Stereo-code filter

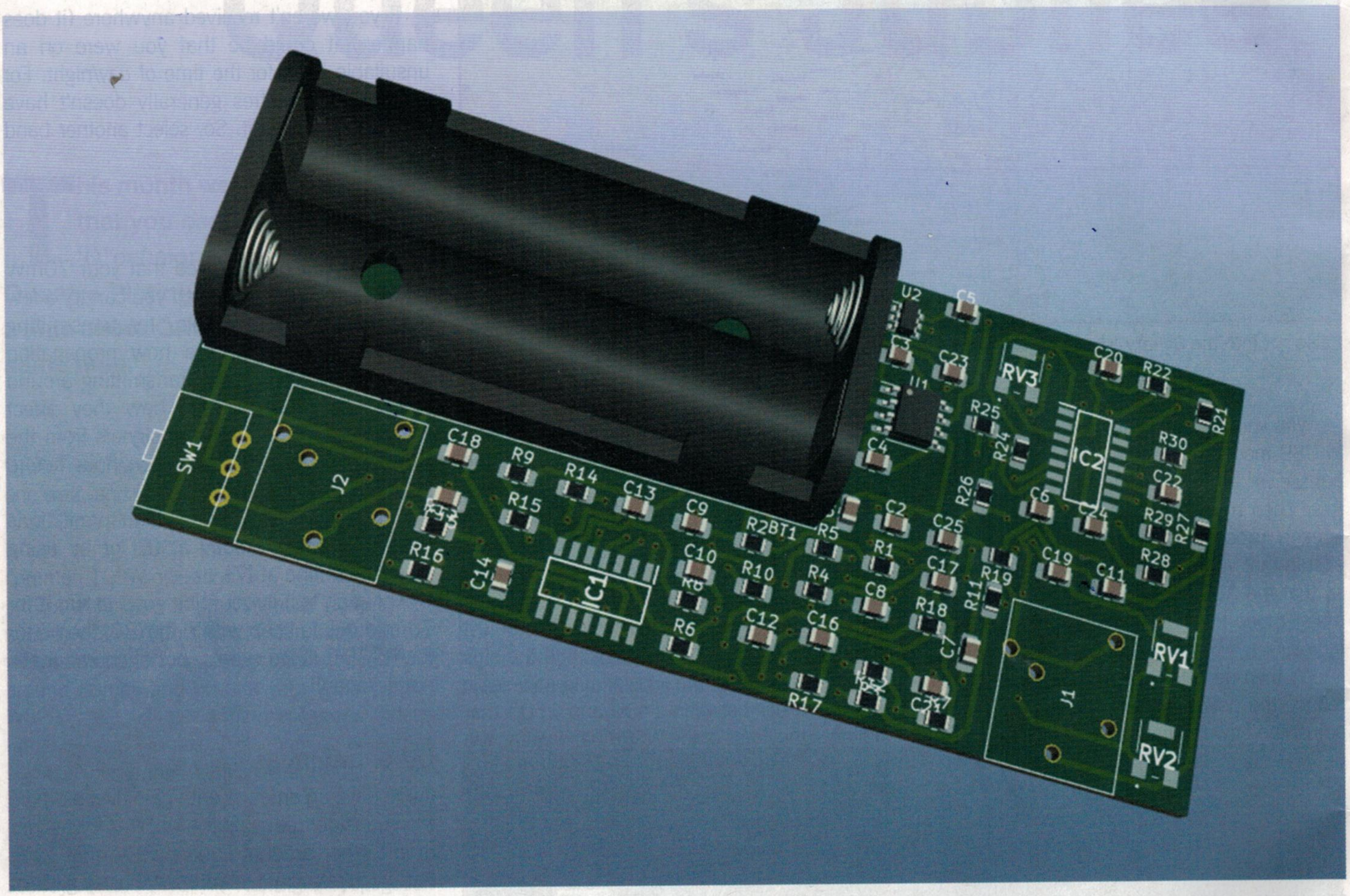

PHOTO 1: The 3D layout of the PCB.

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In the September 1975 edition of *RadCom* magazine [1], the RSGB published an article entitled "Subjective Selectivity and Stereocode" by Dud Charman, G3CJ and R Harris, G3OTK. This article discussed the 'cocktail party' effect where the ears and brain can work to distinguish conversations from different places in the room using the source spacial separation providing small time differences of arrival at each ear. They postulated that this could be used to separate Morse signals of different tone if the spacial time differences could be synthesised in an audio processor. They developed an active filter with a channel for each ear with amplitude shaping and group delay shaping to emulate motion as a source frequency is swept across the audio band and so provides this aural effect.

Some years after publication, having taken out my G4UAZ call (that I now have swapped back to G8CQX!) and become a Morse enthusiast, I built a version of their design on Veroboard. The effect had to be heard to fully appreciate it. It is like the signal you have tuned to the centre of the passband is in the middle of your head and others are off to the sides. Loud signals appear closer so you get a 3D sensation. I stopped using a narrow CW

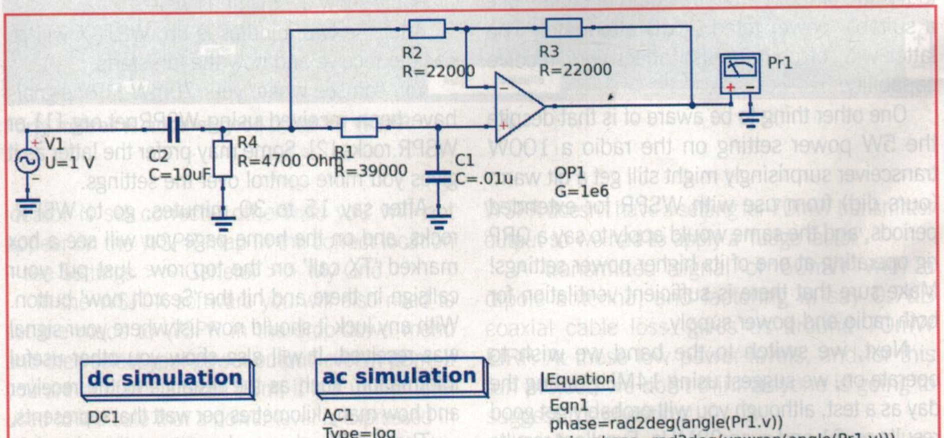


FIGURE 1: The first order schematic.

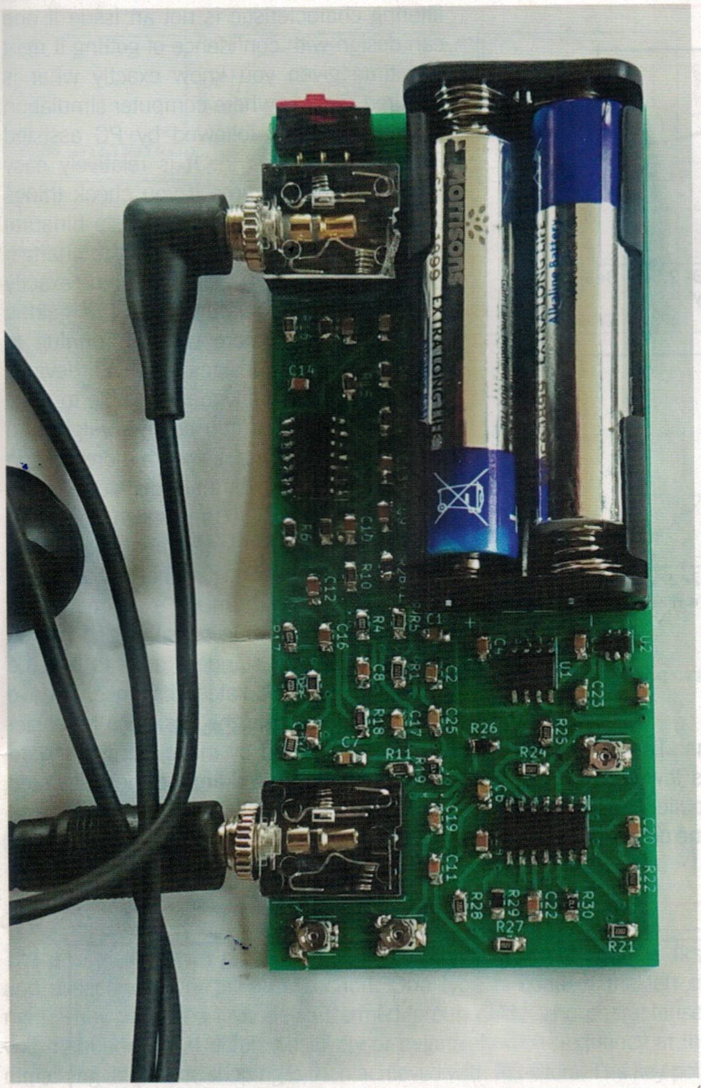


PHOTO 2: The finished project.

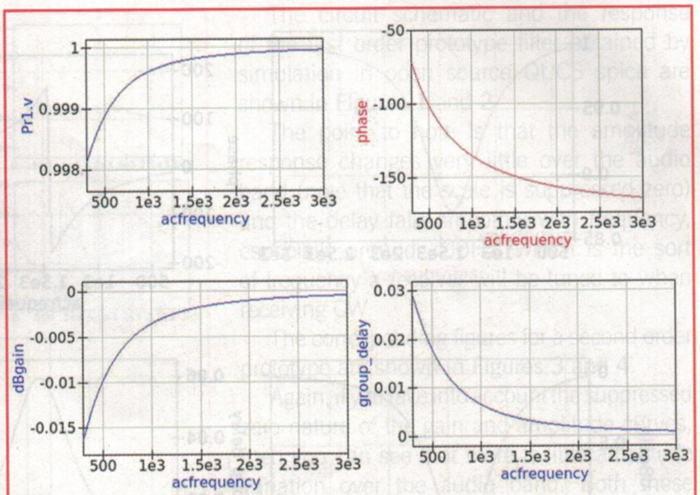


FIGURE 2: First order response graphs.

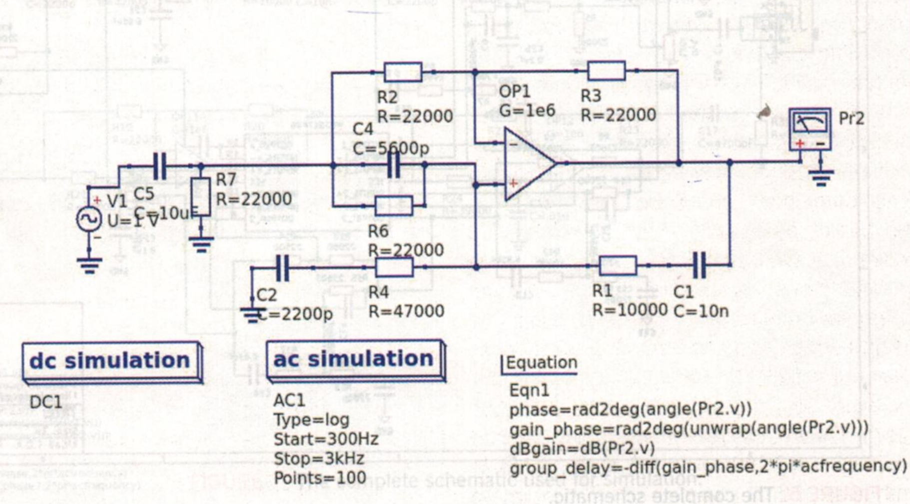
filter preferring to experience the full passband whilst concentrating on the wanted signal, which is easy and not at all tiring for long periods of listening when heard through the stereo-code filter. Sadly, I eventually lost it to a power supply issue and have always hankered after a replacement but couldn't bring myself to re-build it in prototype Veroboard form because of the large amount of effort needed. I considered what a digital implementation would require. If one were to take the amplitude and phase responses of this filter's two channels, then Fourier transform to get the impulse responses, one could use them to provide the weighting coefficients needed for the Finite Impulse Response (FIR) digital filters to do the same job. It would probably need quite a lot of processing power as compared to the modest low powered op-amp active analogue filter. That is fine if you want to be able to flexibly tune the response by changing the weights but, in this case, it isn't required.

Digital implementation

For example, we shall see later we need to implement an audio tuned circuit in each channel as well as a number of low pass elements. The FIR filter will need a digital delay line of a fraction of a second, so one could very roughly estimate the minimum processing requirement might be in the order of 100k 16bit or preferably 32bit floating point multiply-accumulate operations per sample. That does seem easily doable on a modern low power DSP device like a TLV320. There are dedicated IIR and FIR blocks in many devices that might be usable. The FIR approach would have a noticeable delay though, which would be a nuisance for slick break-in contest working. Alternatively, one could take the transfer functions in Laplacian operator form from the original article, parametise and perform a bilinear Z transform to get the z domain function. From those, the coefficients for Infinite Impulse Response filter section implementation can be got. It should be easily within the processing capability of such devices. There are then issues of rounding, scaling and not the least, stability to be tackled as they arise, so this is not a trivial exercise and would take me a lot longer to achieve.

This sort of processing is also within reach of devices like the Cirrus Logic CS497014, which has uses in things like ear-bud noise reduction. An evaluation kit to start writing software for either of these devices would be hundreds of pounds to buy though not dissimilar to the Texas Instruments device kits. This dominates my thinking for a

FIGURE 3: The second order schematic.



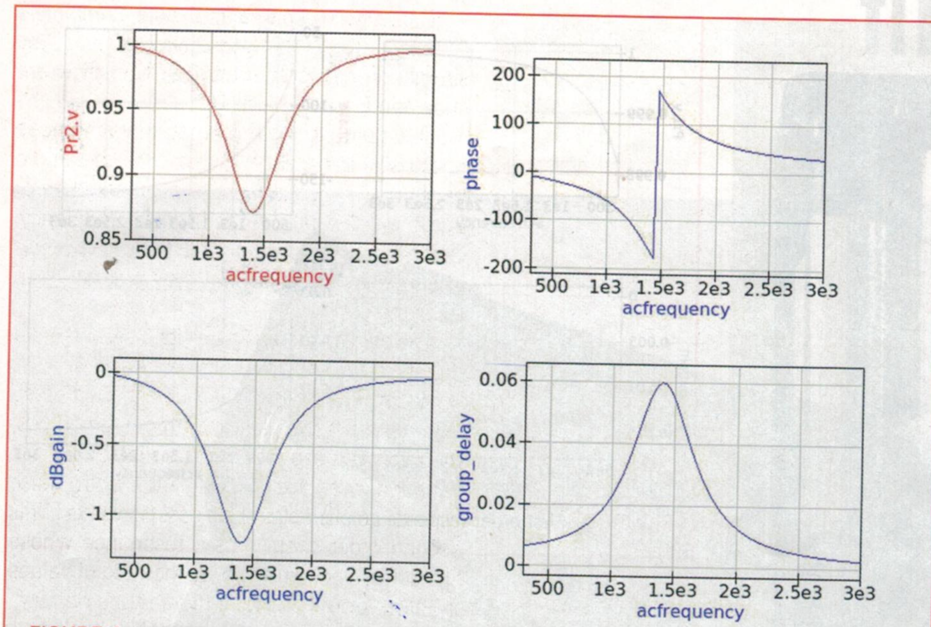


FIGURE 4: Second order response graphs.

hobby project rather than the £10(ish) device cost .

The advantage of a digital solution is one can modify the filtering characteristics easily and add other functions. The advantage of the analogue solution is lower power, lower component cost, easy build and no need to program. The fact that one cannot change the filtering characteristic is not an issue if one can design with confidence of getting it right first time given you know exactly what is required. That is where computer simulation of the schematic followed by PC assisted PCB design comes in. It is relatively easy to get it right first time if you check things carefully this way. And the simple binaural stereo-code function is not going to change. So the active analogue filter gives me exactly what I want without the hassle of getting a development system up and running to develop new filter designs for a DSP device.

So, an active analogue filter able to drive a pair of earphones can be small, light and battery powered. And now with more time on my hands, I have eventually decided to have a go at modernising the design. I started by checking the fundamentals using QUCS Spice simulation. The original authors had not the means to do much experimentation with component value sensitivity so had to specify some non-standard ones to be sure of getting the calculated response. I have been able to update the design to be easier to source and build by make using standard values. I experimented around the calculated values with nearest standard E12 values and checking it is still a good response. I had no

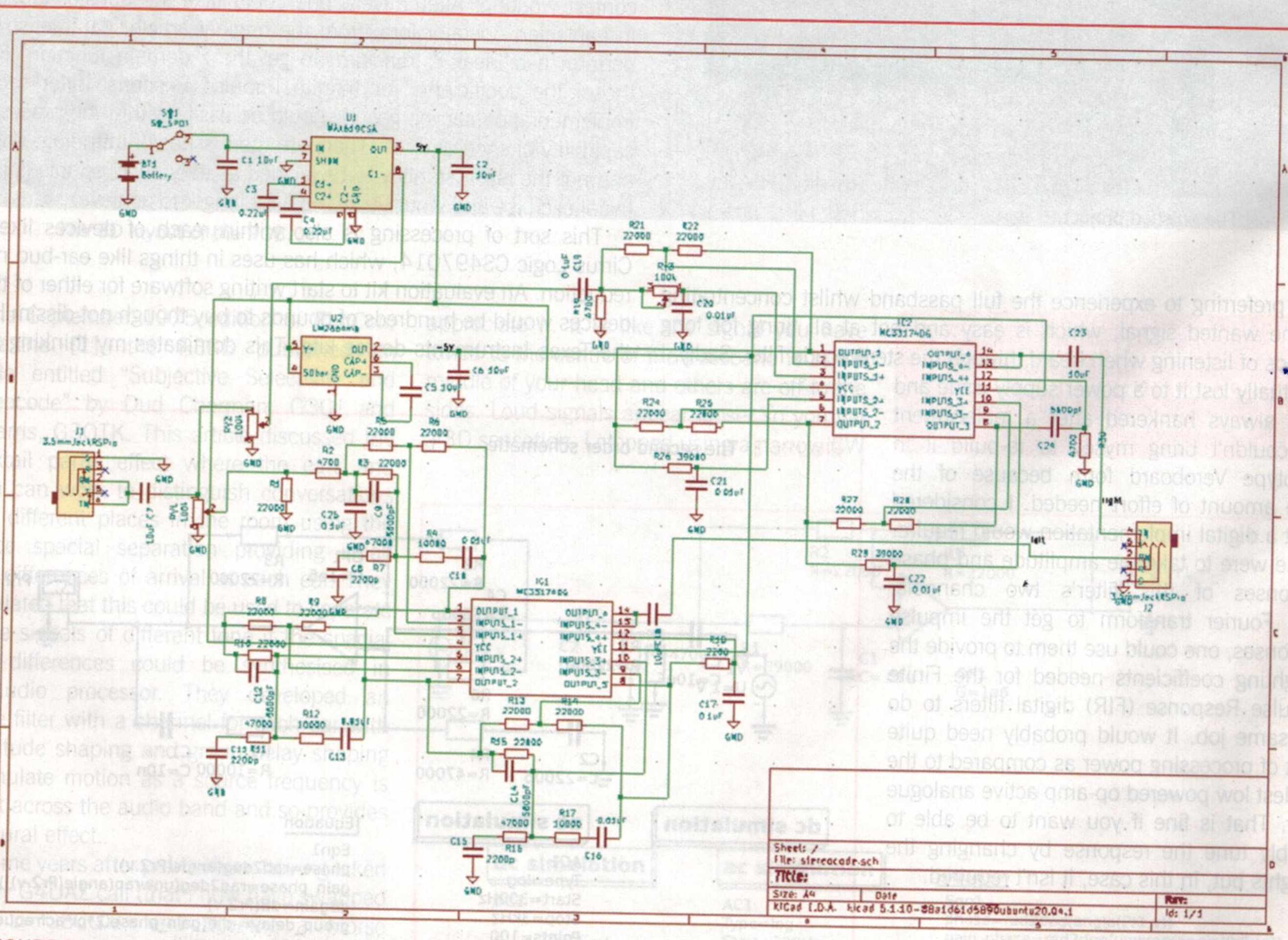


FIGURE 5: The complete schematic.

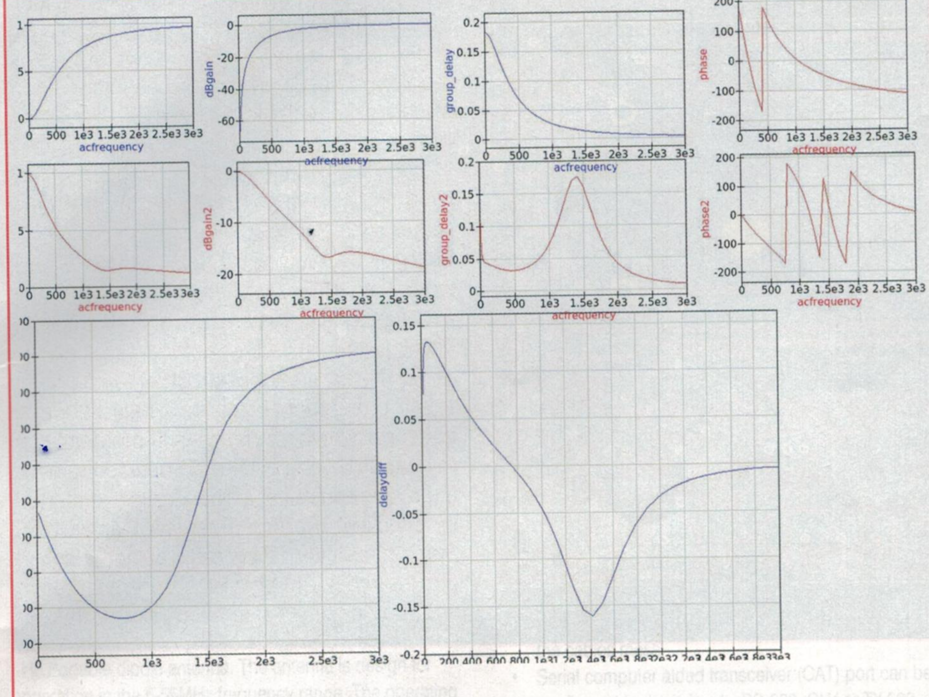


FIGURE 6: The simulated responses.

problem getting as good a response without having to combine components to do it.

Two prototypes

The design uses two prototypes for the shaping of group delay, a first order all pass section and a second order all pass section. One type is used in the left channel and one in the right. They provide a time delay (termed 'group delay' and characterised by the rate of change with frequency) that differs positively or negatively with frequency at the tuning frequency. The remaining amplitude shaping is then achieved with RC filters. The key point is to achieve a relative group delay difference between the two filter channels which alters with frequency in the same way a source delay is presented to a pair of ears as the source transits across in front of the head.

The circuit schematic and the response of the first order prototype filter obtained by simulation in open source QUCS spice are shown in **Figures 1** and **2**.

The point to note is that the amplitude response changes very little over the audio band (note that the scale is suppressed zero) and the delay falls smoothly with frequency, especially around 700Hz, which is the sort of frequency a receiver will be tuned to when receiving CW.

The corresponding figures for a second order prototype are shown in **Figures 3** and **4**.

Again, if you take into account the suppressed zero nature of the gain and amplitude curves, then you can see that there is little amplitude variation over the audio band. Both these circuits are examples of 'all pass networks'. This second order circuit has a resonance whose sharpness is determined by the ratio of values in the feedback to the + input of the op-amp. The resonance gives an opposite characteristic to the delay curve so it rises from the lower audio edge to the resonance just before 1.5kHz. That is because energy is stored in the tuned circuit for longer the nearer you get to the tuning point, so the signal takes longer to get through. The difference between the first and second order characteristics between 300Hz and 1.5kHz has the property we want, namely it moves with frequency as if the source moves in position in front of us when the frequency is changed.

These simulations confirm that the original design was correct in needing 3 sections of each filter to get the delay difference values that corresponds to a delay difference between sounds arriving at ears spaced by a typical head size, nominally about 15cm. A complete

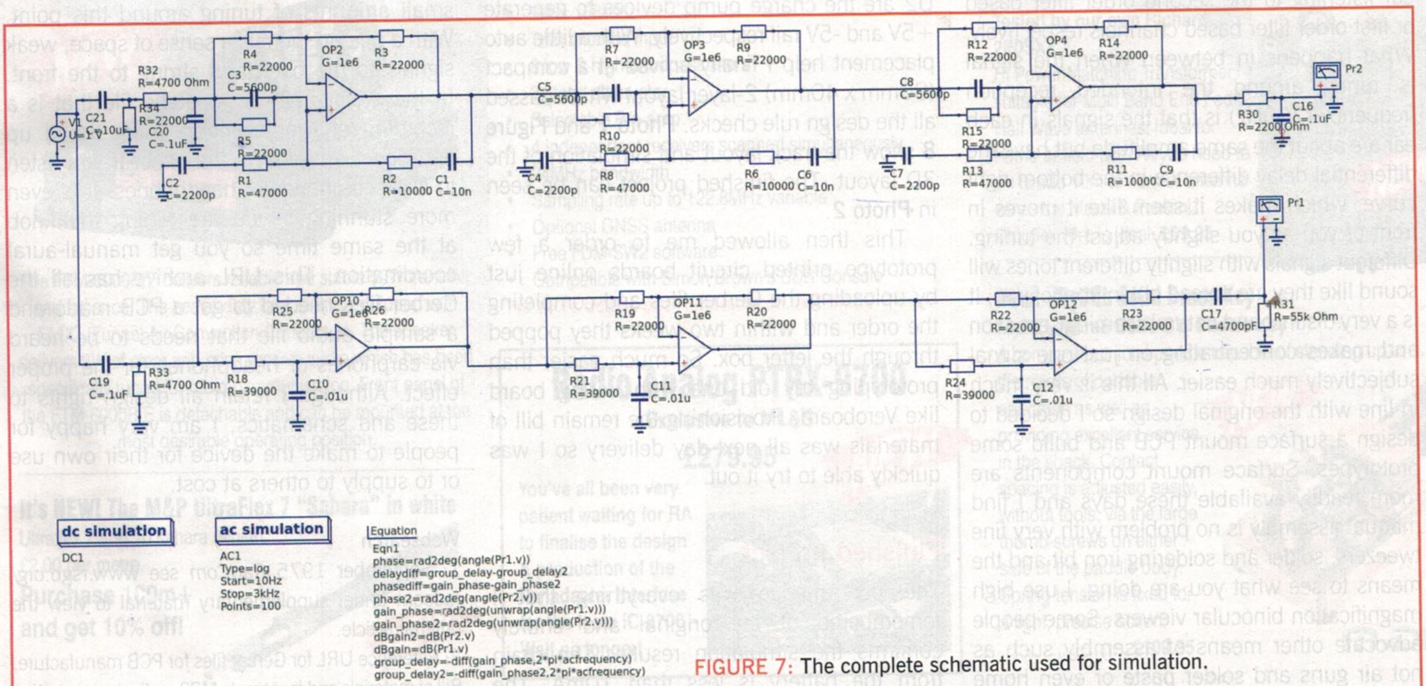


FIGURE 7: The complete schematic used for simulation.

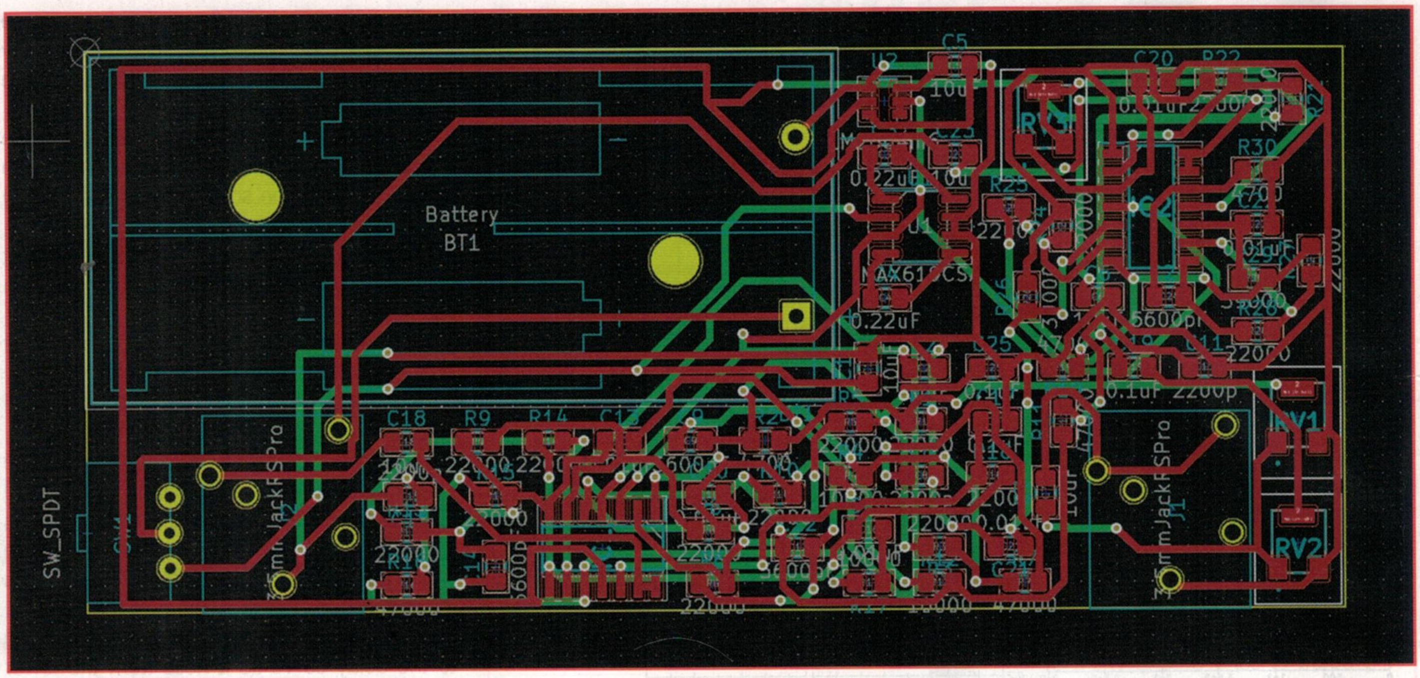


FIGURE 8: The track layout.

stereo-coder has three sections in each of the left and right channel halves to feed each ear and has an appropriate high or low pass amplitude shaping RC section at the input and output of each of left and right channels. The overall simulation schematic to get the required characteristics and simulated responses are as in **Figures 6** and **7**.

The responses show each channel and the differences in phase and delay in the bottom two curves. The rising and falling amplitude responses introduced by the RC filters are clear. Signals at the lowest or highest ends of the audio bandwidth will be mainly heard in the ear listening to the second order filter based or first order filter based channels respectively. What happens in between when the signal is tuned around the intended reception frequency (700Hz) is that the signals in each ear are about the same amplitude but have the differential delay difference in the bottom right curve, which makes it seem like it moves in front of you as you slightly adjust the tuning. Different signals with slightly different tones will sound like they are spread out in front of you. It is a very distinct and natural seeming sensation and makes concentrating on just one signal subjectively much easier. All this is very much in line with the original design so I decided to design a surface mount PCB and build some prototypes. Surface mount components are more readily available these days and I find manual assembly is no problem with very fine tweezers, solder and soldering iron bit and the means to see what you are doing. I use high magnification binocular viewers. Some people advocate other means of assembly such as hot air guns and solder paste or even home based reflow soldering, but I have found direct soldering effective and choose components that are suitable for this. Other methods of assembly will work, this is just the easiest for me. I use open source Kicad for schematic capture, net-list production and layout. The PCB schematic (**Figure 5**) has added an output buffer for earphones, connectors, power switch and a battery holder and charge pump devices to generate plus and minus 5V supply rails from the two AAA size batteries.

It uses two quad op-amps (IC1 and IC2) with a 14-pin dual in line surface mount footprint. The fourth amplifier in each is configured as a voltage follower to drive the earphones. U1 and U2 are the charge pump devices to generate +5V and -5V rail respectively. With a little auto placement help I finally arrived at a compact (90mm x 40mm) 2-layer layout which passed all the design rule checks. **Photo 1** and **Figure 8** show the track layout and simulation of the 3D layout. The finished project can be seen in **Photo 2**.

This then allowed me to order a few prototype printed circuit boards online just by uploading the Gerber files and completing the order and within two weeks they popped through the letter box. So much easier than prototyping my old one with wiring board like Veroboard. Provisioning the remain bill of materials was all next-day delivery so I was quickly able to try it out.

Finished project

The performance was everything that I remembered of the original and entirely confirms the simulation results. The drain from the battery is less than 10mA. The LM324 op amp has just enough ability to drive earphones without too much distortion and the whine from the charge pumps although detectable is not of concern. The two input potentiometers are set to near maximum and adjusted so each channel is equally loud on tune. In my case that is pretty much the same for each. RV3 should allow a small adjustment to the centre frequency. In my case I set it to about 39kΩ and left it there as it was near enough the tuning point for my rigs. You get used to tuning a signal into a 'position' that need not be exactly ahead to be 'netted', but it helps to be close. There is a tremendous sensation of signal motion with small amounts of tuning around this point. With a pile up there is a sense of space, weak signals to the back and strong to the front. In the archive **[2]** is an audio file that is a recording of live reception being tuned up and down and shows the effect if you listen to it on earphones or headphones. It is even more stunning if you are tuning the knob at the same time so you get manual-aural coordination. This URL archive has all the Gerber files needed to get a PCB made and a sample audio file that needs to be heard via earphones or headphones for the proper effect. Although I retain all design rights to these and schematics, I am very happy for people to make the device for their own use or to supply to others at cost.

Websearch

1: September 1975 *RadCom* see www.rsgb.org/radcom under supplementary material to view the original article.

2: Reference URL for Gerber files for PCB manufacture, Bill of materials and for sample MP3 audio demonstration https://github.com/john-g8cqx/Stereocode_2021.