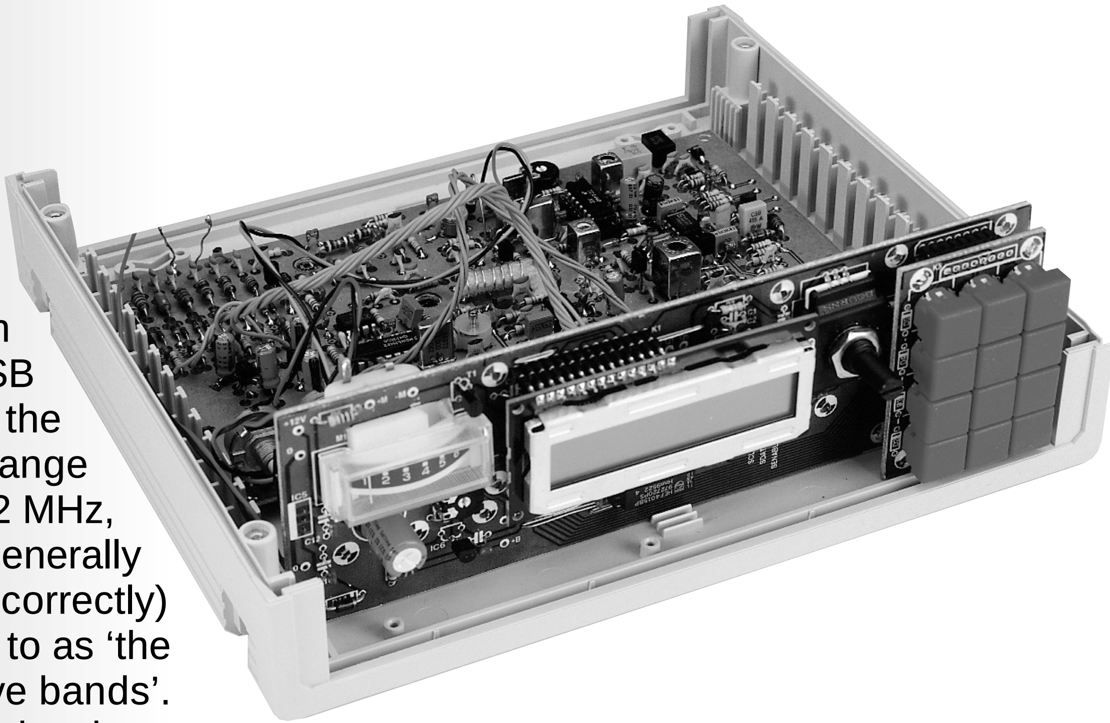
general-coverage receiver

part 1: circuit descriptions



This two-part article describes an AM/FM/SSB receiver for the frequency range 0.15 – 32 MHz, which is generally (but incorrectly) referred to as 'the shortwave bands'. The receiver is microprocessor controlled and avoids many of the pitfalls traditionally associated with RF construction.

Main Specifications

- *Double conversion superheterodyne receiver, 1st IF 45 MHz, 2nd IF 455 kHz*
- *Microprocessor control of synthesizer tuning and other receiver functions*
- *150 kHz to 32 MHz tuning range in 1-kHz steps.*
- *Selectable selectivity: 3 kHz (narrow) or 12 kHz (wide)*
- *Internal 6-band preselector with automatic band switchover*
- *12-key keyboard for frequency entry, mode and bandwidth selection*
- *16-character LCD shows receive mode, bandwidth, frequency and preselector band*
- *Memory for 21 frequencies, incl. bandwidth and mode*
- *Spurious product rejection >50 dB*
- *Audio output power approx. 1 W into 8 Ω.*
- *Power supply 15 V, max. 400 mA (approx. 90 mA without audio and LCD backlight)*

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The receiver we're about to describe is the product of many hours of designing, testing and programming by the author, a licensed radio amateur from the Netherlands. Throughout the design process, the emphasis has been on repeatability, ease of construction and avoidance of many of the pitfalls commonly associated with building radio equipment. As many of you will avow, the two best known pitfalls are winding your own coils and non-availability of specialized test equipment to align the receiver, or, indeed, any other RF project you may want to build.

So how are these problems solved? Well, the present receiver has only one inductor you have to wind yourself, and the use of ready-made filters and transformers in the IF sections obviates the need for complex constructions and adjustments. If you are a careful builder with some experience in RF technology, then the receiver should work spot-on, and a minimum of adjustments is needed to tweak it for optimum performance. The good news is that these adjustments only require the built-in S meter, your hearing ability, and possibly a voltmeter.

The concept

The block diagram of the general-coverage receiver is shown in **Figure 1**. The design is that of a double-conversion superheterodyne receiver with a 'high IF', which means that the first intermediate frequency (IF) is well above the highest receive frequency.

The antenna signal is first taken through a preselector stage whose main purpose is to reduce the risk of interference and cross-modulation products caused by very strong signals. The preselector is manually tuned for best performance. The second function of the preselector is to make the receiver input virtually independent of the antenna used: in fact, anything ranging from a simple telescope antenna to a full-blown 'beam' (with a cable impedance of 50 Ω) or a long-wire may be connected. Alternatively, for indoor use, consider a small magnetic-loop antenna such as the superb DJ8IL design described in the September 1998 issue of *Elektor Electronics*.

The preselector is followed by a preamplifier stage with manually adjustable gain. Here, again, one of the most important design considerations is to keep strong signals away from the input of the next stage, the mixer. If you are new to shortwave reception, then remember that your main concern is not dredging in the noise to get the weakest possible signal *into* the receiver, but to keep multi-megawatt signals *out*.

The local oscillator (LO) signal for the first mixer is supplied by a synthesiser circuit which can be tuned in steps of 1 kHz across the range 45.150 MHz to 77.000 MHz. The synthesizer consists of the usual ingredients: a VCO (voltage-controlled oscillator) a prescaler, and a loop filter for suppression of the reference frequency (here, 1 kHz). Like a number of other sub-circuits in the receiver, the synthesizer is digitally controlled by a central microprocessor.

The output signal of the first mixer is taken through a 45 MHz filter with a bandwidth of about 15 kHz. The main function of the filter is to suppress the image frequency of the second mixer, which occurs at 44.090 MHz (44.545–0.455).

The first IF signal (45 MHz) is heterodyned down to 455 kHz by means of the second mixer and the second LO signal, which is supplied by a crystal oscillator operating at 44.545 MHz. The mixer is followed by two bandpass filters, one with a width of 3 kHz for 'narrow-band' mode (SSB), and one with a width of 12 kHz for FM and AM reception. The gain of all IF amplifier stages (45 MHz and 455 kHz) is controlled by an AGC circuit (automatic gain control). Because the AGC voltage is a measure of the received signal strength, it can also be used to drive the S-meter.

The last 455-kHz amplifier drives two demodulators (for AM/FM reception), and a product detector (for SSB reception.) The oscillator in the product detector can be pulled a little to allow USB/LSB selection. The relevant control is labelled BFO (beat frequency oscillator). Analogue switches are used

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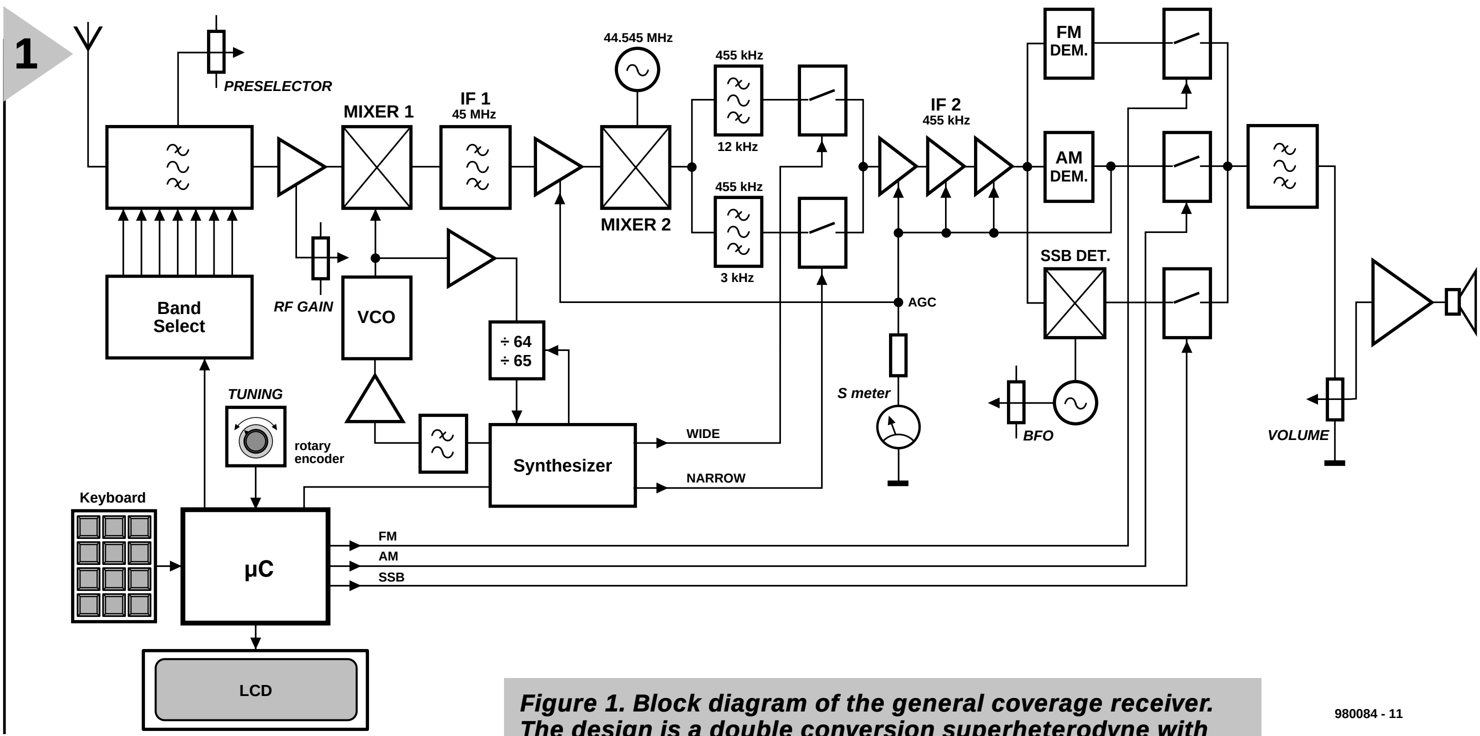

Figure 1. Block diagram of the general coverage receiver. The design is a double conversion superheterodyne with high-side injection for the first LO. The use of a 'high' first IF (45 MHz) guarantees a minimum of in-band generated spurious products while also reducing the risk of IF breakthrough. Note that many functions are controlled by a central microprocessor.

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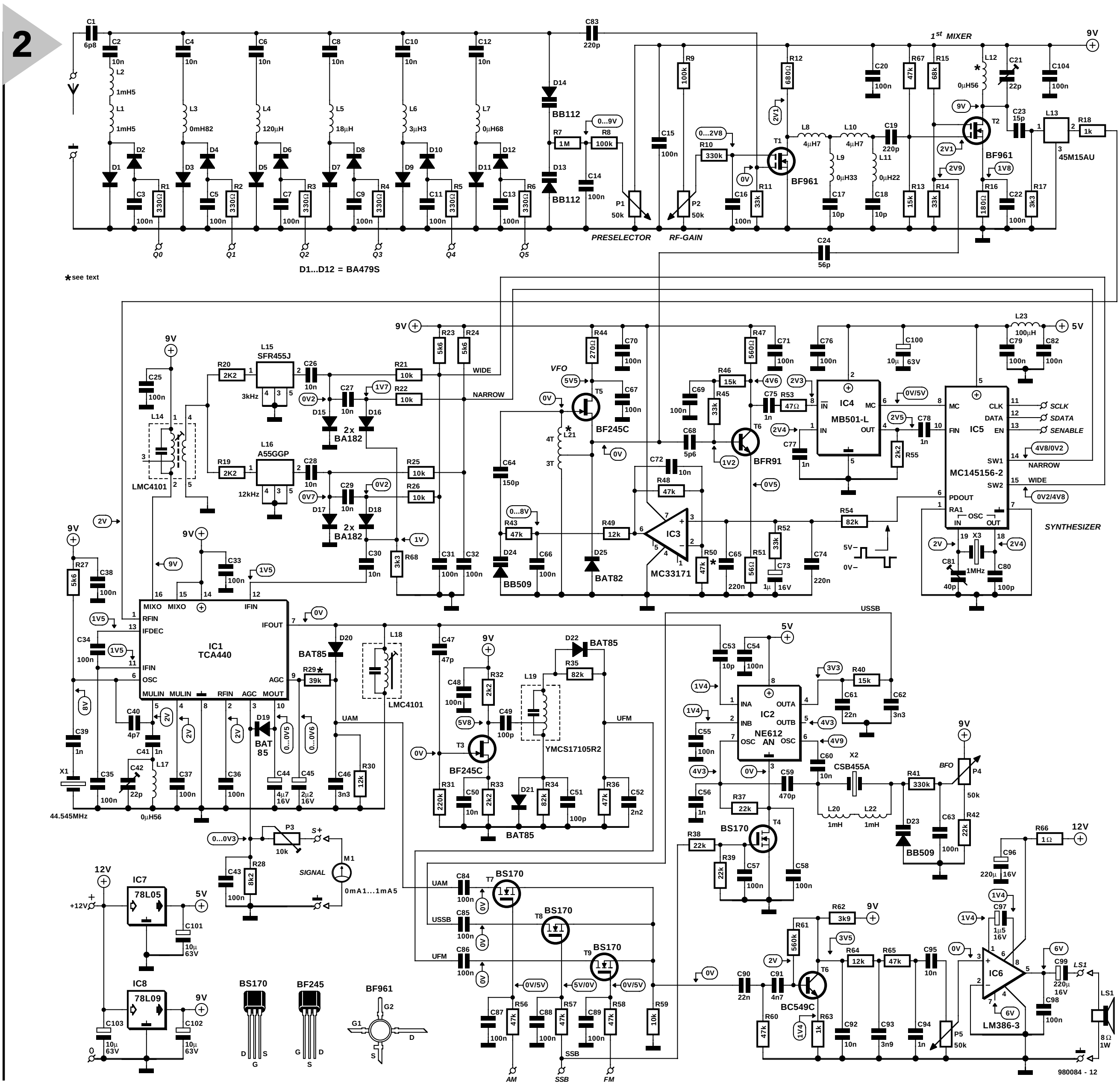

Figure 2. Practical circuit of the RF sections of the general coverage receiver. Most of the functions defined in the block diagram will be easy to find back in this schematic.

to feed one of the demodulator/detector outputs to the input of the audio amplifier, by way of a 'speech' filter with roll-off points at 450 Hz and 3.3 kHz.

The microprocessor circuit controls the preselector, the synthesizer, the IF bandwidth (wide/narrow), the mode selection (AM/FM/SSB, and the LCD (liquid crystal display). Its 'input devices' are a rotary encoder for the receiver tuning, and a small keyboard for direct frequency entry and several other functions like channel memory control, manual bandwidth selection (3 kHz/12 kHz), etc.

PRACTICAL CIRCUIT

Drawing a block diagram is one thing, actually implementing the functions with real components is quite another.

Although the circuit diagram in **Figure 2** may look large and complex at first, its operation is relatively easy to understand thanks to the previous description of the block diagram. Let's take the sub-circuits one by one.

Preselector

The active element is a type BF961 dual-gate MOSFET, T1, which guarantees minimum loading of the inductors in the preselector. PIN diodes are used to allow the outputs of a decimal counter to switch the requisite inductors on and off. The counter, in turn, is controlled by the microprocessor. For the sake of repeatability, ready-made miniature chokes from the E12 series are used in the preselector. Their Q factors remain as high as possible thanks to the small capacitive load presented by the DG MOSFET. The preselector has six ranges:

1: 150 – 370 kHz
2: 370 – 900 kHz
3: 900 – 2200 kHz
4: 2200 – 5400 kHz
5: 5400 – 13200 kHz
6: 13200 – 32000 kHz

The inductive part of the preselector is brought to resonance by the capacitance formed by a pair of varicap diodes, D14-D13. The varicap control voltage has a range from 0 to 9 V, and is supplied by the wiper of the preselector tuning control, P1.

The gain of the DG MOSFET is controlled in traditional fashion by means of a direct voltage on gate 2. Although the preselector already affords considerable suppression of unwanted frequencies, the MOSFET is followed by an additional low-pass filter with two ‘notch’ sections, L9-C17 and L11-C18, for virtually complete suppression (–50 dB) of image frequencies and out-of band products.

1st mixer and synthesizer
In many up-market SW receivers, a double-balanced type (DBM) is employed as the first mixer to guarantee excellent large-signal behaviour. The main disadvantages of a passive DBM are the high level of the LO signal (typically 7 dBm), and the inherent conversion loss of about –7 dB. The present receiver employs a DG MOSFET in the first mixer. As opposed to a DBM, the MOSFET

Figure 3. The microprocessor control circuit is based on a PIC16F84. To keep receiver-internal interference to a minimum, the PIC is in ‘sleep’ mode most of the time.

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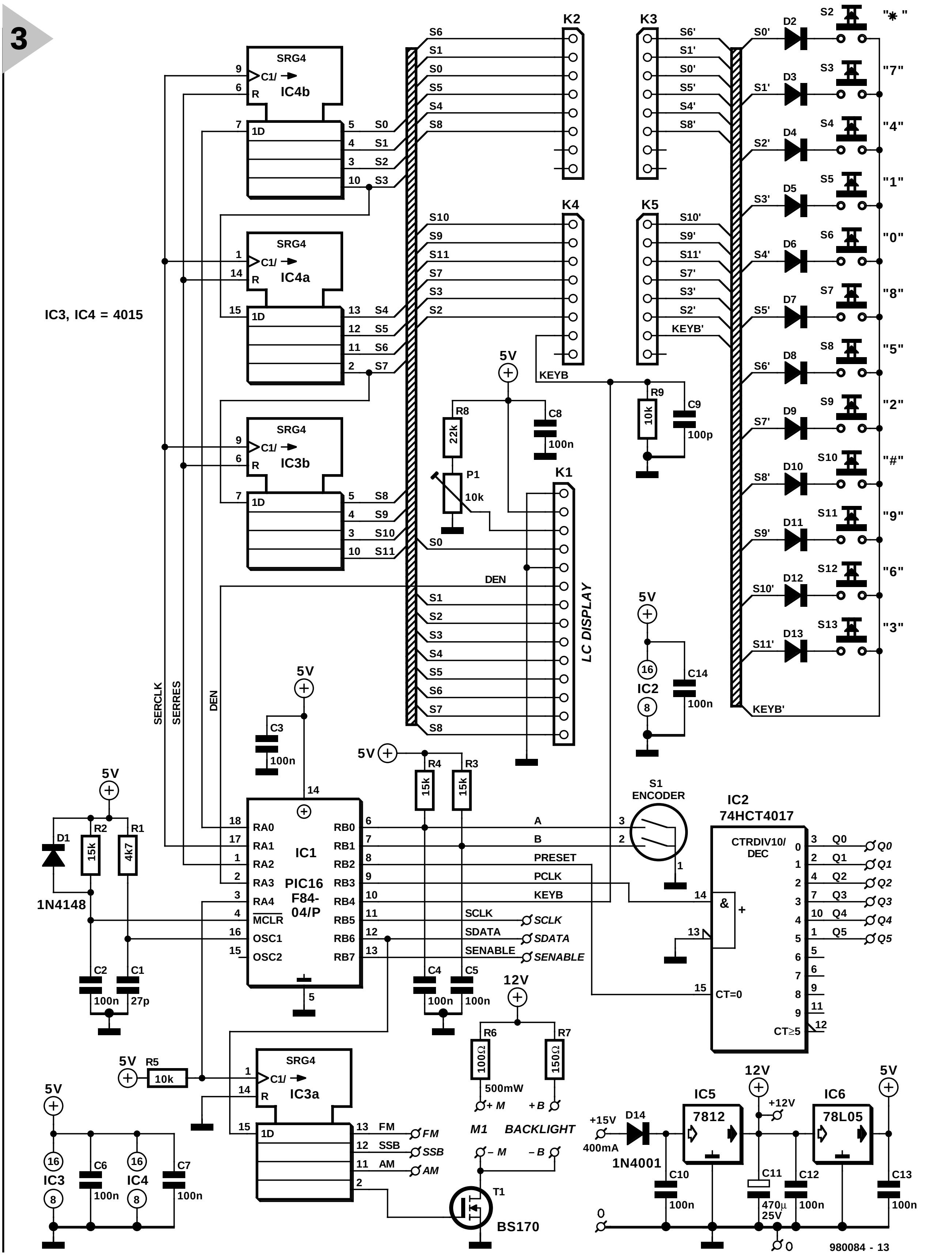

offers a conversion gain of about 10 dB, and it works fine at a relatively small LO signal.

The combination of a synthesizer IC type MC14156-2 (from Motorola) and a dual-modulus (÷128/÷129 or ÷64/÷65) divider type MB501L (from Fujitsu) forms a phase-locked loop (PLL) whose step size equals the reference frequency of 1 kHz, which is derived from quartz crystal X3 by an on-chip divider. The MC14156-2 is controlled by means of serial information supplied by the microprocessor. The error signal supplied by the synthesizer IC is filtered by a loop filter built around an opamp type MC33171 (IC3). Because the 1-kHz reference-frequency component has to be minimized in the filter, the PLL should allow for a relatively long lock time. Here, the largest frequency change of the local oscillator (45.150 MHz to 77.000 MHz) takes about 100 ms. Using the single-ended 'PDOUT' terminal of the MC14156-2 allows the loop filter to be kept simple. The MC33171 is used here because it is capable of supplying a rail-to-rail swing of the output voltage. This is a must if the VCO based on FET T5 is to cover the required frequency range (theoretically, 45.15 MHz to 77 MHz) without 'dying' as a result of a low varicap control voltage. In practice, the VCO is slightly overdimensioned, covering a frequency range of 37-85 MHz with a control voltage of 0-9 V. The VCO output signal is capacitively coupled to the first mixer (T2) as well as to a buffer stage around T6, which is designed to drive the ECL inputs of the MB501L divider chip.

IF amplifiers, AM/FM demodulators and SSB detector

Referring back to the block diagram, the good news is that all sub-circuits between the first IF filter and the output of the last IF amplifier are contained in a single IC, the TCA440. This old faithful from Siemens contains a preamplifier, an oscillator, an IF amplifier, and an AGC with a dynamic range of no less than 100 dB (which is no mean requirement for SW listening). The two 455-kHz IF filters for narrow (3 kHz BW) and wide (12 kHz BW) reception are connected into and out of the TCA440 external circuitry by means of PIN diodes and control signals supplied by the microprocessor. Other filters than the Toko types indicated here may be used as long as their input impedance is 2.2 kΩ, and the respective 3-dB bandwidths are about 3 kHz (narrow) and 12 kHz (wide). The TCA440 drives the S-meter directly via its AGC output. Meters with different sensitivities are accommodated with preset P3.

The injection signal for the second mixer is supplied by the oscillator inside the TCA440. This oscillator only needs an external quartz crystal and a couple of passive parts to supply a rock-steady 44.545 MHz signal.

The SSB detector is built around the familiar NE612 (or NE602), which contains a balanced mixer and an oscillator. The latter is connected to an inexpensive 455-kHz ceramic filter which is 'pulled' by a varicap, D23. The resulting deviation of about ±2 kHz is sufficient for USB and LSB reception (upper/lower sideband) if you turn the BFO control pot.

The FM demodulator is a classic ratio detector with a FET amplifier in front of it. This detector has been designed to supply enough output even if an NBFM (narrow-band frequency modulation) signal is received. NBFM is commonly used in the 27-MHz (11-m) CB band.

The AM demodulator consists of a single diode, D20, which also supplies the AGC drive signal.

The three tuneable inductors in this part of the circuit are all 455-kHz, ready-made types from Toko. These units contain internal tuning capacitors. Other 455-kHz transformers than the ones shown here may be used, as long as the primary-to-secondary turns ratio is 20:1 (in case of L14 and L18), and the tap is exactly at the centre of the primary (in case of L19).

Audio signal sections

Three BS170 FETs are used as analogue switches, feeding either the FM, AM or SSB signal to filter/amplifier T10. The control signals at the gates of the FETs are, again, supplied by the microprocessor circuit. The audio bandfilter is designed for speech at radio communications quality, i.e., roll-off points are defined at 450 Hz and 3.3 kHz to keep out most unwanted noise, and in the case of SSB, neighbouring stations. The LM386 audio amplifier, finally, supplies about 1 watt into 8 ohms, which is good for a small external loudspeaker in your shack, or a pair of low-impedance headphones (preferred by veteran DXers).

MICROCONTROLLER SECTION

The schematic of the microcontroller section in the receiver is given separately in **Figure 3**. This circuit also contains most of the power supply components.

The microcontroller used is the familiar PC16F84 from Microchip. Here, it executes a user program of about 1 kBytes from its on-chip ROM. The PIC controller is supplied ready-programmed by the Publishers.

The on-chip EEPROM is used to store and retain frequencies. Because the processor clock does not have to be particularly stable or accurate, the cheapest clock option, an R-C network (R1-C1), is used. The processor runs at about 4 MHz, however, it is only 'active' when its action is required, for example, when a key is pressed, or the synthesizer has to be reloaded. To keep spurious signals to a minimum in the receiver, the PIC will be 'asleep' most of the time!

Three of the four shift registers type 4015 expand the I/O functionality of the PIC into a 12-bit shift register which is used to drive the keyboard and the LCD. The keyboard is not a matrix type. As indicated by the circuit diagram, each switch has a separate connection, while the other goes to a 'common' rail. Pressing a key causes an interrupt which serves both as a wake-up call and a service request for the 'sleeping' processor. Turning the rotary encoder also generates a hardware interrupt and causes the processor to wake up. The encoder used here is a Bourns type with 24 turns per full rotation. It enables the complete tuning range of the receiver to be covered — just keep turning until the LCD shows the desired frequency, and then carefully adjust the preselector for best reception. Alternatively, type the desired start frequency into the keypad, and tune from there. The rotary encoder is connected directly to two PIC I/O pins. Debouncing is effected by hardware and software.

The remaining I/O pins of the PIC are used to control the serial synthesiser (RB5, RB6, RB7), and the preselector, by way of decimal counter IC2 (RB2, RB3).

The power supply is conventionally based on 3-pin fixed voltage regulators from the 78 and 78L series. Three voltages are supplied: 12 V, two times 5 V, and 9 V. The latter and one of the 5-V supplies are part of the main receiver circuit discussed above (refer back to Figure 2). They obtain their input voltage from the 12-V regulator on the microprocessor board. The heaviest loads on the 12-V rail are obviously the audio power amplifier IC, the S-meter lighting and the LCD backlight (if used). The unstabilized input voltage should be at least 15 V. An inexpensive power mains adaptor may be used, but do note that the receiver may draw up to 450 mA, so go for a relatively powerful adaptor.

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The construction, adjustment and operation of the receiver will be discussed in next month's concluding instalment.