Building a Broad Band Solar Radio Telescope

This is the most complex of the projects presented in this book. However it offers a lot of potential to learn about how receivers work, and as a useful radiometer or even spectrometer for solar flare observation.

Construction of ultra-high frequency circuits is not really suitable for a beginner. However, some of the difficulties usually encountered have been removed by the use of a readymade tuner module that translates the frequencies in the range 51–853 MHz down to a workable 37 MHz. The remaining components required to turn it into a radio telescope you can construct.

The type of tuning module used in this project is commonly found in television receivers, cable TV boxes, and other similar television tuner applications. Suitable units can be removed from used equipment, although they are inexpensive to purchase new if you can find a supplier willing to sell a single unit.

Background

The approach taken here is to break down the radio into its basic modular units. Each module will be treated as a separate entity and built on its own board. The modules are then shielded where necessary and installed into an RF screened box. There are two main reasons for this approach. First of all, each unit can be built and tested separately, so that when it comes to putting it all together, troubleshooting problems will be easier. Secondly, the resulting instrument would be easy to modify later by replacing individual modules, or adding more. For example, the frequency band it operates on can be changed by replacing the front end RF section.

Before we begin to design the instrument, let's outline our goals:

- 1. The instrument should be capable of measuring the strength of a fixed frequency channel, covering a reasonably wide but not excessive bandwidth. In other words, it should operate as a total power radiometer telescope.
- 2. It should be able to scan a range of frequencies in a predetermined time interval and to measure the changes in power levels across the band of frequencies. In other words, it should be able to act as a spectrometer.

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- 3. It should be possible to automate the control of the radio functions, so it can be left unattended to do its work.
- 4. The output of the receiver should be in a form easy to log using either a PC or a stand-alone data logger.
- 5. The frequency of coverage should be a wide as possible.

These tasks may seem daunting. However, following these plans is not that hard, so long as you have patience to persevere when things are not working out right.

The design was inspired by the E-Callisto project, a low-cost solar flare spectrometer designed by the team at ETH Zurich. You may wish to build that instrument as an alternative. The plans may be found at: http://helene.ethz.ch/instrument/ callisto/ecallisto/applidocs.htm.

In Chap. 7, you saw the block diagram of a superheterodyne radio receiver. That is what we are about to design here. The basic superheterodyne will have a single intermediate frequency section, but it can have more. This design has two intermediate frequencies, and is therefore called a double conversion superheterodyne.

Design Overview

The heart of the receiver is a television or multimedia tuner module. This avoids the difficulty of constructing stable high frequency circuits. The television tuner chosen for this project is a CD1316L, manufactured by Philips for the European market. There is a similar model, the CD1336L, designed for the U.S. market, but this project implies the use of the CD1316L because the IF outputs differ on the two models, and the frequency coverage is slightly different. While we are on the subject of tuners we should explain there are many different variations out there.

Older tuners used an analog 2–30 V signal to select the frequency of operation. The CD1316L uses a digital data line to set the operating conditions. The control bus is known as I²C. Most modern tuners use this technique now. Those tuners aimed at the terrestrial television market usually have gaps in their range, because television channel allocations are broken up into several bands separated by bands used for other purposes. The CD1316L is more commonly used for cable television boxes or similar multimedia units, and therefore has continuous frequency coverage, although the range is broken into three switchable bands. The design could easily be modified to use any available I²C tuner, but it would be important to obtain the data sheet to identify any changes that need to be made such as tuning calculations, and to see what frequency ranges are available. Any of the tuner modules could be used as a UHF scanner from about 470–860 MHz.

There is a third flavor of tuner that is less useful for our purposes. This includes demodulators on board providing a video and audio output. This type is sometimes found in computer-based television cards. This telescope design requires access to the intermediate frequency output before any demodulation can occur.

The output of the tuner, the first IF, is a fixed frequency band 7–8 MHz wide in the range 32–39 MHz. Since here we are proposing a much narrower bandwidth than that for back end processing we can select a suitable frequency of 37.7 MHz as our first intermediate frequency with a bandwidth of, say, 330 kHz. By adding switchable filter banks we could offer a selection of bandwidths to suit the application.

The first intermediate frequency is passed to a mixer stage of our construction. The reason a first IF of 37.7 MHz is suggested is that by mixing this with an oscillator signal at 27 MHz, a convenient second IF of 10.7 MHz is achieved. This is a common IF frequency used in television circuits, and suitable ceramic filters are easily obtained. An overtone crystal of 27,000 MHz is also readily available to construct the down converter.

Finally, the second IF can be further amplified and passed to a detector stage, in this case a logarithmic amplifier, the output of which is a DC voltage proportional to the signal power and is easily captured to a PC with an ADC (analog to digital converter).

Construction

The front end in this case is the TV tuner module, which is designed to simply connect directly to an antenna. There is a variable gain amplifier built into the unit that is controllable via a DC voltage that we can easily make manually adjustable using a potentiometer.

Figure 12.1 shows the receiver prototype, and Table 12.1 identifies the pin functions. Pin number begins next to the RF input sockets. The only pins needed here are 2, 5, 7, 8, 9, 11, 14, and 15. The IF output pins provide a balanced output from pin 14 and 15, or you can use an unbalanced output from either pin 14 or 15 and ground. The ground connections are four tags on the four corners of the base that form part of the case.

The I²C needs some introduction. It is a databus technology introduced by Philips and is found in a number of modern IC devices such as EEprom memory chips. It defines the way in which communications of digital data are handled and is a form of serial interface bus. The data is sent via a two-wire system. The lines are identified by the SCL and SDA references. SCL refers to the "clock" line and SDA



Fig. 12.1. To the left is the tuner module, to the right is the home built mixer oscillator. The small board upper centre is a USB to L2C interface.

 Table 12.1. Tuner module pin numbers and their function

Pin number (from the antenna socket)	Function				
1	No connection				
2	+ 5 V DC at 80 mA max				
3	No connection				
4	No connection				
5	RF gain control 0.4—3.3 V DC				
6	No connection				
7	I ² C address select				
8	I ² C SCL				
9	I ² C SDA				
10	4 MHz reference output $-$ not required in this design				
11	+ 5 V DC at 110 mA max				
12	ADC input — not required in this design				
13	No connection				
14	IF output				
15	IF output				

Table 12.2. Five bytes of data are required to program the tuner - see text

Name	Byte	Bits (mo	Bits (most significant bit first)								
Address	ADB	1	1	0	0	0	CA1	CA0	R/W		
Divider byte 1	DB1	0	N14	N13	N12	N11	N10	N9	N8		
Divider byte 2	DB2	N7	N6	N5	N4	N3	N2	N1	NO		
Control byte	CB	1	СР	0	T1	TO	1	1	0		
Band select	BB	0	0	0	0	0	P2	P1	PO		

ſable	ə 12.3.	The voltage	e on pin 7	7 defines t	he L2C address
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CA1	CAO	Voltage applied to 7 (V)
0	0	0
0	1	Terminal open
1	0	2–3
1	1	4.5–5

to the "data" line. Pin 7 is not really part of a standard I²C bus, but it is used in this device to define the address to which the bus will respond. The tuner can only be set to one of four addresses. Table 12.2 illustrates its function.

To set up the tuner, the address and four data bytes are sent to it. The bytes are made up from binary bits, but are sent to the tuner in hex format.

Since we're only using a single tuner, it is easiest to ground pin 7 to the 0 V power line; this defines the address as binary 1100000(*). Bits 2 and 3 (CA0 and CA1) of the ADB byte are then 0. The last bit tells the tuner whether we want to read or write data to it. The R/W bit is 0 when writing data to it, so the first byte becomes 11000000, which in Hex is C0. This will not change. See Table 12.3 for addressing options.

The second and third bytes DB1 and DB2 define the frequency of operation. A single step relates to a change of 62.5 kHz in tuning frequency. The following formulae is used to calculate the byte values for a given operating frequency.

Table 12.4. Bits PO, P1 and P2 of the last byte BB define which of the three bands the tuner will operate.

Band	Minimum frequency (MHz)	Maximum frequency (MHz)	P2	P1	PO
Low	51	171	0	0	1
Mid	178	450	0	1	0
High	458	858	1	0	0

$$N = \frac{f_{in} + f_{if}}{62500}$$

where f_{in} is the input frequency and f_{if} is the required IF (37.7× 10⁶ in our case).

The resulting N needs to be rounded to the nearest whole number and expressed as a hexadecimal value. The minimum value for N is 1419 decimal or 58B in hex where DB1 = 05 and DB2 = 8B. Do not set the frequency to lower values than this, and do not try to exceed the upper value representing 858 MHz.

The control byte should have T1 and T0 set to 0. CP sets the charge pump value. For fast tuning CP should be set to 1, and if the frequency is going to remain fixed for a while, should be changed back to 0. In spectrometer mode the binary value of CB is 11000110 or C6. For the radiometer user it should be 10000110 or 86.

The final byte of data is the frequency band selection. Table 12.4 shows how this is broken up into three parts.

To set low band the byte BB is 00000001 or hex 01, mid band is hex 02, and high band is hex 04.

Now let's test the tuner. For this you will need a spectrum analyzer. The best value analyzer to start with is a computer-based USB oscilloscope adapter with an FFT function. One model from China, that is sometimes available on eBay is a DSO 2150. The bandwidth is a decent 60 MHz; this makes it very useful for testing and analyzing receiver IF circuits. Other models are available with up to a 200 MHz bandwidth.

In order to actually send the data to the tuner we need some form of interface hardware. This is usually done with a microcontroller. A microcontroller is a form of single chip computer. If the bug really bites, and you get involved in more hardware development, it is recommended you take the time to learn about using and programming microcontrollers, such as the range available from Microchip known as PIC's, or the range available from Atmel. However, a whole book could be dedicated to this task alone, and many are available. So to ease that extra burden we recommend experimenting with a serial to I²C or a USB to I²C interface, and use your PC to control the receiver. You can purchase a BV4221 for development testing from a company called ByVac Electronics (see http://www.byvac.com). The drivers supplied with it install it as a virtual COM port on the PC, and a simple terminal emulator was supplied on a CD called "bv_com_a.exe," which is highly recommended for this job.

Now connect three wires from the I²C adapter, the ground wire to the tuner case, and the SCL and SDA lines to pins 8 and 9, respectively.

Next connect the positive lead from a 5 V regulated power supply to pins 2 and 11, and the power ground to the case of the tuner.

Normally the receiver gain is controlled automatically by external circuitry. However we have already seen that we need to manually control the gain. For this purpose a 0-3.3 V variable voltage is supplied to pin 5. This needs to be derived from the 5 V supply using a potential divider (refer to the Chap. 8). Use a potentiometer to vary the voltage, along with a fixed resistor to limit the maximum voltage to 3.3 V. For example if you chose a 1K potentiometer R2, the fixed resistor R1 will be

$$R1 = \left(\frac{R2.V_{in}}{V_{out}}\right) - R2$$

This gives

$$R1 = \left(\frac{1000x5}{3.3}\right) - 1000$$

This means R1 is 515 Ω . Chose the nearest fixed resistor value higher than the calculated value which is 560 Ω in standard ranges. Connect the fixed resistor between the 5 V supply and one end of the potentiometer, and connect the other end of the potentiometer to ground. The variable wiper output is then connected to pin 5 of the tuner.

Check all connections carefully. Then connect the BC4221 adapter to the PC and open the terminal software; set the speed to 115,200 baud and the COM port installed by its driver and click on the button to the left of the COM port box. The icon will turn from red to green. Press the enter key and you should see a hex address appear. Apply 5 V power to the tuner, and connect an FM radio antenna (in practice a piece of wire is probably sufficient for testing).

To test the tuner chose a strong FM broadcast channel and determine the divider bytes DB1 and DB2 for that channel. You might choose a channel called Smooth Radio at 106.6 MHz, a strong FM radio channel in the UK.

ADB = C0

DB1 and DB2 are $(106.6 \times 10^6 + 37.7 \times 10^6)/62500 = \text{decimal } 2308$, so DB1 = 09 and DB2 = 04 hex

CB = C6

BB = 01

In the terminal software window type the following string: s-c0 09 04 c6 01 p and press the enter key. The "s" command tells the adapter to be a master controller and open communication to address c0. The four data bytes follow and the p closes communication. The tuner should now be locked onto a frequency 106.6 MHz converting it to 37.7 MHz at the IF pins 14 and 15. To check it is working, connect channel 1 of the oscilloscope to pin 14 and ground and turn on the FFT. Adjust the settings so you see 37 MHz roughly central in the FFT window. You should see a peak at around 37.7 MHz, which is the radio transmission. Alternatively you can use an RF signal generator and try it on other frequencies. Adjust the RF generator frequency to see the peak move sideways in an 8 MHz window (the bandwidth of the tuner output). The spectrum from my tuner is shown in Fig. 12.2



Fig. 12.2. FFT spectrum plot of the tuner output showing a peak at 37.75 MHz. The signal is from an FM radio station at 106.6 MHz.

Tab	le	12.5.	When	reading	back	information	from	the	tuner	only	2 b	ytes	are	neede	d.
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Name	Byte	Bits (most	significant firs	t)					
Address	ADB	1	1	0	0	0	CA2	CA1	R/W = 1
Status	SB	POR	FL	1	1	0	AD2	ADT	ADO

Finally you can read data from the tuner, but only one byte is returned. However there is no way of confirming what frequency it is tuned to. Table 12.5 shows the format of the Status byte.

When the address sent to the tuner has a 1 added in the least significant bit position, it tells the tuner to return the Status byte. The POR flag is set to 1 if the tuner has not been configured since a power on reset; otherwise it is 0. The FL flag returns a 1 if the phase locked loop inside the tuner is locked. The AD2, AD1, and AD0 bits return the value of the internal analog-to-digital converter. Mostly there is little value in reading the Status byte, although sometimes it would be useful to confirm that the tuner has locked after a request to tune a channel.

The Second IF Stage

The IF output of the tuner can be treated as either a balanced or an unbalanced output. In this project we will use the single output – it does not matter which one. It is always a good practice after a mixer stage to use a filter to remove unwanted products. These TV tuners are designed to drive a SAW filter (surface acoustic wave). But you can try making your own. Filter design is covered later.

The next stage is a frequency mixer, which will reduce the first intermediate frequency to 10.7 MHz. This means an ordinary HF communications receiver could be used to help set it up and test it. It also means useful ceramic filters are easily obtainable for 10.7 MHz, in a wide variety of bandwidths.

A mixer circuit is a fundamental building block of nearly all radios. So this part of the construction process is a useful learning exercise. The mixer itself is a readymade unit by Minicircuits, the part number being SBL-1, and can be ordered online direct from Minicircuits or from other RF component stock lists. A mixer combines the RF signal with a local oscillator, to produce two outputs – the sum of the two inputs and the difference of the two inputs. The SBL-1 is an example of a double balanced mixer, meaning the original radio frequency and the local oscillator frequency are strongly suppressed and don't form a part of the output to any significant level.

The schematic of the oscillator is shown in Fig. 12.3; it is a Colpitts type.

Note here that the Crystal X1 merely controls the frequency. The inductor L1 and capacitor C1 form a series tuned circuit that also resonates at the desired frequency. If the crystal is removed, the oscillator should still work at approximately the correct frequency, but it will be a lot less stable and much more temperature sensitive. The values of L1 and C1 are calculated to resonate at 27 MHz (see Chap. 8).

The first transistor is essentially an amplifier. However some of its output is fed back via C2 to the base pin and is re-amplified. The amplifier is set up to be unstable, therefore creating the oscillations we desire. The output of the oscillator is fed via C5 to a buffer amplifier, to isolate the oscillator from the mixer to avoid loading problems.



Fig. 12.3. Colpitts oscillator.



Fig. 12.4. A mixer built dead bug style.

Note the series tuned circuit formed by L1 and C1 is not only resonant at 27 MHz, but also presents low impedance at resonance. Together with R1 this damps out the chances of the crystal resonating on its fundamental frequency, and forces it to resonate at the chosen design overtone. When building the oscillator the crystal can be removed and L1 and C1 tweaked to produce near to the design frequency. A digital frequency counter or an oscilloscope is needed for this task. Once set up return the crystal to its position and the counter should confirm a solid frequency lock to 27,000 MHz (Fig. 12.4).

The mixer used is a Minicircuits SBL-1+. Its pin configuration is shown in Fig. 12.5.

The output of the oscillator is connected to pin 8. The oscillator ground is connected to pins 2, 5, 6, and 7. The output from the tuner is connected between pin 1 (ringed in blue) and ground. Pins 3 and 4 are connected together. Finally the output is taken from the junction of pins 3 and 4 to an RF connector or to the next stage where the body of the connector is also grounded.

This author's prototype mixer was constructed dead-bug style. A photograph is shown in Fig. 12.4. You could, of course, etch a circuit board instead for a more professional look. I would suggest you try both examples of construction. Build one module ugly style and another by etching a board as part of a learning process and then decide for yourself which you prefer. The board layout is in Fig. 12.6.

The holes for drilling for drilling are shown as circles, attach the overlay drawing to the top of the board with tape, and drill the component holes carefully. Most holes will be about 0.6 mm in diameter, but the trimmer capacitor and pad holes will be more like 1 mm.

X2 is the IF output, X1 is the RF input, refer to Fig. 12.6 text for power connections. The board should be mounted into a metal box, preferably a die cast type ensuring the box is well grounded to the circuit ground.

Pin Connections

LO	8
RF	1
IF	3,4^
GROUND	2,5,6,7

* pins must be connected together externally



Outline Drawing

Fig. 12.5. Pin configuration of Minicircuits SBL-1+.

The IF Filter

The down converter circuit illustrated here outputs 10.7 MHz. This is a common IF frequency used in television and commercial radio receivers, so there are many filters available readymade to clean up the output of the mixer before it is processed by the next stage. These come in ceramic or crystal forms, but are simple – a package with usually three pins, although some crystal types come with four pins because they have two ground pins. Murata makes a range of them with varying bandwidths of a few kHz to over 400 kHz. In practice use these wherever you can. However for more general filtering needs where filters are required to operate at custom frequencies or non-standard bandwidths you can build your own using groups of inductors, capacitors, and sometimes resistors. This is covered later.



Fig. 12.6. (a) Mixer board including component layout. Note: power connections are not marked. Positive 12V is applied to the track connecting R2,R4,R6 and R8. Ground is the track connecting R1,R3,R5,R7, and R9. Confirm this by comparing with Fig. 12.3 (b) Mixer board ready to print onto heat transfer. The layout suggests the BNC sockets are attached to the board. In practice use chassis mount sockets and connect to the board with a short length of wire.

In the prototype for this project a Murata 3 pin ceramic 10.7 MHz filter with a 330 kHz bandwidth was used. Since this mixer is passive, there is a loss of around 7 dB. So it is usual to add an amplifier stage after the mixer. The amplifier used was based on a MAR6 chip, also available from Minicircuits. The circuit is shown in



Fig. 12.7. (a, b) An RF amplifier used at the output of the mixer, constructed on a small piece of strip board. The pair of resistors made 660 Ω for a 14 V supply. The signal is input via a 3 pin ceramic filter (*lower center*). Note that a piece of copper was removed from beneath the Mar6 chip.

Fig. 12.7. The input pin 1 is marked with a small dot. It is a very small surface mount component, but can be soldered onto the board with a fine-tipped soldering iron while holding the chip down with a miniature screwdriver. The prototype was constructed on a small square of strip board. A small circle of copper was first removed from around one hole, and the chip was soldered such that an earth strip bordered both sides, and the input output pins straddled the broken strip.

The bias resistor is chosen to suit the power supply; the values can be found in the data sheet obtainable from the Minicircuits website. The gain is around 22 dB, which is more than enough to overcome the loss in the mixer stage.

The Logarithmic Amplifier

The final stage is the logarithmic amplifier. It is based on a series of amplifier modules, but its purpose is not to amplify a signal but to be a measurement device. The output of a log amp is a DC voltage whose value is directly proportional to the power of the RF signal applied to it. The output is scaled logarithmically. This offers a way of compressing a signal with a wide dynamic range into a compact logarithmic scale, which is easier to view. Since I'm recommending the Analog Devices AD8307 chip we will not go into its internal function in any detail. If you

want to know more about it download the data sheet from the Analog Devices website. You should treat the chip as a "black box," a device that you can apply in practice but don't necessarily understand what goes on inside.

The AD8307 is a very simple device with only eight pins, and you can get it in the through-hole style; a suffix of N on an IC chip normally means through hole pin style. The package is also known as PDIP, for plastic dual inline package. The Table 12.6 details the pin functions. IC pin number 1 on any PDIP IC can be identified by either a small circle marked beside it, or by a semicircular indent in one end, where if the indent is held uppermost, pin 1 is on the top left. The numbering is always down the left side and continues in a vertical manner up the right side, so pin 8 is opposite to pin 1 in this case (Table 12.6).

The AD8307 requires very few external components to function. It has a dynamic range of 92 dB and will operate with signal power levels from -78 to +17 dB. The output voltage will vary from 0.25 to 2.5 V, so that 25 mV represents 1 dB change in signal power, assuming a power supply of 5 V. This would drop to 15 mV/dB at 2.7 V. Since the tuner requires 5 V, it makes sense to supply the log amp from the same power supply. Therefore an 8-bit data logger is perfectly adequate (see Chap. 13).

The Log Amp Detector Circuit

The circuit diagram is shown in Fig. 12.8.

The PCB layout for the log amp is shown in Fig. 12.9. Pad 1 output and ground can then be fed to an analog-to-digital converter for logging the data, or to a digital voltmeter for system testing.

If this module is built in a separate box of its own, it can be used to interface other instruments you make. It will work from audio frequencies up to 100 MHz to full specification and will continue to operate up to 500 MHz with slight reduction in performance. So this could be used on its own connected to the audio output of a conventional radio receiver as a way of logging changes in signal power.

Antenna Choice

If the tuner front end was restricted to the UHF band of operation, then the easiest solution would be to buy a high gain television antenna. If the full range of frequencies were required then a log periodic array would be more suitable, at the expense of reduced antenna gain.

Pin number	Mnemonic	Description					
1	INM	Negative signal input					
2	COM	Common pin usually grounded					
3	OFS	Offset adjustment					
4	OUT	Logarithmic output voltage 25 mV/dB					
5	INT	Intercept adjustment					
6	ENB	Chip enable, turn high to activate it (apply power supply voltage)					
7	VPS	Power supply 2.7–5.5 V					
8	INP	Positive signal input					

Table 12.6. The AD8307 log amp pin connections



Fig. 12.8. Logarithmic amplifier/detector.

DIY Filter Designing

To design a filter from scratch involves a fair bit of tedious mathematics. Not that the mathematics is very difficult. It's just that there is a lot of room to make a mistake. We will not subject you to an early evening's sleep by trying to explain the mathematics here, but there is a rather good explanation of one method of filter design in the book *RF Circuit Design* by Chris Bowick. What is recommended is to download a cool piece of freeware software simply called Filter Design. You can get it from AADE via the website http://www.aade.com. There are lots of free filter design tools and some decidedly not so free ones, but this is one of the best. When you start the software there is a small advertisement for AADE's own LC meter, which is an invaluable tool in its own right.

Once you open the program you get a window with a help box displayed. Start by selecting the design menu, and then Butterworth. We will not go over all of the features of this software – you can of course explore it more fully yourself. But let's cover an example design the way you might intend to use it.

The terms Butterworth, Chebyshev, etc., refer to design styles; each has its own particular characteristics. Butterworth types tend to have a very flat pass band response, while Chebyshev types can show a fair amount of ripple. By selecting Butterworth you are asked what kind of filter you wish to design, bandpass, low pass etc. Select bandpass, so we can design a filter for the output of our mixer, and press OK. You are now presented with a dialog box, as in Fig. 12.10.

The box CF is where you define the center frequency, in this case 10.7 MHz. BW is for bandwidth. Now, making very narrow band filters is not a good idea using Butterworth principles, but that's OK. Let's chose 500 kHz. Next set SF, or shape





Fig. 12.9. (a, b) Logarithmic amplifier PCB layout.



Fig. 12.10. Dialog box from Butterworth.

factor. When you click in the SF box help appears in red under the graph that explains what it is. SF defines how fast the roll off is in the design.

Very steep-sided designs are unreasonable, but the F1 and F4 frequencies are chosen to define the attenuation required at a specific distance away from center frequency. Now there is an amateur radio frequency band operating at up to 10.15 MHz, so let's make sure we attenuate that strongly. The F2 is now 10.7–0.25 MHz, which is 10.45 MHz, and F1 is 10.15 MHz, so SF is F2/F1 = 1.03. APass is chosen to the point where the power is reduced by 3 dB, so enter 3 there. AStop is how much attenuation we want at frequency F1, so let's make that 30 dB as a first try anyway, so enter 30 in the box. Leave order blank; the software will decide this value. Order effectively means how many components are required. Finally enter the input and output impedances in ohms. These can be different, but keep them to 50 for this exercise. (As an aside, note that a filter can also be used to match the impedances of dissimilar stages.) Now press enter. And enter again for a lossless design. You get the chance to name your design, and then the filter is shown as a schematic with component values. That's it – job done! Hmmm, maybe not. For one thing the values of the components are unlikely to exactly match real-world values.

Next step is to analyze the design, so click on the Analyze menu, and select the top item power gain. In the resulting dialog box set the upper and lower frequencies for the analysis. In this case say 9 and 12 MHz, and set the attenuation range from 0 to 50 and click enter. The resulting graph should look like Fig. 12.11.

The graph represents the attenuation of RF power over a range of frequencies. Unwanted mixer products will be removed from the output, and a good degree of attenuation is applied to potential sources of nearby interference.

Now examine the component values calculated. Figure 12.12 shows the schematic and the values.

Since you can't make your capacitors but you can make your own inductors, you could try changing the capacitor values to those that you can buy and re-analyzing the design to see what effect it has. Then adjust its neighboring inductor value to bring the power gain plot back to where it was. Do this one component at a time, using only small changes all through the circuit. The resulting inductors can be constructed using toroid cores in the manner explained in Chap. 8. However some designs can be very sensitive to tiny tolerance changes in values. It is not a good



Fig. 12.11. Attenuation of RF power over a range of frequencies.



idea to design these simple filters, which have a very narrow bandwidth and a sharp cutoff. Be realistic about bandwidth and roll off. Experiment with the software until you find a design that is practical.

When building the filter, use screening plates between each dipole of the circuit. If the filter is constructed "ugly" style onto double-sided copper board, then pieces of double-sided copper board can be soldered to the base, mounted vertically to act as screens.

Use this technique for preselectors, mixers, IF amplifiers, and even audio circuits. These can all be analyzed for practicality before any components are even purchased.

Converting to a Spectrometer

There are at least three basic ways to build a radio spectrometer: the scanning receiver, the filter bank receiver, and the multichannel receiver.

The scanning receiver is the simplest. In the case of this project software can be written on the PC that causes the tuner to scan through its range. A single scan will provide a radio spectrum of the range of frequencies used in the scan. This project is very much an experimental bench test. PC software discussion is beyond the scope of this book. It is left to the reader to explore ways of developing software for this purpose. If you refer to Chap. 13 where the Arduino hardware is discussed, not only is this a suitable solution for data logging the output of the log amp but the PC software Processing should make it easy to code applications capable of tuning and receiver and to make it scan frequency ranges. To help learn the Processing language the book called *Processing: A Programming Handbook for Visual Designers and Artists* by Casey Reas and Ben Fry is highly recommended.

The filter bank spectrometer simply sends a copy of the signal to a bank of filters, so the output from each filter is a small part of the whole spectrum, and the group of filter outputs make up the complete spectrum. The advantage of this technique is that a spectrum is built up instantaneously, where a scanning receiver takes time to build up the complete spectrum. This is not suitable for this receiver due to its restricted bandwidth.

The final type is the multichannel receiver. A group of identical receivers could be set up on different channels to instantaneously record signals, avoiding the time-delay problem of a scanning receiver. It would not be a big advantage to have large numbers of receivers on contiguous channels, but it would be useful to have 6–8 set up to cover channels evenly spread across their operating range. Since many solar flare events involve bursts of radio energy several megahertz in bandwidth drifting in frequency, the passage of the burst could be followed in the data from the separate channels. Many of the recommended data logging devices have 6–8 channels available, so only one logging device is needed in this project. The total cost of one of these receivers is fairly small, so it is a viable way of studying the time and frequency dependence of solar flare events.