PROGRAM CONDITIONER

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THE usual way to copy a personal computer program stored on cassette is to load the program into the computer and record it onto another cassette. This method has both advantages and disadvantages. On the plus side, only one cassette recorder is required, and the program listing can be checked for errors on the visual display unit screen—usually a TV set—as it loads. Further, when it is recorded, the new recording is retimed by the computer so that any timing imperfections on the original due to flutter or other cassette deck limitations are not compounded.

On the debit side, the computer is tied up whilst loading and recording are in progress. Also, it is sometimes found that the program won't load correctly in the first place, due to inadequacies in the original recording or in the cassette machine used for playback, or due to the combination of the two—there are fairly startling differences in performance between different models.

The circuit forming the subject of this article can often circumvent this problem and is thus useful as a signal conditioner to rescue poor recordings and enable them to be successfully loaded into a computer. Also, if two recorders are available, it permits programs to be copied directly without tying up a computer. The latter method is particularly useful for the enthusiast who is still saving up for a computer system but who wants to be able to collect programs from friends in the meantime.

The circuit consists of four sections—a selectable high-pass filter, an adjustable all-pass filter, an output stage providing clipping and shaping, and an audio stage driving a monitor loudspeaker. To understand the purpose of the various stages—particularly the all-pass filter (also known as a phase equaliser)—one needs to know a little of how tape recording works.

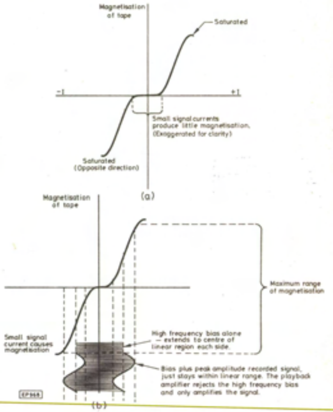

Fig. 1. Magnetic tape recording with (a) no bias and (b) the effect of bias

RECORDING ON TAPE

Magnetic recording tape consists of thin plastic, coated with finely divided iron oxide or other suitable magnetic powder. You may have noticed that once pins have been picked up with a magnet, they will tend to stick together even in the absence of a magnet. This effect is called remanence and is the basis of tape recording. Fig. 1 shows how a recording head records the signal onto the tape.

The relation between the record current and the remanent magnetism is nonlinear, so a high frequency (50 to 100kHz) bias current is added to spread the recording to the linear parts of the characteristic. On cassette recorders the same head is usually used for playback, so that monitoring from the tape is not possible. Another head uses the a.c. bias current at a much higher power to erase previous recordings. Quarter track heads are used in stereo cassette recorders, providing two tracks in either direction. Mono recorders use half track heads in the same way as older half track reel to reel recorders. With stereo cassette machines, the two tracks of the stereo signal are on the same side of the tape, and both are erased at the same time by the erase head. As with reel to reel recorders, the erase head is energised during recording and the tape passes over it first, before reaching the record head. Thus four track mono use is not available, but mono cassette recordings can be played on

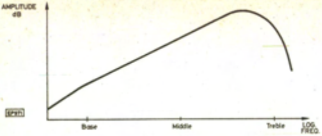

Fig. 2. Typical tape playback curve for constant peak recorded flux density

stereo cassette decks and vice versa. Of course the result is always mono except in the case of a cassette recorded on a stereo machine being replayed on a stereo machine.

There is nothing to prevent one using an older reel to reel recorder for storing personal computer programs on tape, though cassette machines are almost invariably used because of their greater convenience.

Also, the greater demands made on the tape by the very low tape speed ($1\frac{7}{8}$in per second) and narrow track of the heads used in cassette machines has resulted in the development of much better tape, suffering less from drop outs (random areas of low sensitivity along the tape) than older reel to reel tapes. However, whatever type of tape is used, one still has to cope with the basic frequency response of magnetic recording tape, as shown in Fig. 2. The 6dB per octave roll-off in the middle and lower frequency region is a result of the law of induction, the induced voltage on playback (for constant peak recorded flux level) is directly proportional to the rate of change of flux density, and hence to the frequency.

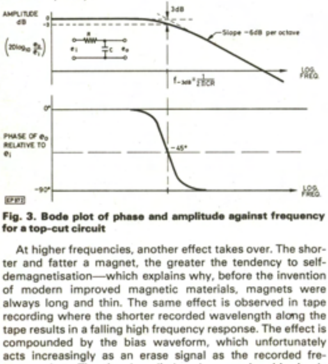

Fig. 3. Bode plot of phase and amplitude against frequency for a top-cut circuit

At higher frequencies, another effect takes over. The shorter and fatter a magnet, the greater the tendency to self-demagnetisation—which explains why, before the invention of modern improved magnetic materials, magnets were always long and thin. The same effect is observed in tape recording where the shorter recorded wavelength along the tape results in a falling high frequency response. The effect is compounded by the bias waveform, which unfortunately acts increasingly as an erase signal as the recorded frequency rises. The signal applied to the erase head is simply the bias waveform, but at a very much higher level.

To obtain a level overall response independent of frequency, on playback the middle and bass are boosted by 6dB per octave and the high frequencies, above the frequency of maximum response are also boosted. The rate of fall-off of the high frequency response is in fact faster than 6dB per octave, so boost is also applied during recording.

EFFECT OF PHASE

It is usually claimed that the phase response of audio equipment is unimportant as the ear is not sensitive to phase, and this is certainly largely true as otherwise tape recordings would all sound awful. The fact is that audio tape recorders of all types do dreadful things to the relative phase of the harmonics of, for example, a squarewave. To see why we must now digress a moment and look at phase and amplitude responses versus frequency for various circuits and processes, usually called bode plots.

Fig. 3 shows a top cut or high frequency roll-off circuit, together with its Bode plots. It can be seen that at low frequencies, where the response is level, there is little if any phase shift; whilst at high frequencies, where the response is falling off at 6dB per octave, the phase shift approaches 90° lagging. At the point where the 6dB/octave asymptote crosses the level asymptote, the amplitude response is −3dB, and the phase shift is −45° (45° lagging). Bode's relations show that generally a lagging or negative phase shift is associated with an amplitude response which is falling with increase of frequency, and a leading or positive phase shift with a rising amplitude response. The change in amplitude of response in dBs (decibels) between two frequencies f_1 and f_2 is proportional to the integral of the phase shift between limits f_1 and f_2, and the phase shift at any frequency is proportional to the rate of change of amplitude response. The going rate of exchange is 90° (lead or lag) for a 6dB/octave rate of change of amplitude (rising or falling with frequency respectively).

Does this apply universally? Well, almost; that is it applies to all "minimum phase" circuits and processes and this includes most tone control networks and filters. There are, however, "non minimum phase" networks. This can occur for example where there are two parallel paths through a network. In such cases Bode's relationships do not apply—a well known example is the twin tee network. There are a number of non minimum phase processes and two other examples are all-pass filters and tape recorders.

In tape recording, it has been explained that the high frequency roll off is due to self- and bias-demagnetisation, and in fact there is no associated phase shift involved—a non minimum phase process. The treble boost applied on both record and playback to maintain a level response is a minimum phase process and thus, overall, the amplitude response is level, but the phase advanced.

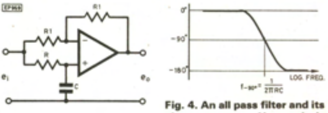

Fig. 4. An all pass filter and its phase response. Here gain is unity at all frequencies

The all pass filter in Fig. 4 has an amplitude response independent of frequency, but provides a phase response which varies with frequency. At very low frequencies, it is a non inverting amplifier with unity gain, i.e. 0° phase shift. At a frequency $f_{90}=1/(2\pi CR)$ it has a 90° phase lag and at high frequencies the phase shift becomes 180° lagging, all the while with a flat amplitude response. Fig. 5 shows the effect on a square wave of changing the product CR. When $1/(2\pi CR)$ is very much lower than the fundamental frequency of the square wave, f_f, all the frequency components of the square wave are shifted by −180°, so the waveform is unaffected. If f_{90} rises—say by reducing the value of R—the effect on the squarewave at first is not unlike the slope or tilt produced by an inadequate coupling capacitor—Fig. 5(a). As f_{90} approaches f_f, the effect is more marked, but as the fundamental is now shifted in phase relative to the harmonics without its amplitude being reduced, we see the waveform of Fig. 5(b), and as f_{90} becomes higher and then much higher than f_f we see waveforms like Figs. 5(c) and (d).

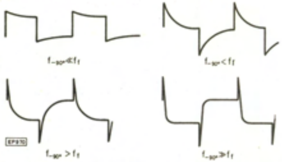

Fig. 5. Effect of the all pass filter on an ideal square wave of frequency f_f

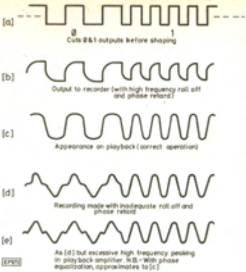

Fig. 6. Waveforms associated with cassette program recording

RECORDING PROGRAMS

Returning now to the topic of recording programs on cassettes, the usual format is CUTS (computer users' tape system), also known as the Kansas City interface. This is an FSK (frequency shift keying) system where a frequency of 1200Hz represents a zero and 2400Hz a 1. As the signalling rate is 300 bauds (300 signal elements per second) one signal element consists of either 4 cycles of 1200Hz tone (a zero) or 8 cycles of 2400Hz tone (a 1). The format is a 1 as a start bit, eight data bits (1s or 0s according to the byte to be transmitted) and two stop bits. On replay, the computer looks for a 0 (start bit) following a long string of 1s, and thereafter for succeeding start bits following the two stop bits. Signal conditioning circuitry in the computer's CUTS input turns the two tones into 0 or 1 levels as appropriate.

Thus the system is asynchronous, and provided the difference in speed between the recorder used to make the recording and that used to reproduce it (allowing also for the effects of flutter and noise) is not more than a few percent then no problems should arise. Nevertheless, the phase distortion already noted is bound to occur in the cassette recording/playback process, so compensation for the phase advance of the harmonics is applied by a phase retard (top cut) circuit in the computer's CUTS output—Fig. 6(b). Allowing for the limited frequency response of most cassette recorders, this would ideally appear on playback as in Fig. 6(c).

Several deviations from the ideal playback waveform are commonly observed. The waveform often has superimposed hum at 50Hz (poor layout, induced hum from the recorder's mains transformer), at 100Hz (inadequate smoothing of full wave rectifier output), and at 150Hz (a third harmonic component of the mains transformer's magnetising current due to using the core laminations at a peak flux density approaching saturation). The smallish internal speaker used in many machines is relatively insensitive at 150Hz and completely dead at 50Hz, so the manufacturer has little incentive to worry over much about these hum components in the playback signal. To cope with this problem, the circuit described later has an active 2 pole high pass filter which can be switched into circuit if required.

Another problem one encounters is excessive phase distortion, due to a variety of causes. One commercially produced program appeared to have been recorded without sufficient phase retard, or maybe excessive peaking in the record circuitry. The result was that playback on the author's Trophy CR100 cassette recorder resulted in the 1200Hz tone appearing as in Fig. 6(d). The program loaded o.k., but on a colleague's cassette deck (which was evidently equipped with extra enthusiastic treble peaking in the playback equalisation section) the waveform appeared as in Fig. 6(e). The additional zero crossings got sliced in the computer's CUTS input circuit, turning occasional 0s into 1s and corrupting the program. The extraneous wiggles could be reduced with the aid of the tone control (which, like the volume control, controlled the output at the recorder's 5 pin DIN auxiliary socket) but that also had the undesirable effect of attenuating the 2400Hz 1 tone. The all pass filter in the signal conditioner described in this article permits the harmonics to be properly relocated in phase relative to the fundamental, resulting in the 1200Hz tone again resembling a squarewave, without attenuating the 2400Hz tone at all. So, having prepared the ground, let's look at the full circuit of the Program Conditioner, Fig. 7.

THE SIGNAL CONDITIONER

This uses a TL084 quad op-amp and the whole instrument fits in a small sloping panel plastic case, powered from an internal PP7 battery. Two input sockets are provided, the first (JK1) being a 3·5mm jack socket. This accepts a jack to jack lead from a cassette recorder's earphone socket, a suitable level being set with the recorder's volume control. The second input socket (SK2) accepts an input from the recorder's auxiliary socket. The latter is usually a 5 pin 180° DIN socket and to prevent the DIN plug to DIN plug lead from being connected the wrong way round, a 5 pin 270° DIN socket is used on the Program Conditioner. On the 180° DIN plug, the two signal output pins (usually pins 3 and 5) are connected in parallel so that a stereo cassette deck is used in the mono mode. Most recorders provide the AUX output signal at a fixed standard level, unaffected by the volume control or tone control (if any), but on some machines the AUX level is affected by these controls—in which case a medium volume setting (and no top cut) should be selected.

The input signal is applied to IC1b which acts as a non inverting buffer amplifier when S1 is closed, and as a high pass filter when S1 is open. In this mode, it provides, nominally, a 300Hz corner frequency with 12dB attenuation at 150Hz and 30dB at 50Hz, with little effect at 1200Hz—the lower of the two tones used in recording programs.

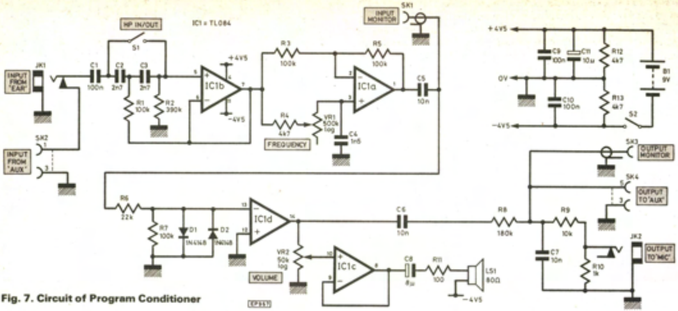

Fig. 7. Circuit of Program Conditioner

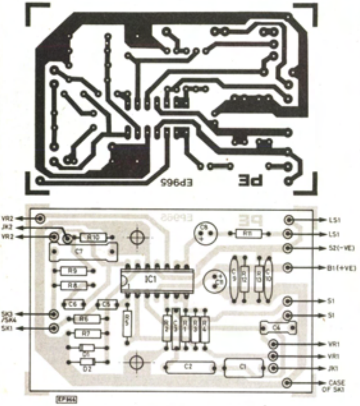

Fig. 8. Printed circuit detail and component overlay

COMPONENTS . . .

Resistors

R1	100k
R2	390k
R3	100k
R4	4·7k
R5	100k
R6	22k
R7	100k
R8	180k
R9	10k
R10	1k
R11	100
R12–R13	4·7k (2 off)

All resistors ¼W 5%

Capacitors

C1	0·1µ
C2–C3	2·7n (2 off)
C4	1·5n
C5–C7	10n (3 off)
C8	8µ 16V
C9–C10	0·1µ (2 off)
C11	10µ 16V

Integrated Circuit

IC1	TL084

Potentiometer

VR1	500k log
VR2	50k log

Switches

S1	S.p.s.t. miniature toggle
S2	S.p.s.t. miniature toggle

Battery

B1	PP7

Loudspeaker

LS1	80 ohm 2in

Sockets

JK1, JK2	miniature sockets
SK1, SK3	BNC connectors
SK2, SK4	5 way 240° chassis mounting sockets

The output of the high pass filter stage is applied to the all pass filter, IC1a. The turnover frequency of this stage (at which the phase shift is 90°) is adjustable by VR1 from about 200Hz to 20kHz. At the latter setting (VR1 fully anticlockwise) the all pass filter is effectively out of circuit since the vast majority of cassette recorders cannot reproduce frequencies this high. Thus, with VR1 set anticlockwise and S1 closed (high pass filter bypassed) the playback signal from the recorder can be observed by connecting an oscilloscope to SK1, the "Input Monitor" socket. Any hum or phase distortion can thus be observed, as can the corrective effect of the two filters. The filtered, phase corrected signal is fed to IC1d where it is sliced to produce a near ideal squarewave, as this op-amp is run open loop a.c. coupled as a comparator. IC1c, controlled by volume control VR2, provides a drive to the miniature loudspeaker, permitting audible monitoring. This is useful when using the EAR output of the cassette deck, as this cuts out the recorder's internal loudspeaker. R8, 9, 10 and C7 provide shaping, phase retard and level adjustment for the two outputs and the monitor output. The latter (SK3) is provided to permit viewing of the output waveform on a 'scope, whilst the output to a computer's cuts input (or to a second cassette machine for direct copying) is taken from SK4, or from JK2 if the cassette's microphone input is used instead of AUX.

CONSTRUCTION

With a quad op-amp and modern miniature components the circuit board takes up very little space and weighs very little. The p.c.b. assembly is stood off from the front panel as shown in Fig. 9 and the latter includes all circuit components

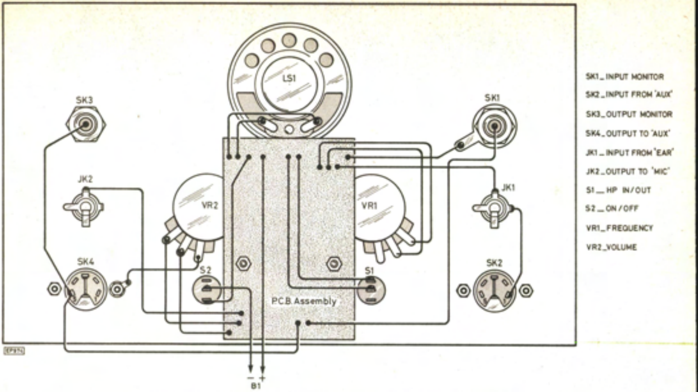

Fig. 9. Placement of case mounted components with board interwiring

except the battery. This combines for neatness and easy access if servicing is required.

Etching details for the printed circuit and the component overlay are shown in Fig. 8. Other layouts, such as on Veroboard, can be used but watch out for op-amp outputs running close to inputs—this could cause instability. The component assembly is built into a plastic box with a sloping front panel 12·5 by 21cm. This provides space for an uncluttered panel layout and just sufficient depth to accommodate a PP7 layer type battery. As the whole instrument draws only about 8 milliamps with intermittent use battery life should be almost indefinite. A photograph of the front panel is shown with appropriate legends.

USE

To obtain the best from the Program Conditioner one really needs to observe the signal at the input monitor socket on a 'scope. The simplest of these suffices, e.g. a single trace instrument with a bandwidth covering the audio band is quite adequate, though good triggering is, as always, important. One can soon tell from the appearance of the trace and the sound from the monitor speaker whether the playback signal is of good quality. If the trace is fuzzy in the vertical direction, this usually indicates hum, as can be verified by switching to a slow timebase setting—switching in the high pass filter should cure this. Fuzziness of the trace in the horizontal direction usually indicates speed variations (wow and flutter), either due to the machine which recorded the program in the first place, or due to the cassette deck being used for playback. The ear becomes quite good at detecting this with practice. There is little that can be done about this (other than using a better cassette deck if the flutter is due to playback) but it does make correct equalisation more important. If the 1200Hz tone looks a bit peaky and triangular, advance the phase equaliser control VR1 from the fully anticlockwise position. You should see the 1200Hz tone become squarer, with faster rise and fall times as it passes through the mean value, when the fundamental is correctly phased with respect to the harmonics.

This is the correct condition, over compensation will result in a return to the peaky, triangular waveform. There will be comparatively little effect on the 2400Hz tone on most small cassette machines as they will already be falling off considerably in frequency response at the odd harmonics of this frequency.

Use of this Program Conditioner can enable a program to be successfully loaded which otherwise might be difficult or impossible; it should then of course be recorded from the computer to avoid further problems. Alternatively, the Program Conditioner can be used to clean up and optimise a program on cassette for rerecording on another machine without tying up a computer at all. Providing the original recording loads, the rerecorded copies should likewise do so. ★