A TERMINAL UNIT USING SWITCHED CAPACITOR FILTERS

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WITH SWITCHED CAPACITOR FILTERS now available to the radio amateur at reasonable prices. it is possible to design audio systems with great versatility which previously would have been much more complex had they been implemented with conventional RC op-amp components.

This article describes the use of some of these new generation filters in a terminal unit for use in data communications. The design is readily modifiable for practically any data rate and standard, however the design was primarily intended for hf Amtor/rtty applications

Over the past year the design has shown itself able to cope with large amounts of interference, and still detect the incoming traffic with relatively few errors. On the vhf and uhf bands its performance has been excellent, often detecting data which is practically inaudible The performance of the internal filtering make it an ideal addition to any data communication set up which lacks pre-detection filters.

Switched capacitor filter operation

To enable a better understanding of the circuit operation for amateurs unfamiliar with switched capacitor filters (scfs), here is a brief introduction to their operation.

Usually, in the design of filters; resistors, capacitors and inductors are used to form the filter network. However, for high Q (narrow band) filters a high quality inductor is usually required; that is an inductor with a high intrinsic Q This type of inductor is normally quite large and/or expensive.

A common method of eliminating the inductor in a lower frequency filter is to use an active element, eg an op-amp, in conjunction with resistors and capacitors.

Unfortunately, using an active element combined with passive components can lead to instability and drift in high Q filters, and therefore tends to need very careful design.

SCFs use a modulation technique, from an external frequency source, to overcome the above problems. There are several configurations of scf each with its own particular properties, but in this article we shall consider just the shunt switched bandpass filter as it is among the easiest to understand.

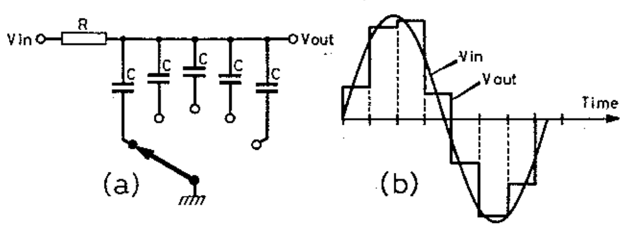

Fig 1 (a) Circuit diagram of the shunt-switched bandpass filter. (b) Resulting waveform

The basic circuit arrangement is shown in Fig 1(a) and the resulting waveform in Fig 1(b) Providing the rc time constants are much greater than the rate at which the switch is operated then each capacitor is exposed to a segment of the applied waveform and so eventually reaches the average value of the applied signal The voltages on the capacitors only remain absolutely constant if the applied waveform has a frequency of Fclk/n, where Fclk is the frequency of rotation of the switch and n is the number of capacitors

In real life, however, the applied waveform is not in exact synchronism with the clock frequency and so the average voltage on each individual capacitor during each successive time interval varies. Because of this lack of synchronism the voltage across each capacitor varies at a rate dependent on the difference between the applied waveform frequency and Fclk/n.

For large differences, the capacitors do not accumulate appreciable charge and the output voltage remains close to zero

When the input frequency and switching frequencies are in synchronism, each capacitor will, after a few cycles, charge to the average sample value of the input signal. As this happens to each capacitor, the output becomes a stepped approximation of the input. Thus, the bandpass function is formed. A simple lowpass filter is then required to remove the sampling frequency components.

Due to the sampling nature of this type of filter, it is possible for responses to occur at the sampling frequency and at the harmonics of the filter frequency. If required to prevent this 'aliasing' occurring an anti-aliasing filter is normally placed ahead of the scf.

In order to realise other types of filter (notch, highpass, lowpass etc), this basic filter can be used with op-amps in various circuit configurations, which are outside the scope of this article

However. for further reading on this matter I recommend the *Siliconix Handbook on Analogue Switches* This handbook is available from Siliconix Ltd Morriston, Swansea

Components List

R1, R2	1kΩ
R3, 16	56kΩ
R4, 5, 6, 7, 8, 9, 13, 14, 15, 17, 19, 21, 30 33, 37, 38, 63, 64, 65	10kΩ
R10	150kΩ
R11, 26, 27 28, 53	3·3kΩ
R12, 59	2 2kΩ
R18, 57	2 7kΩ
R20	18k
R22	50kΩ pre-set
R23 24	47kΩ
R25	220kΩ
R29	10kΩ pre-set
R31, 32, 34	680Ω
R35, 42, 45	27kΩ
R36, 43, 46, 49	5·6kΩ
R39, 40	33kΩ
R47, 44, 62	470Ω
R47	5kΩ pre-set
R48	8·2kΩ
R50, 52, 54, 56, 58. 60	1kΩ pre-set
R51	3kΩ
R55	3 9kΩ
R61	1 8kΩ
All resistors 5% 0.25W	
C1, 3, 5, 6, 7, 8. 11, 14, 21, 22, 23. 24	2·2μf tantalum
C2	220pf ceramic
C10	470pf ceramic

C12	22μf tantalum
C9, 15	22μf electrolytic
C13	220μf electrolytic
C16	10μf electrolytic
C19 20	47μf electrolytic
C27	4 700μf 25V electrolytic
C17, 18	0·01μf paper
C25	0·22μf paper
C26	0 22μf 25V paper
C28. 29, 30	1nf suflex
All capacitors 16V wkg unless otherwise stated	
IC1	RS 306-803
IC2, IC3	RU 5620
IC4, IC5, IC8, IC9, IC10, IC11	741
IC6	4538
IC7	R 5609
IC12	78L08
IC13	7812
IC14 IC15, IC16	4047
TR1, TR2, TR3, TR4	BC108
D1, D2, D3, D4, D5, D6, D7, D9, D16	1N914
D9, D10, D15	LED
D11, D12. D13. D14	1N4001
T1	15V 10VA
M1	± 100μA
S1 S4	SPCO
S2	2P3W
S3	1P3W

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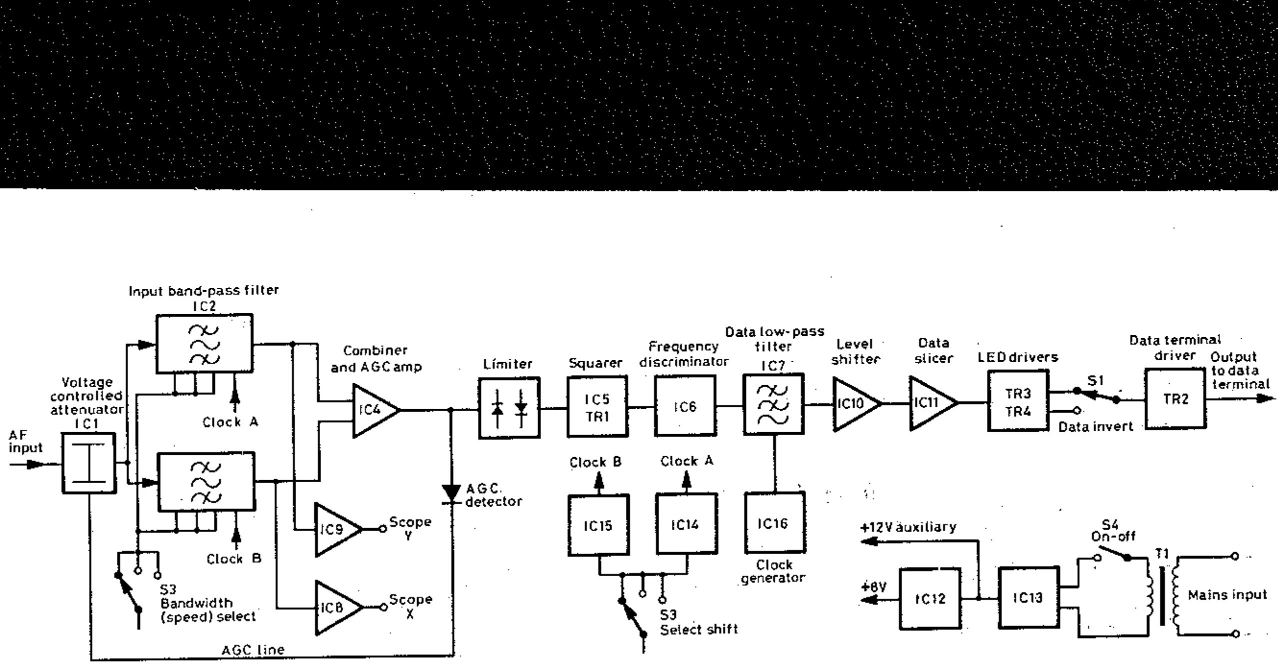


Fig 2 System block diagram

Circuit description

This circuit description should be read in conjunction with the block diagram Fig 2 and circuit diagram Fig 4.

The incoming audio from the receiver is first fed to IC1 This serves to control the level of the audio to the following filters, by means of the agc voltage generated by D1 and D2, which is applied to pin 2 This levelling of the audio signal serves to combat fading and as it is achieved by using "linear" techniques, as opposed to the conventional "back-to-back diodes"; it does not introduce any great deal of distortion into the signal which could have a detrimental effect on the data.

After the audio has been suitably levelled it is passed onto IC2 and IC3, which form the mark and space bandpass filters. These ICs are of the programmable switched capacitor type, which make it easy to set the required centre frequency and bandwidth for any particular system requirement.

For instance, the centre frequency is determined by the clock frequency applied to pin 7, divided by n. Where n is a number selected from Table 1. Likewise, the bandwidth or Q is determined by the code applied to pins 2,3,4 5 and 6. The required Q can be selected from Table 1.

Table 1

Q	Code Q4 . . Q0	Fc/Fo	Code F4 . . F0	Q	Code Q4 . . Q0	Fc/Fo	Code F4 . . F0
0.57	00000	200.0	00000	5.00	10000	97.8	10000
0.65	00001	191.3	00001	5.80	10001	93.5	10001
0.71	00010	182.9	00010	7.20	10010	89.4	10010
0.79	00011	174.9	00011	8.70	10011	85.5	10011
0.87	00100	167.2	00100	10.0	10100	81.8	10100
0.95	00101	159.9	00101	11.5	10101	78.2	10101
1.05	00110	152.9	00110	12.0	10110	74.8	10110
1.20	00111	146.2	00111	13.5	10111	71.5	10111
1.35	01000	139.8	01000	15.5	11000	68.4	11000
1.65	01001	133.7	01001	17.5	11001	65.4	11001
1.95	01010	127.9	01010	20.0	11010	62.5	11010
2.20	01011	122.3	01011	24.0	11011	59.8	11011
2.50	01100	116.9	01100	30.0	11100	57.2	11100
3.00	01101	111.8	01101	35.0	11101	54.8	11101
3.50	01110	106.9	01110	55.0	11110	52.3	11110
4.25	01111	102.3	01111	85.0	11111	50.0	11111

In this application IC2 and IC3 are configured as bandpass filters; however it is equally easy to configure them into any of the following:

a) Lowpass
b) Bandpass (as described in this article)
c) Highpass
d) Lowpass elliptical
e) Highpass elliptical
f) Notch
g) Allpass

To use these filters in any of the above modes, refer to Table 2.

Fig 3 details the pinouts for IC2/3; should further data be required on these ICs, it can be obtained from: EG&G Reticon, 34/35 Market Place, Wokingham Berks RG11 2PP, telephone 0734 788666.

Table 2

Filter type	LPin	HPin	BPin
	Connections		
Lowpass	Vin	GND	GND
Highpass	GND	Vin	GND
Bandpass	GND	GND	Vin
Notch	Vin	Vin	GND
Allpass	Vin	Vin	Vin

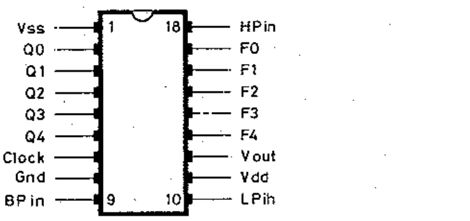


Fig 3. RU5620 pinouts. Note that this ic is designed for + and − supplies. However, by biasing inputs to mid-rail it is possible for the device to be powered from a single supply rail. Q0 to Q4 program the "Q", see Table 1. F0 to F4 program the Fc/Fo ratio, see Table 1. BPin = bandpass input LPin = lowpass input HPin = highpass input

In this design the centre frequencies are set by the clock frequencies generated by IC14 for IC2, IC15 for IC3 and IC16 for IC7. For IC2 and IC3 the centre frequency is the clock frequency divided by 50.

The bandwidth or Q of the filters is set by S3, which programs pins 3,4,5 of IC2 and IC3. Pins 2 and 6 are connected to the supply rails and are not programmable in this design.

After the audio tones have passed through IC2 and IC3 they are substantially free of noise and interference, at this point they are amplified in IC8 and IC9 to drive an oscilloscope in XY format to aid the accurate tuning of the signal This feature is extremely useful to get the best out of the system and also makes it very easy to 'net' accurately onto a station calling CQ.

After the tones have been filtered on IC2, IC3 they are combined and amplified by IC4. The output of this amplifier is rectified in D1 and D2 to provide the agc voltage for IC1. IC4 also removes any residual clock leak-through and reconstitutes the waveform

The output of IC4 is symmetrically limited in D3 and D4 before being amplified in IC5 and squared up in TR1

The collector of TR1 is used to trigger IC6, which is a precision monostable set to 0.7ms. The output of IC6 is used to charge up a capacitor (C18), and the resulting voltage across this capacitor is a measure of the predominate frequency.

As there may be some residual noise present on this voltage, due to interference etc, a lowpass filter is inserted by means of IC7. This IC again is of the switched capacitor type, and has a very fast roll-off typically, >60dB per octave.

The cut-off frequency of IC7 is set by the clock frequency generated by IC16 divided by 100. Typically, the cut-off frequency is set to 1.5 times the bit rate A cut-off frequency lower than this will cause the data to become blurred and so make the operation of the data slicer inaccurate

The filtered data from IC7 is fed to IC10, where the transition frequency is accurately set by R29, and the data is sliced in IC11 M1 provides an indication of tuning, should an oscilloscope not be available.

TR2 provides an open collector output to the terminal, while TR3 and TR4 light the appropriate l e ds to indicate the presence of either tone accordingly.

S1 provides a convenient means of inverting the sense of the data. S2 selects the clock frequencies for IC2 and IC3, to set the appropriate shift (170Hz, 425Hz, 850Hz). S3 sets the bandwidth of the mark space filters so as to be optimum for 50, 75 or 110 baud rates.

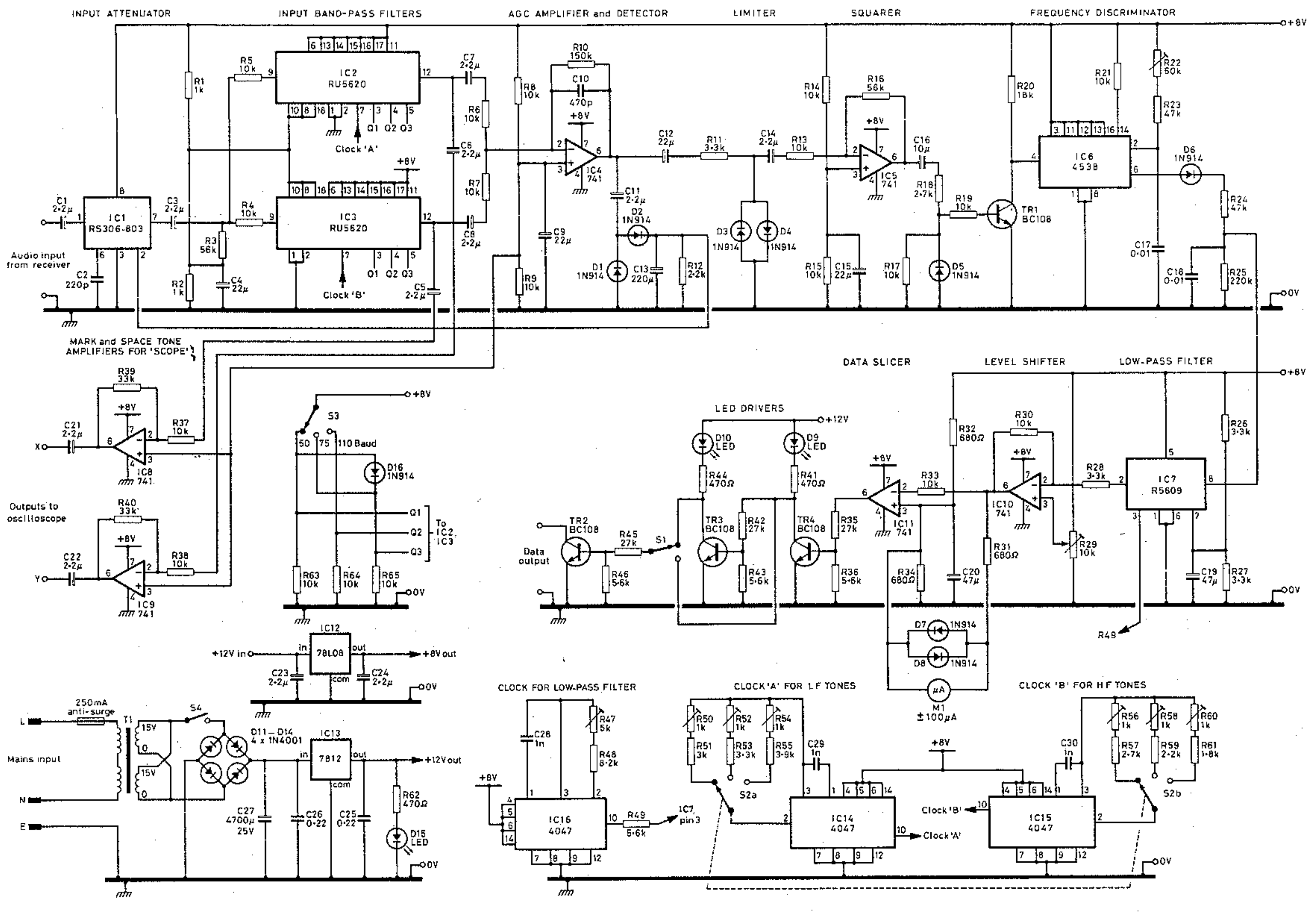
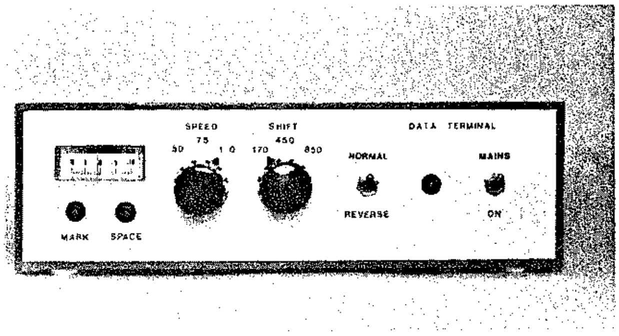
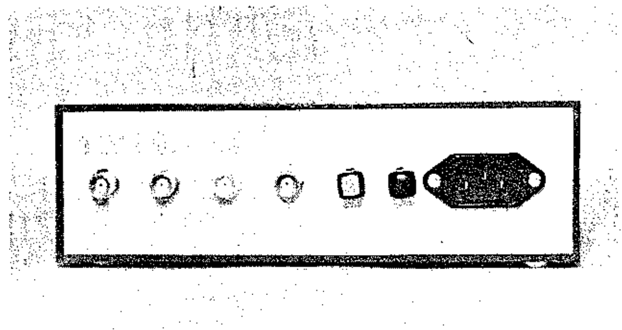


Fig 4. Circuit diagram of the terminal unit



Front view of the data terminal



Rear view of the data terminal

IC14, IC15 and IC16 are conventional cmos oscillators with an integral divide by two stage, to provide the filters with a 1:1 duty cycle clock. However, a 1:1 duty cycle clock is not essential for the correct operation of the filters, and any other clock oscillator (such as NE555 etc) could be substituted

The power supply is of conventional design using the popular 78 series of regulators. IC13 being the main 12V regulator, which in my case is used to power an external code converter, while IC12 provides a regulated 8V supply for the terminal circuitry. T1 is a miniature 10VA toroidal type which gives good safety isolation from the mains

In the unlikely event of a catastrophic failure of the psu a 1A fuse is located in the mains plug, however with psus of this size it is very difficult to provide adequate protection of the transformer due to inherent current limiting in the mains transformer

Construction

The construction of the prototype was carried out on Veroboard, to aid development of the circuit. As yet, no pcb has been made If demand is great enough then a pcb will be laid out

Due to the cost of some of the ics it is preferable to use ic bases and to check the power supply voltages prior to inserting the ics into the completed circuit.

No special techniques are required in the construction of this unit with the possible exception of proofing the circuitry against rf on transmit, by means of suitable chokes and capacitors for the frequencies in use.

IC13 should be fastened down to a suitable heatsink and the decoupling capacitors C26 and C27 mounted adjacent to the device to prevent instability.

As this unit consumes less than 10W, it is permissible only to switch the secondary of the mains transformer so reducing the amount of mains wiring and the risk of electric shock

Although separate op-amps devices are show for individual stages, there is no reason why multiple op-amp packages such as LM324 should not be used with the appropriate changes to the pin-outs.

The prototype was built into a Schroff case, which was a convenient size for this type of project. The case is not particularly rf proof, but as yet no problems have been experienced while operating at full legal maximum power on any band from 1 8MHz to 432MHz. Therefore, it seems that the circuitry is fairly immune, providing sensible precautions are taken to prevent excessive amounts of rf energy entering the case

Setting up

To accurately set this unit up, it is preferable to have either a frequency meter or a source of the tones to be used and an oscilloscope. Ideally, all the above would aid setting up and any fault-finding necessary. It is possible to adjust this unit without any test gear at all providing the constructor has the time and patience

Setting up the filters

To set the centre frequencies of the filters it is necessary to set the clock frequencies as follows:

For 170Hz shift, set R50 for 63,750Hz, this sets a filter centre frequency of 1 275Hz then set R56 for 72 250Hz, this sets a filter centre frequency of 1 445Hz.

For 425Hz shift, set R52 for 57,375Hz this sets a filter centre frequency of 1 147 5Hz, then set R58 for 78,625Hz this sets a filter centre frequency of 1 572 5Hz

For 850Hz shift set R54 for 46 750Hz, this sets a filter centre frequency of 935Hz, then set R60 for 89,250Hz, this sets a filter centre frequency of 1 785Hz

The above clock frequencies are all measured at pin 7 of the filter ic under adjustment

This procedure sets the various mark and space frequencies for 170, 425 and 850Hz symmetrically around a transition frequency of 1,360Hz. The reason for this is that the design is primarily intended for amateur 170Hz shifts, and as such the frequency discriminator formed by IC6 is optimized for frequencies around 1,360Hz Therefore, to ensure optimum performance at other commercial shifts the filter frequencies are centred around 1 360Hz

Setting up the discriminator

Initially, 170Hz shift should be selected at a speed of 110 baud, then with an audio source of 1,275Hz at approximately 200mV rms connected to the input, R22 should be adjusted for approx 2V at the junction of C18 and R24 This voltage should be measured with a high impedance voltmeter. If no suitable instrument is available then R22 should initially be set mid way

The clock frequency for IC7 should be set for 20,000Hz, this sets a low pass frequency of 200Hz and is adequate for data rates up to 110 baud.

With an audio source of 1,360Hz at approximately 200mV rms connected to the input R29 should be adjusted for a centre zero indication of M1, both mark and space l e ds should be lit or rapidly flashing alternately

Reset the audio source to 1,275Hz and note that the meter moves to half scale to the left and that only the space l e.d. lights. Then set the audio source to 1,445Hz and note that the meter moves to half scale to the right and that only the mark l e.d. lights

Repeat the above procedure for input frequencies of 425Hz and 850Hz shifts, as previously stated

Check also that the oscilloscope outputs are operating and that the input level can be varied from 400mV rms to 25mV rms without the level at pin 6 of IC4 varying by more than 20 per cent. This check ensures that the internal agc is operating.

Operation

With the unit connected up to a suitable receiver, either to the headphones or loudspeaker taking care not to short out the output stage in the receiver, and with the data output of the unit connected to a suitable terminal for the data type in use, tune the receiver into a transmission which matches the type selected on the unit (shift and speed) and centre the tuning meter If random characters (rubbish) appear on the screen try inverting the data to the terminal by operating S1.

If an oscilloscope is being used as an additional tuning aid then tune the receiver for a "+" shape This indicates the correct tuning.

Other applications

While this article describes the design of an rtty/Amtor terminal, there are many more applications for these types of filters, eg auto tracking notch filters adjustable cw filters, sstv filters etc. Practically anywhere an audio filter is required a digital filter is possible and usually is easier to implement

Conclusion

This design is a prototype and as such is still under development, therefore if anyone has any suggestions or questions regarding this design I would be pleased to hear from them. □