Crafting the signal

In some circumstances, a pure, unaltered audio signal might be all that is required. However, some 'crafting' of the signal is often required and the signal must travel through another maze of components. ASHLEY STYLES looks at some classics to see what is happening.

MANY UNITS DESIGNED for adjusting the frequency response of an audio signal can generate more problems then they are trying to correct. One of the many problems is that of phase shift in which the area of frequency being modified by boost or cut has a shift in phase from the rest of the signal. This can lead to the harmonic structure of the signal having incorrect phase coherence with the fundamental frequencies of the signal and causing unwanted colouration.

The majority of classic valve EQ/tone control units used passive EQ. The audio signal is filtered through inductors and capacitors via suitable variable resistors, then the modified audio signal is amplified to bring it back to the required operating level. Basically, this means that the tonal adjustment of an audio signal is obtained by filtering out what you want to keep and attenuating the rest. Many passive EQ units suffer from incorrect operation if they are loaded incorrectly and not matched to other pieces of equipment in the audio chain.

In 1952, a tone control circuit designed by Peter Baxandall was published in Wireless World. This was to change the way in which tone control circuits were to be designed forever. Although the design was primarily intended for hifi use, the design also found its way into recording equipment. Baxandall realised that by using a gain stage after EQ had been applied to the signal, a percentage of the signal could be fedback to the EQ section, improving the way in which the EQ section functioned. Designers could now use linear law potentiometers, offering linear scales on the controls, together with more flexibility in circuit design and reducing the sensitivity of the EQ unit to external loading/mismatch.

It is often the case that removing what you don't want is more satisfying then boosting what appears to be lacking. A certain amount of tweaking of the EQ can offer a sound that would appear to be phase correct, rather then just tonally accurate. It is all down to what the things on the sides of our head find acceptable.

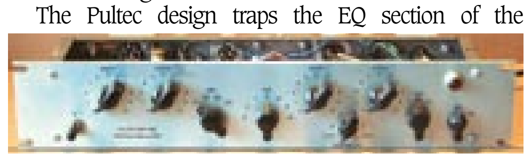
Let's have a look at a few examples of units designed to modify the frequency response of our audio signal.

FONOFILM INDUSTRY KF 571 — A single channel unit with a circuit built around a single E80CC double triode valve and designed to operate in conjunction with other external buffer amplifier units. Therefore the connections to the unit are unbalanced. The input signal is taken directly to the bass/LF section of the passive EQ network and then on to the treble/HF section. A choice of -6, 0, 6dB is then applied to the equalised signal via the first section of the valve. The second section of the valve then operates as a cathode follower output. The audio path is DC coupled from the input, right through to the output coupling capacitor at 0.5uF. Providing the output of the EQ is connected to a high impedance input, then the coupling capacitor is large enough in value to not filter out any audible LF response of the system.

ORTOFON KFS 600 SERIES — The controls of the 600 series are basically the same as those found on the KF 571 — 11 steps of rising/roll-off in each section — but it has an additional mode in the treble range for lift/attenuation. It uses a single E80CF valve that has two sections, one being a triode and the other being a pentode. While the circuitry is very much the same as the Fonofilm, the use of a Pentode section offers a different tonal character.

PULTEC — For many people, the best example of passive EQ tone control is found in the famous Pultec range of units — whether in the MEQ range for mid frequencies, the EQP range for program equalisation, or the HF range for HF correction.

The Pultec design traps the EQ section of the



electronics within a pair of matching transformers. This helps to maintain a constant input/output loading on the EQ network and allows the designer to produce a very accurate EQ section. When compared to many other EQ units, the Pultec offers increased control and flexibility.

The Gain stages within the Pultec design are based around a pair of double triode type valves, both working in push-pull fashion. The output of the EQ network is connected to the input of the first stage via a suitable matching transformer. The amplified signal is then capacitively coupled to the output stage, interfacing to the outside world via the output transformer. A percentage of feedback is taken from the output transformer and applied to the cathodes of the first amplification stage. So, feedback is only applied to the buffer amplification stages, not the combined EQ/amplifier stages.

The Pultec units are self-powered, using the 6X4 rectifier valve to obtain the rectified HT. Maybe it's the valve rectifier that also helps to produce the sound quality we find so enjoyable!

PECLA 1VV — This must be the original recording channel, but unfortunately is of unknown origin. The signal first passes through the compressor section, being very similar in design to the Altec 436. The PECLA offers additional front-end gain to the Altec, together with an extra control for selection of compression ratio. The signal then enters the EQ stage. The circuitry would suggest that the EQ stage is of Pultec program equaliser design. I rece ived this unit as a gift in non-working order. I have since brought it back to life and all I can say is, whatever its origin, it sounds very good.

We often need to correct the perceived dynamic range of sounds and it's well known that compressors and limiters do sound very different. Some units have design aspects that would help to explain this so here are a few examples of valve compressor/limiter designs.

ALTEC 436 — A fairly straightforward and very popular design using just three valves. The

compression is carried out by a single 6BC8 double triode valve with the control voltage being fed to the grid of each half of the valve, via the centre tapped secondary winding of the input transformer and associated pair of input level controls. The rectified audio, used for the control voltage, is extracted from the anodes of the output valve (6CG7 double triode) and then rectified by a 6AL5 double diode. Resistors and capacitors are then used to obtain attack and release times as required. The audio path is push-pull throughout. With such a simple design, there is very little clutter to change the quality of sound passing through the unit.

GATES STA-LEVEL — Designed for use in radio stations, the Sta-level has found its way into many recording studios. In design, it is very much the same as the Altec 436, the main difference being the additional 6V6 pentode output stage of amplification. This helps to increase the response time of the DC control voltage used to adjust the bias of the first stage compression valve.

The type 6386 valve that the Sta-level uses for compressing is the same used in the Fairchild 660/670, so that could offer a clue to its popularity.

CCA LA-1D — A clone of the Gates Sta-level was manufactured by the CCA Company, which was founded by a group of RCA engineers. CCA is better known for its AM/FM transmitters and radio station consoles.

The main difference in the CCA LA-1D is the use of a 12BH7 valve for the output stage. It also offers greater release time flexibility by using two switchable time stages.

FAIRCHILD 660/670 — The 660 mono unit is 6U and weighs 41lbs. The 670 stereo version is an awesome 8U and weighs in at some 65lbs. Originally designed for use with disc cutting systems, these are now some of the most sought after units in the world. Alas, there were not many stereo models produced and their secondhand value reflects this.

Both these units are just so fast in operation. For those of us who can remember, it's almost like using the pre-listen head of a tape recorder to adjust the Vari-pitch of a disc cutting lathe. For most engineers a Fairchild quite simply works. One of the prime reasons for its ability to respond so fast is the very high working voltage of the HT supply rail (some 440V) and the use of so many 6386 compression valves wired in parallel. Indeed, the mono 660 uses 4 x 6386 valves in parallel on each half of the balanced audio signal flow through the unit for 8 in total. The

670 stereo unit only uses 4 in total per channel. With the high cost of these now scarce valves, servicing a Fairchild compressor can be a very expensive affair.

RCA BA-6A — Another unit designed for broadcast use. A design much like the Gates Sta-level, the RCA unit used pentode valves for the audio path. The compression stage used the 6SK7, a 6J7 valve for the intermediate stage and finally the famous 6V6-GT for the output stage/control voltage amplifier. Therefore, the sound quality of the RCA is quite different to that of the Gates and one that you would associate with medium/large-sized Pentode type valves.

TELETRONIX LA-2A (REISSUED AS THE UREI LA-2A POST-1969) — The LA-2a was originally called the LA-2, a product made by the Teletronix engineering company.

The method Teletronix used to compress the audio signal is different to that used in the units we have already mentioned. Instead of using a valve for compression (attenuation), an electro-optical attenuator is employed. Based around the famous T4B electro-optical attenuator, the LA-2 is able to offer a gain reduction of some 40dB with very little distortion.

The T4B attenuator consists of a photo-conductive cell and an electro-luminescent light source, optically coupled together in a light-proof, plug-in case. The amount of light emitted by an electro-luminescent light source relates directly to the voltage applied to it. The varying resistance of the photo-conductive cell is then used to attenuate the input signal, prior to any amplification.

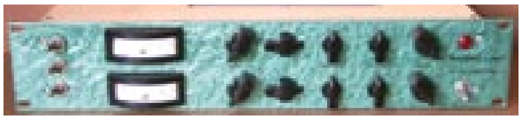
Through the use of such useful components in the T4B attenuator, the unit offers an attack time of 50-100 microseconds from ‘dark’ and just 10 microseconds within the next 30 seconds of use. An additional advantage is the logarithmic release law obtained by using such components — high gain reduction equals long release times and little gain reduction equals faster release times.

The audio input signal is applied to a 12AX7A, both halves in parallel, via the input transformer and peak reduction control. Then onto the 6AQ5 ‘Luminescent’ driver amplifier used to supply the required control voltage to the T4B attenuator.

After attenuation, make up gain is obtained by passing the signal through two halves of a 12AX7 valve and finally through a single-ended push-pull output stage, based around a 12BH7A and output matching transformer.

UNIVERSAL AUDIO LIMITING AMPLIFIER TYPE 176 — This is a cross between the Altec 436 and the Gates Sta-level — still using the 6BC8 valve for gain reduction control (as in the Altec 436) but with the extra stage of amplification courtesy of the 12BH7, as in the Gates Sta-level. Therefore this compressor offers more ‘weight’ to drive the compression stage, which shows through in use. The Universal Audio offers ‘limiting’, not seen on the Altec, and together with switchable compression ratios of 2:1 through to 12:1, the unit offers many more options.

MISSING LINK VC-931 — A stereo/dual mono unit, designed and built by me, which is based around a circuit that is similar to the Altec 436. With the option of using the sum and difference principle, the VC-931 offers a degree of extra flexibility that is not found on the majority of commercially produced units. In this mode of operation and prior to compression, the left and right



channels can be added together to give the ‘sum’ signal and subtracted to give the ‘difference’ signal. These two signals are then passed through the two channels of the compressor. The amount of compression applied to the ‘separated’ sum and difference signals can give some very interesting/pleasing results. The signals are then decoded to provide the original Left and Right channels. Addition and subtraction of the audio signals is accomplished by the use of transformers, much like that in the Fairchild 670. ■

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