EQUIPMENT REVIEW
Timewave DSP-9 and DSP-59 Audio Digital Signal Processors

Ron Fisher VK3OM and Ron Cook VK3 AFW

The DSP-9 and DSP-59 sitting on top of the Kenwood TS-430S transceiver used in the evaluation of the filters.

The latest device to fight interference in radio reception is the digital signal processor. Of the several being produced in the United States, Daycom (the new name of Stewart Electronics) have selected the Timewave DSP-9 and DSP-59 audio noise reduction filters. I must admit that I am not an expert in digital electronics, so I have asked my good friend Ron Cook VK3AFW to write a few words on just how these things work. Before handing over to Ron, I intend to report on the operation of these units connected to typical amateur equipment and used in a quieter than normal location.

The first illustration shows them sitting on my TS-430S transceiver which was used throughout the tests. Both filters are contained in neat black plastic cabinets with the controls on the front panels and the input/ output and power connectors at the rear (see the second illustration). The input to both units comes from the external speaker output of the associated transceiver and an inbuilt audio amplifier drives the speaker. Both units require 12 to 14 volts DC at about 500 ma. Of course a reasonable quality external speaker is also needed to complete the setup. For some strange reason, Timewave seem to be confused as to which type of connector to use. The DSP-9 uses phono sockets for both input and speaker output while the larger DSP-59 has 6.5 mm phone sockets for the same functions plus an extra 6.5 mm socket for line output. Seems odd that they didn't use a 3.5 mm socket for at least the speaker output. Both use a standard DC connector with the centre pin for positive. A DC connector is supplied with the processors.

So what do Timewave claim their processors will do? Firstly, they are designed to reduce all types of residual noise. Secondly, they will eliminate any number of heterodynes audible within the band pass, and finally they have very steep sided audio filters useable on both voice and CW signals. In the case of the DSP-9, voice filters are provided for 1.8, 2.4 and 3.1 kHz and for CW, 100, 200 and 500 Hz filters are selectable. The larger DSP-59 has basically the same features but with much greater flexibility and a wider range of filter selections. All functions on the DSP-9 are selected via six front panel push buttons plus a normal rotary audio gain control which also has an off position to cut the 12 volt power supply. The DSP-59 uses three rotary controls to select the various functions plus an audio gain/on/off control.

Two LEDs help to set the input audio level, one flashing with normal input and the second flashing to indicate an overload condition. A 3.5 mm socket is also included on the front panel to take a pair of headphones. This is compatible with stereo phones so you will be able to borrow a pair from your teenager's Walkman and plug in.

The Timewave DSP-9 & 59 in Use

As all of my speakers terminate in 3.5 mm plugs, I used adaptors to connect into the two Timewave processors. Another lead with a 3.5 mm plug and either a phono plug or 6.5 mm single circuit plug is needed to get the audio into the processors. The 12 volt DC supply needs to be well filtered and regulated. I tried a 500 mA plug pack power supply but it caused all sorts of funny hum problems.

A reasonably high audio output level from the transceiver is needed to get the "normal" LED to flash and it is very important that this should happen. When I first hooked the unit up, I thought I would take a short cut and feed the processor from the headphone socket on the TS-430. It proved impossible to drive the processor hard enough so a quick

The rear panels of the filters with the DSP-9 on top.

change was made to the speaker output.

First, I had a play with the filters. Changing from the 3.1 to the 2.4 kHz band pass produced a just perceptible change in audio quality but heterodynes and noise above 2.4 kHz disappeared like magic. Changing to 1.8 kHz brought an even greater reduction in off frequency noises with a slight reduction in top audio response.

The effect on CW with the narrow band pass is equally dramatic. The larger DSP-59 allows not only the choice of selectivity but also the choice of the centre frequency which makes it very suitable for digital modes.

Next on to the noise and heterodyne reduction features. As I mentioned earlier, I live in a very quiet location. There is only one thing that makes life on the air difficult. Rain static. Well, I had to wait to test that one out. Reduction of general noise, such as static, was interesting. With a moderately strong signal, the effect was often amazing. The signal would take on a slight synthesised sound, the noise would fade into the background and make the audio really stand out. However, as the signals became weaker the effect diminished with it. Enough to say that I could not find a case where I could make a totally unreadable signal readable. By far the best effect was to remove noise on moderately strong (about S6 to 7) signals. At long last the rain came (you don't have to wait too long in the Melbourne area) and with it S9 rain static. I regret to say that the processors made no improvement at all.

The heterodyne reduction facility is startling in its effectiveness. Just push the button on the DSP-9 or select the switch position on the DSP-59 and the whistles go. What more can be said? Ron VK3AFW tells me that he uses his DSP-9 for weak two metre CW use and claims that it really does make almost unreadable signals readable. I also noted that use of the transceiver noise blanker in conjunction with the DSP can add to the effectiveness of the processor.

The DSP-9 and DSP-59 Conclusions

Are these units worth while or not? The answer to this is a very definite yes. If you are a keen CW operator it would be hard to live without one. I don't doubt that in time, perhaps a short time, these units will come built into transceivers. Crystal IF filters will become a thing of the past and steep sided digital filters will become the norm. In the meantime, enjoy the advantages of digital signal processing with the DSP-9 and DSP-59.

I hope that when Timewave update these models they might sort out the connector problems and maybe even have time to write a better instruction book. The DSP-9 is priced at $339 and the DSP-59 at $629 from Daycom Electronics.

Now over to the other Ron to tell us just how these little electronic marvels work......

How Does a Digital Filter Work?

We are all familiar with analog filters, an example of which is shown in Fig 1. This is a single section RC low pass filter. DC and low frequency AC signals are not attenuated but as the frequency is raised, so the output falls. That is, the attenuation increases with frequency. The phase of the output signal also lags behind the input signal, the phase difference being greater at higher frequencies. Cascading several of these circuits gives both a sharper band edge roll-off and a greater phase shift.

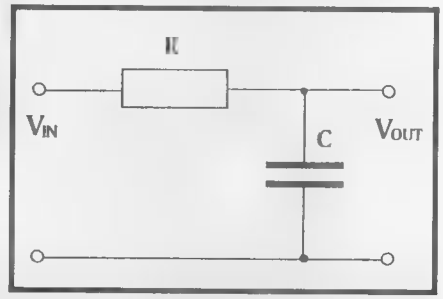


Figure 1 — Simple analog low-pass filter circuit.

We also know that if a DC voltage is suddenly applied to the input, the output rises slowly to the input value, the rate of rise being determined by the product of R and C. In other words the circuit in Fig 1 is also a delay circuit. We already know this as RC networks are used for the basis of many timing circuits. Further study would reveal that an analog delay circuit using discrete components usually has a bandwidth which is inversely proportional to the delay.

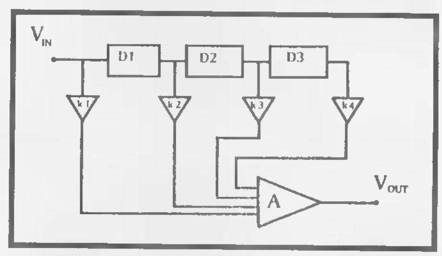


Figure 2 — Filter using a tapped delay line.
D1, D2, D3, incremental delays
K1, K2, K3, K4 multiplication constants
A summing amplifier.

Fig 2 shows a series of delays. This could be an analog system with taps. The input signal and the delayed signals from the taps are amplified or

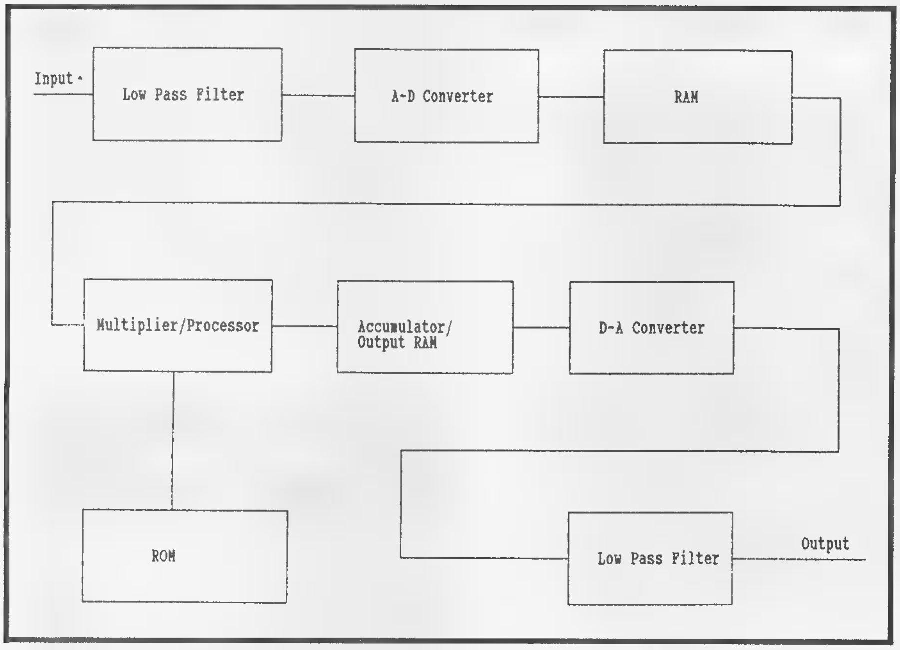


Figure 3 — Block diagram of a digital filter.

attenuated and added together to produce the output signal. With the proper components and values this will give exactly the same response as three RC cascaded filters. The circuit can be implemented using analog components or with digital circuitry.

Fig 3 shows a digital version of Fig 2. Firstly, the analog signal is passed through a low-pass filter to avoid a problem called aliasing. An analog to digital converter (A-D) then converts instantaneous values of the analog input signal to a digital number. The effect is like a picket fence where a solid fence is approximated by many thin pieces of the same height (see Graph 1). Sampling allows reproduction of frequencies up to half the sampling rate. It can down convert even higher frequencies to ones less than the sampling rate, a situation not wanted in this application. This creation of new frequencies is called aliasing and is avoided by filtering out all frequencies above half the sample rate.

Each digital sample is stored in digital memory (RAM). Each memory location represents a tap on our delay line. We select the stored value when we read the memory location. The delay is generated by reading memory locations at specified rate. A multiplier chip accepts the stored values and multiplies them by predetermined constants which are stored in digital memory (ROM). The results are added into an accumulator. The digital number in the accumulator is converted into an analog voltage by a digital to analog converter (D-A) and the digitising noise removed by a simple low-pass filter.

The resultant output signal will be filtered in exactly the same way as would have occurred in the analog circuit of Fig 2. A digital filter may have more than 25 taps in even a cheap system, resulting in very sharp roll-off at the band edges.

Having implemented a low-pass filter with only a few readily available chips, the question arises, can high-pass filters be constructed? The answer is yes, and what is more they can be combined to form a band-pass filter with linear phase response and very sharp roll-off. The circuit connections remain the same, however the constants used in combining the delayed samples can be changed for each calculation if required.

A system where the input signals only are used implements what is called a Finite Impulse Response filter (FIR filter) and a system that takes the processed signal and subjects it to delays and combines it with the input can be used to implement an Infinite Impulse

Ham Log is the gold standard log program — world-wide

Neil Duncan in ARA said: "Professionally-presented product... This is the way to do it properly... I have no hesitation in recommending the package...". **And that was Version 1!!!.**

On version two, **Len Shaw** wrote: "The author has gone considerably further (on features) than in any log program... ...you are unlikely to find a better log-keeping program anywhere. Having seen and used a wide variety of shareware and commercial programs, I believe this one to be excellent value for money... I would say the same if it was double the cost." Born in 1990, we now have version 2.4!

Some user comments: "Golly I am impressed... Over the Moon" "I have a number of logs and *Ham Log* is on the hard disk to stay!" "It is without doubt the best Log I have ever used — you have covered everything — a breeze to use" "A super program and Australian" "Thanks for the excellent back-up service" An FK8 wrote to the magazine ***Megahertz:*** "My job — programmer of 14 years... I've acquired 10 log programs without ever making use of them until I purchased *Ham Log.* It has maximum ease of use whilst keeping a rare functional richness... perfect statistics, listings and rare tools — torture it: one use will make you appreciate its flexibility." VK6PY, VK4VHP, VK5QB, FK8GV.

Put your IBM (640k) computer to use. ***Ham Log*** overflows with useful features (space does not permit to describe — ask for brochure). Full DXCC info. and Statistics; 8 Modes; Contest Mode; Prints QSLs; Country's Time, Beam Headings Distance, Language translations, plus **stacks** more! Fast, Menu driven, Help + 80 page published manual. Enjoy the fascinating benefits *Ham Log* provides.

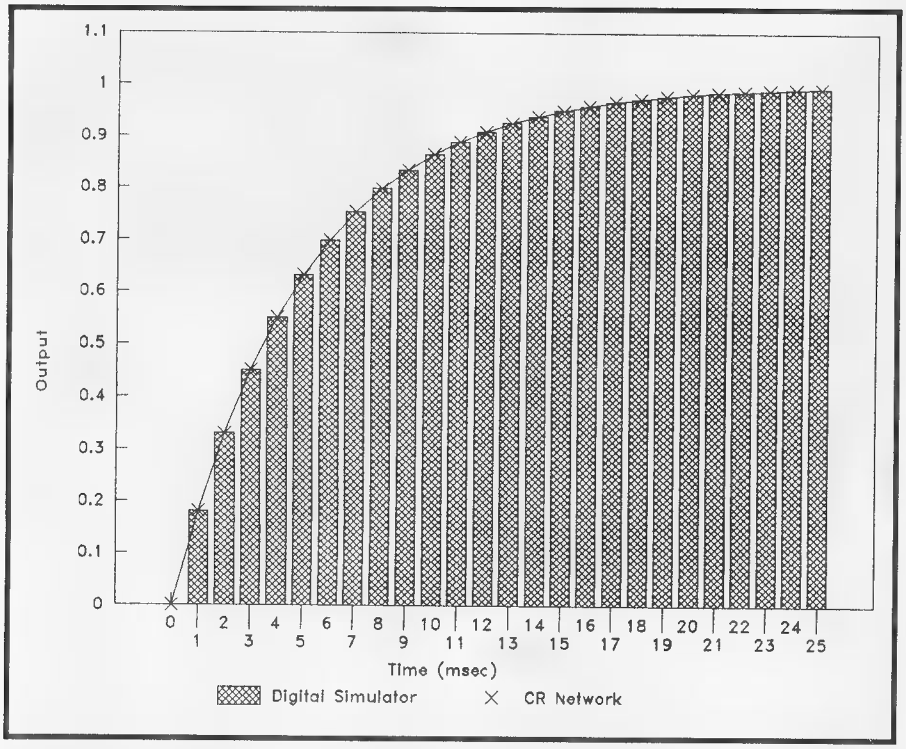
ONLY: **$57.50** Demo: Free

"Limited to first 100 orders; 720K"

incl. P&P to VK

BANKCARD VISA MASTERCARD ORDERS PLEASE SUPPLY... FULL NAME, ADDRESS, PHONE, CALL SIGN, DISK SIZE.

Robin Gandevia, VK2VN
Dr Hi Fi P/L 74 Carrington Rd.
WAVERLEY 2024 PH: (02) 369 2008
FAX: (02) 369 3069



Graph 1 — Simulation of CR network.
The output of a CR network as in Fig 1 is compared to a digital simulation of the same network. The time constant is 5 milliseconds and the digital algorithm is: Vout = 0.18127 Vin + 0.81873 Vout.

Response (IIR) system. The DSP-9 used the FIR implementation partly because it provides a linear phase response which gives the minimum amount of ringing for narrow filters.

There are of course some compromises. The multiplier must read each memory location, do the multiplications and transfer results to the accumulator before the A-D takes its next sample and the D-A puts out the next signal. The speed required limits inexpensive digital filters to the audio range but military receivers are using digital filtering at an IF of 1.6 MHz.

The DSP-9 uses a special chip which performs both A-D and D-A conversion and includes all necessary filtering. It uses a 16 bit process. The processor contains sufficient RAM to accommodate all the samples. The program and all constants are stored in a 256 k EPROM. The results of each computation are placed in a quad flip-flop for the D-A to convert back to audio. A 5 watt amplifier drives an external speaker.

Digital filters in the amateur radio market are mostly using 16 bit word lengths and combined multiplier accumulator chips. A 16 bit word allows a dynamic range of 96 dB but the band reject attenuation is around 60 dB for most digital filters on the amateur market. This is usually very adequate.

One problem that appears is the small number of bits available to represent weak signals. Consequently they sound like a strangled Dalek. Faster systems with higher resolution D-As and 32 or 64 bit processors will no doubt appear eventually and give improved weak signal recovery, increasing the fidelity and out of band attenuation.

Having converted to digital form many samples of the signal, plus noise and interference, there is an opportunity to perform other functions as well as filtering. Notching of constant tones can be achieved by various means, depending on the computing power available. Reduction of random noise can also be achieved by comparing delayed samples with the present sample. Audio signals show a high degree of correlation but noise does not, so the noise can be rejected. This provides a basis for implementing a noise reduction scheme in addition to that obtained by bandwidth reduction.

ar

Support the advertisers who support Amateur Radio magazine.

WIA News

"Instant licences" for US Hams?

A US amateur radio society has petitioned the Federal Communications Commission (FCC) seeking a rule change that would permit "instant" amateur radio licensing.

The Western Carolina Amateur Radio Society (WCARS), based in Knoxville, has asked the FCC to allow amateur operating privileges to start immediately someone passes the required exam, without having to wait for a licence to be issued.

When licence exam candidates in the US pass their exam, they're issued with a Certificate of Successful Completion, with which they can apply for their first licence. WCARS' instant licensing scheme would save the frustrating waiting period for new hams, the protagonists claim, as well as saving the FCC time and money as the impatient new hams keep calling them for news of their licence.

WCARS proposed a callsign structure based on the US Class D Citizen's Radio Service (CRS) precedent, set a few years ago when the FCC deregulated the CRS.

From the *Westlink* Report.