1.** The block diagram for the shortwave receiver is fairly elaborate. At the bottom-left you can see the DDS generator and the microcontroller.


2
EXT ANT
K2
1N4001
U+ 250mA
U+ 150mA
+13V...+15V 500mA
+5VD
IC1 NE612
DRM
BF991
BF992
BFR93A
(SOT143)
(SOT23)
ANT GAIN
JP1 INT EXT
IC11 78L09
K4
LSB/USB
WIDE
MEDIUM
NARROW
IC12 78L09
44,545MHz
MIX1 ASK-1
45G15B1
RF GAIN (15dB)
2x 1N4148
J310
SFR455J
SFR455H
SFR455E
BA482
TR1 LMC4202A
K3
SSB
FM
AM
IC13 78L10
MAR8
Synth.
IC2
IC15 78L05
IC5 NE612
BS170
CSB455
2x BF494
IC14 78L10
IC3 AD603
IC4 AD603
BC547B
150µA S-meter
FS
NUL
KV1235
IC6 NE612
LMC4100A
BF451
IC16 78L05
IC8 MAX7400
LF356
IC9
IC10 LM380N-8
VOLUME
8Ω 2W
IC7 NE612
030417 - 11

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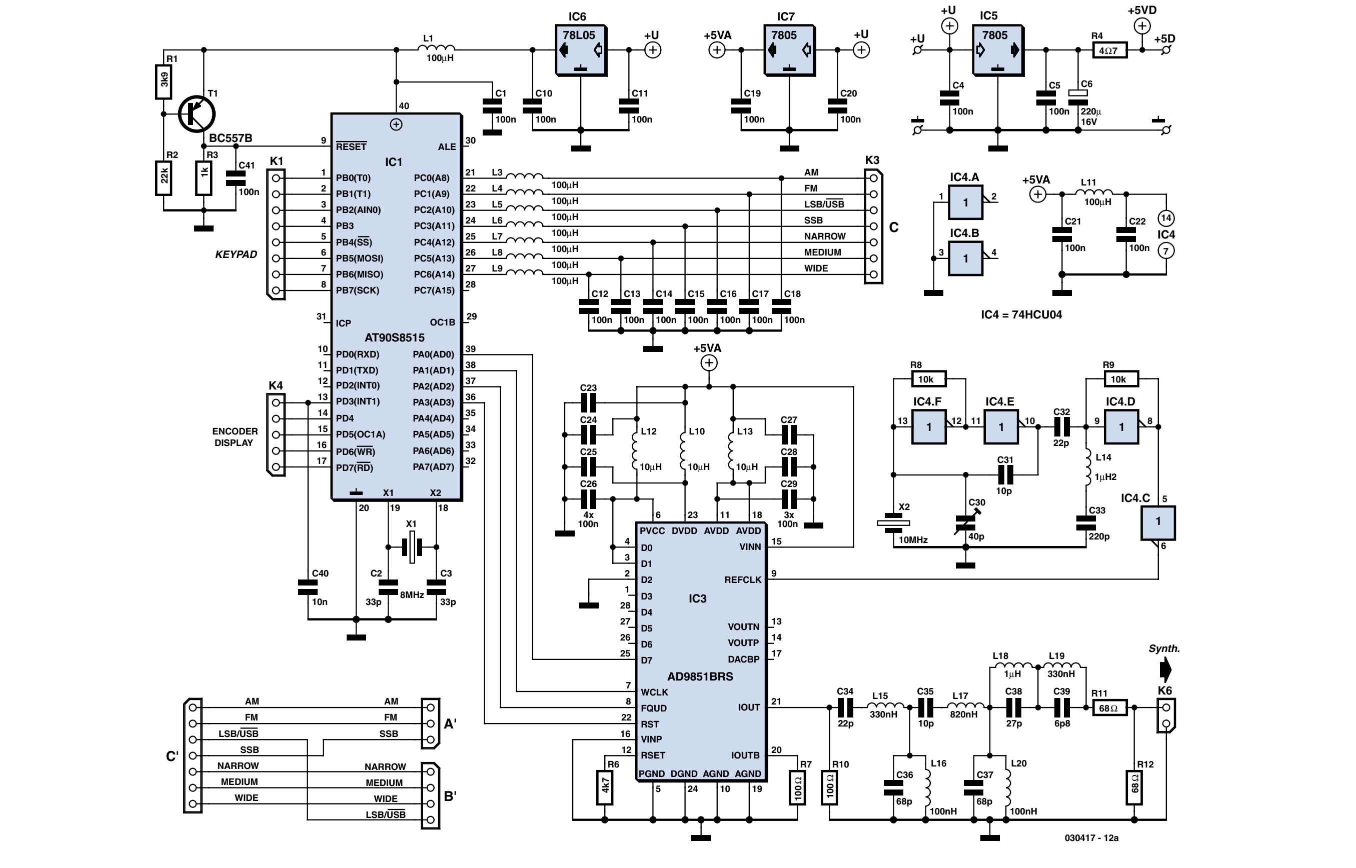


Figure 2. Schematic for the receiver section. There is a heap of components, but with the help of the block diagram you should be able to recognise the different sections.

Figure 3. The electronics for the microcontroller board. The wiring diagram at the bottom-left shows how K3 and K4 on the receiver board should be connected to K3 on the microcontroller board.

Figure 4. This part is on the display board: display driver, six-digit 7-segment display, rotary encoder and a keypad with 16 keys.

Two BA482 switching diodes (D2 and D3) have been used to switch between the pre-amplifier and the input for an external aerial. This type of diode has a very low AC resistance when a certain forward current flows through it.

From the aerial switch the signal is fed into a low-pass filter consisting of inductors L1 to L3 and capacitors C11 to C17. These form a Cauer-like filter, which cuts of steeply at 30 MHz and which has a fairly flat transfer function from 0 Hz onwards. The input and output impedance of this filter is 50 Ω. Following this input filter is the first mixer (MIX1). We have used an ASK-1 diode-ring mixer made by Mini-Circuits Labs because of its ability to handle

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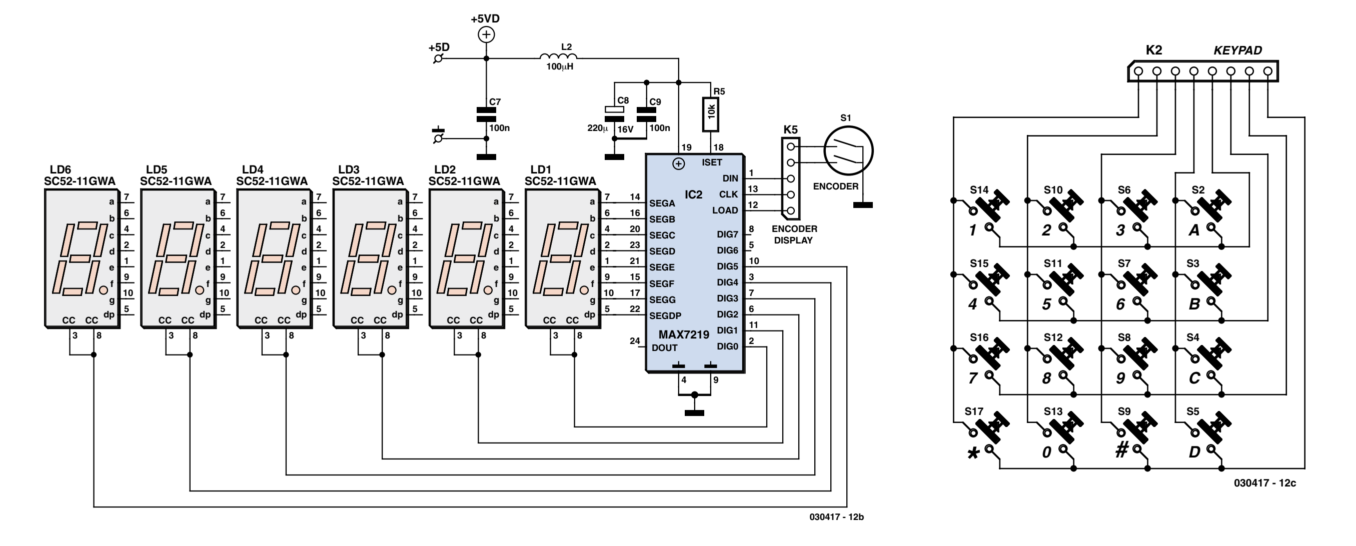


output impedance of 50 Ω. It also has enough output power to provide a good signal to the first mixer.
After buffer T4 come the three IF-filters. These can be individually turned on using BA482 switch diodes (using the same principle as for the aerial switch). In this way the processor can switch between bandwidths of 3, 6 or 15 kHz. The IF transformer (TR1) provides a load to the filters that is equal to their characteristic impedance (about 2 kΩ).
Following buffer stage T7 we end up at the IF amplifier consisting of two AD603 ICs made by Analog Devices (IC3/IC4). The gain of each IC can be adjusted over a range of 40 dB using a control voltage. This gives us a total AGC range of 80 dB. The AGC voltage is also used to drive the S-meter (M1) via T8.
The detector for DRM signals has been placed straight after the IF filters. The IF amplifier is therefore not used for DRM signals, as we saw from the block diagram.
The DRM signal is decoded using a product detector that mixes the input signal with a fixed frequency signal. The mixer chosen for this job, an NE612 (IC1), has an internal Colpitts oscillator. This can be set up in such a way that it will oscillate at a slightly lower frequency than the resonant frequency (470 kHz) of the CSB470 resonator used here. In this case we require 467 kHz, which is used to convert the 455 kHz signal into an audio signal with a centre frequency of 12 kHz and a bandwidth of 10 kHz.
We've now arrived at the detection stages for AM/FM/SSB. Each of the three stages has been built around an NE612 mixer IC.

large input signals. Eagle-eyed readers will have noticed that we've swapped the input (RF) and output (IF) of this mixer. The mixer still works well in this configuration, but it has the advantage that the input frequency can be a lot lower (almost down to 0 Hz, which is exactly what we require).
At the output of this mixer is a filter (FL1) that suppresses unwanted by-products and the image frequencies of the second mixer. An LC network at the output of the filter is used to match the impedances.

We now come to the second mixer, which is built around T3. Again we've chosen a low-noise BF992 DG MOSFET, which has a somewhat better gain than most of its equivalents.
As far as large signals are concerned, we don't demand as much from this mixer as from the first one because we're only really dealing with one frequency here.

The purpose of the second mixer is to convert the first IF (45 MHz) to the second IF (455 kHz). Specifically designed crystals with a frequency of 44.545 MHz are available for this.
The local oscillator signal for the first mixer is supplied by a VFO. This signal should have a frequency range from 45 to 75 MHz, with a resolution of 100 Hz. We have chosen an AD9851 DDS made by Analog Devices (IC3 in **Figure 3**), which provides a stable output that can be accurately programmed in small frequency steps. The frequency range is just about wide enough for use in our application.
At the output of the DDS is a band-pass filter (C34 to C39, L15 to L20) with a pass-band of 45 to 75 MHz. The impedance of this filter is about 100 Ω. The impedance is lowered to 50 Ω using resistors R11 and R12. The resulting signal is then fed into a MAR8 (IC2 in Figure 2). This IC (made by Mini-Circuits) is a wide-band amplifier with an

The SSB detector is a product detector, as is usual for SSB detection. The internal oscillator of IC5 has been configured in such a way that only one resonator is required for USB as well as LSB operation. The trick used here is that transistors T13 and T14 switch a trimmer capacitor either in series or parallel with the resonator.
The quadrature detector for FM demodulation is built around IC6. This detector is very suitable for demodulating narrow-band FM, which is normally used in the 11-metre CB band. Because this type of demodulator is somewhat sensitive to AM components, the signal is first fed through a limiter. This consists of a push-pull transistor pair (T10/T11). With the val-

ues for the base resistors as shown, the signal fed to the NE612 will be limited to about 250 $mV_{pp}$.
The simplest way of demodulating AM is with an ordinary diode. A disadvantage of this method is that distortion is introduced when the signal weakens. This is particularly a nuisance when the signal fades, which happens fairly often in the shortwave bands. A synchronous detector (IC7) is more resistant to these occurrences. For this we use the signal from the limiter in the FM demodulator as the mix signal. The amplitude of this signal is virtually constant, even when the signal strength varies a lot.
There are now three available audio signals, which can be selected by the microcontroller with the help of three BS170 FETs that are used here as analogue switches.

The receiver is equipped with an effective audio filter. This is an adjustable low-pass filter built around IC8, which is a switched capacitor filter. The MAX7400 contains a very steep 8th-order elliptic filter where the corner frequency can be set with a capacitor. If a varicap (D8) is used for this capacitor then the corner frequency can easily be adjusted using a potentiometer.
An active fifth-order high-pass filter built around opamp IC9 then filters out all frequencies below 300 Hz, thereby suppressing hum and other low-frequency noise.

The final stage is an audio power amplifier, for which we've chosen an LM380N8. This 8-pin IC is capable of delivering 2.5 W of audio power when supplied with a high enough voltage. In this case we supply the IC with the maximum permitted 15 V, which means it can output 1.7 Watts into 8 Ω. That should provide enough volume in even somewhat noisy environments.

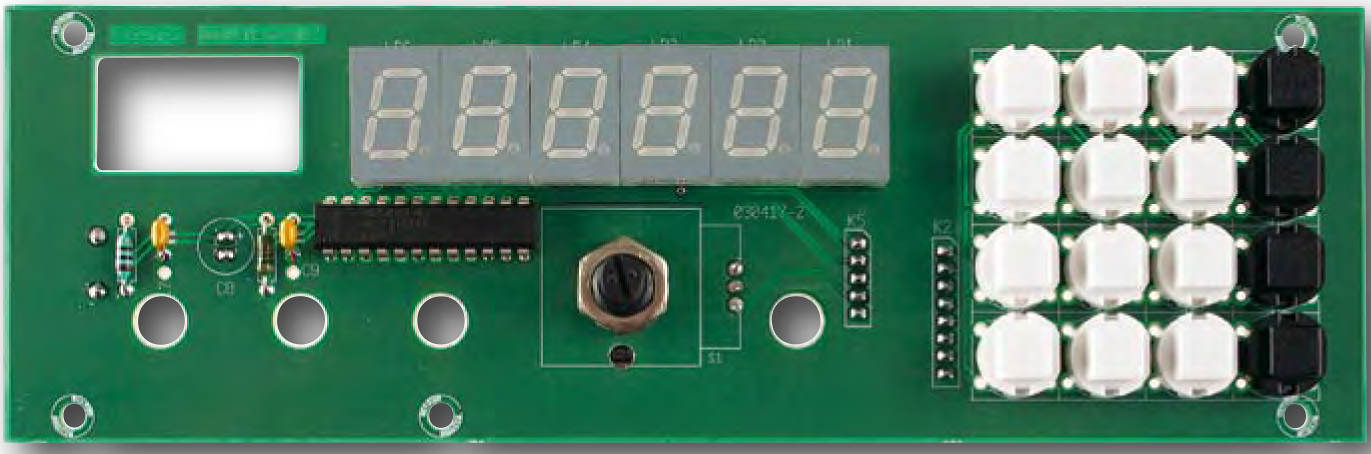
At the heart of the control section is an AT90S8515, a microcontroller made by Atmel (IC1, **Figure 2**). One of its many tasks is to provide the DDS chip (IC3) with the correct data. This happens serially, which requires fewer I/O lines. For each change in frequency the microcontroller sends a string of 40 bits to the DDS. This happens extremely quickly so you won't experience any delays when changing the frequency.
A rotary encoder (S1 in **Figure 4**) is used for tuning purposes, which supplies 24 pulses per turn to the microcontroller. The encoder output is connected to an interrupt line, so not a single pulse will be missed by the software. In this way you can tune through 24 kHz per complete turn at 1 kHz resolution, which is sufficient for AM and FM. The resolution can also be set to 100 Hz (via the D key on the keypad), which results in 2.4 kHz per turn. This is more suitable for tuning to a SSB station on the amateur bands. It would be nice if the encoder could provide more pulses per turn, so that you could always tune with a resolution of 100 Hz. In this case you would need to replace it with an optical encoder.
The keypad is another input device that is read by the microcontroller. This consists of 16 keys arranged in a 4x4 matrix. In this configuration you only need eight I/O lines to scan the keypad. The scanning is fast enough never to miss a key press.
Apart from the 10 digits there are four extra keys on this keypad (A, B, C and D), which are used as function keys. This reduces the number of switches required on the front panel, which makes the operation of the receiver easier and clearer.
Three I/O lines from IC1 provide the display driver (IC2, a MAX7219) with data. This display driver can drive a maximum of eight 7-segment displays. This IC also has the ability to control the segments of each digit individually. In this application we've made use of this to display an L for LSB mode, a U for USB mode, an F for FM and an A for AM, followed by the bandwidth of either 3, 6 or 15.
And last but not least, the microcontroller also controls the settings for the bandwidth and modes. This is implemented very simply using seven I/O lines, which are set high or low by the software as required.
The software for the microcontroller was written in assembly language and consists of about 1800 lines. The program has 21 subroutines, a reset routine, two interrupt routines (encoder, timer) and a main program loop.
A programmed microcontroller can be ordered from Elektor Electronics using order code **030417-41**. In contrast with other Elektor Electronics projects we can't provide you with the source and hex files.

Each of the boards for the receiver has been provided with a liberal number of voltage regulators, suppression chokes and capacitors. The supply voltage of 13 to 15 V (0.5 A) is provided by a mains adapter.

## Construction

The electronics for the whole receiver have been divided across three boards: a receiver board, a microcontroller board and a display board. Due to a lack of space we can only show you photos of the completed boards in this article. All PCB layouts, component overlays and the accompanying parts lists can be downloaded free of charge from http://www.elektor.com/ (look in magazine/December 2006/Shortwave Capture). Ready-made circuit boards are available from Elektor's business partner 'The PCBShop' (Eurocircuits)
In a radio receiver you obviously have to use several special RF components. The best places to buy these are specialist electronics firms, such as internationally operating Barend Hendriksen (www.xs4all.nl/~barendh) or HaJé Electronics (www.haje.nl).
A reasonably experienced electronics hobbyist shouldn't have many problems with the construction. There is only one component that is difficult to solder: the DDS chip, which comes in an SSOP package!
There are two parts lists: one for the receiver board (-1) and one for the microcontroller and display boards (-2). Take care not to mix up the part numbers between -1 and -2.
The receiver board (030417-1) has been made very compact. Because of this, the space for resistors is smaller than usual and the leads have to be bent closer to the body. All SMD FETs and SMD transistors have to be soldered on the underside of the board. The MAR8 can be soldered either way up, since its connections are symmetrical. Do take care that the lead with the indicator dot is soldered to the correct pad.
We decided to make JP1 (aerial input selection) a header with a jumper, but if you intend to change the aerial input on a regular basis then it will be much easier if you add another switch to the front panel instead.

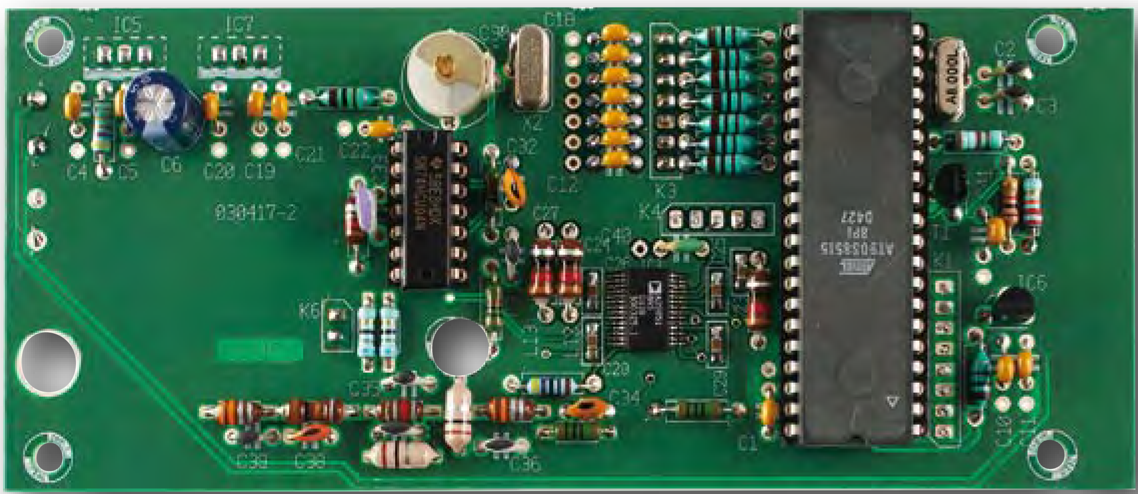
As we mentioned earlier, the soldering of IC3 and its associated components on the microcontroller board requires very delicate work. The decoupling capacitors are also SMD; these should preferably be 0805 types, but 1206 ones fit as well. You should first solder two decoupling caps (C24, C27) underneath (!) L12 and L13.
The headers for K3, K6 and the supply are soldered onto the solder-side of the microcontroller board (assuming that you don't want to solder the con-



nection leads straight onto the board). The voltage regulators (IC5 and IC7) are also found here. These should be provided with a heatsink, otherwise they could overheat when the case is screwed shut.
On the display board the following components are mounted on the underside of the board: the rotary encoder S1 (so the spindle sticks out through the front), the headers (if required) for K2, K5 and the supply, and electrolytic capacitor C8.

It is best to use flexible wire for the connections between the boards. If you have plugs at one end it will be easier if you ever need to separate the boards in the future. The cables should be kept away from the receiver board as much as possible, otherwise you're asking for trouble (unwanted oscillations etc.). Route the cables directly away from the board and only then bend them towards the other boards. Keep in mind that the order of the pins on K3 and K4 on the receiver board is different to that on K3 on the microcontroller board. We've added a wiring diagram in Figure 3 to remind you. You should use a short length of thin 50 Ω coax cable like RG174 for the synthesiser connection.
The supply for boards -1 and -2 loops from one to the other, which is why there are two connectors for the power supply. On the receiver board you could use K6 for the incoming +13 V for example, and use K7 to link it to the microcontroller board and display board. To avoid buying many different types of plugs we have used only 5 or 8-way types and just removed the unwanted pins when necessary.
The dimensions of the boards are such that they all fit in a standard case made by Bopla. The receiver board is mounted flat on the bottom of the case and the other two boards are mounted vertically behind each other (display board at the front). The spindles of the potentiometers of the receiver board fit through the holes in the other two boards.
The potentiometers should therefore have long spindles so they can stick through the front panel. When you use the recommended Bopla case, the receiver board can be fitted using a few M2.5 bolts.
It won't do any harm if you put some aluminium foil on the bottom inside the case and connect it to ground. This provides some extra shielding, which improves the stability of the receiver. An unetched piece of PCB is also suitable. We don't want any short-circuits, so the PCB should be placed with the copper side down, or if you used aluminium foil you should cover it with insulation tape.

The microcontroller and the display boards are mounted vertically inside the case. Although there are slots on the left and right-hand side of the case for this purpose, we didn't use them because the boards wouldn't fit. Luckily the boards can also fit on either side of the slots. The 7-segment display ends up just behind the front panel and there is also enough room for the microcontroller board this way.
At the back of the case is a connector for the supply, a feed-through for the telescopic aerial and a hole for a phono-socket for connecting an external aerial.



## Adjustments

There are nine trim points in this receiver. To start with it is best to set all trimmers at their halfway position. Before you continue, make sure that an aerial is connected and selected.

Select the AM mode using the A key on the keypad. Next you should type in the frequency of a strong medium wave transmitter. If you use an internal aerial you should set the gain to its maximum; this also applies to the RF gain. Set the audio volume control to halfway. If necessary, use the rotary encoder until you hear a medium wave station. First set trimmer C30 in the collector of the local oscillator of the second mixer to obtain maximum volume.

Next you should set the trimmers of the first IF (C18 and C19) to obtain maximum volume.

The full-scale trimmer (P3) and the null-trimmer (P4) for the S-meter should now be adjusted. The rest of the settings can now be completed with the help of the S-meter reading.

If a very strong signal is received you should reduce the aerial gain a little.

Adjust the core of the IF transformer following the IF filters (TR1) to get the maximum S-meter-reading.

When the sensitivity of the receiver improves you can try tuning to a weaker station. The previously mentioned trimmers should again be adjusted to get the maximum S-meter-reading.

In a double conversion superheterodyne receiver there are two oscillators that determine the accuracy of the reception frequency: the reference oscillator for the DDS and the local oscillator for the second mixer. You will need a frequency counter to set these correctly. First type in a frequency of 30.000.0 MHz. Then adjust the trimmer of the 10 MHz reference oscillator (C30 on the microcontroller board) until the frequency at the output of the MAR8 (IC2) is 75.000.000 MHz. The frequency counter should now be connected to the second gate of the second mixer (T3). Use trimmer C26 near the 44.545 MHz crystal to adjust the frequency to 44.545.000 MHz.

Once all these settings have been completed we can start with adjusting the FM demodulator. For this we need to tune in to an FM transmission in the 27 MHz band. We then need to adjust the core of the IF transformer (L20) in the FM demodulator. (Make sure that you have selected the FM mode with a 15 kHz bandwidth, using the A key.) As an alternative, you could adjust it for minimum audio output when receiving an AM station in FM mode, which is actually a bit more precise.

The last trimmers to be adjusted are the two trimmer capacitors in the SSB product detector. Tune in to a known AM station, for example in the medium wave. Then use the B key to select LSB, and adjust C94 until you have a beat frequency of 0 Hz. Now use the B key to select USB, and adjust C93 until you have a beat frequency of 0 Hz.

The receiver is now ready for use and the case can be put together.

## DRM use

In order to process DRM signals you have to connect the DRM output of the receiver to the line input of a soundcard in a PC. Next you have to start a program called Dream. The bandwidth of the receiver has to be set to 15 kHz using the C key. The mode doesn't matter, but if you select AM you may be able to tell via the loudspeaker if there is any interference in the DRM signal.

After setting up the soundcard on a PC (this is described in the March 2004 issue of *Elektor Electronics*) the receiver can be tuned to a DRM transmitter. Many DRM stations don't transmit continuously, but according to a certain schedule. You can check via the loudspeaker of the receiver whether you've tuned into a DRM transmission. A harsh noise should come out of the loudspeaker. You can then turn down the volume on the receiver. Finally, you use the recording control on the PC to set the required level.

(030417-I)

# Operation

**Overview of all keypad functions:**

| | |
|---|---|
| **0 to 9** | numerical input of frequency |
| **D** | 'Enter' after inputting the frequency |
| **A** | AM(6)/FM(15) mode |
| **B** | LSB(3)/USB(3) mode |
| **C** | select 3, 6 or 15 kHz bandwidth, independent of the current mode |
| **D** | select 100 Hz or 1 kHz tuning resolution |
| ***** | store the frequency, mode and bandwidth into memory |
| ***mm** | store into memory entry mm (mm = 00 t/m 63) |
| **#** | recall the frequency, mode and bandwidth from memory |
| **#mm** | r ecall memory entry mm (mm = 00 to 63) |

(When the receiver is turned on, memory entry 00 is used by default.)

The display for the frequency readout consists of 6 digits and has a resolution of 100 Hz. This also means that when you input a frequency via the keypad you have to enter it down to the last 100 Hz. For most stations you'll find that you have to input the frequency in kHz followed by an extra zero. But to receive the DCF time signal on 77.5 kHz you just type in 775 and D.

There are two fixed decimal points on the display. So for channel 14 in the 27 MHz band for example, the display will show 27.125.0. When the A or B keys are used to change the reception mode, the display will show the selected mode for several seconds. For example, after pressing the A key it will show A-6, which means AM mode with a 6 kHz bandwidth. Pressing the A key again will show F-15 (FM mode, 15 kHz bandwidth).

Pressing the B key changes the display to L-3 (LSB mode, 3 kHz bandwidth). Pressing the B key again shows U-3 (USB mode, 3 kHz bandwidth).

Repeatedly pressing the C key doesn't change the mode, but rather the bandwidth. Again, the display will show the last selected setting.

When none of the A, B or C keys has been pressed for a period of 2 seconds the display will automatically revert back to the frequency readout.