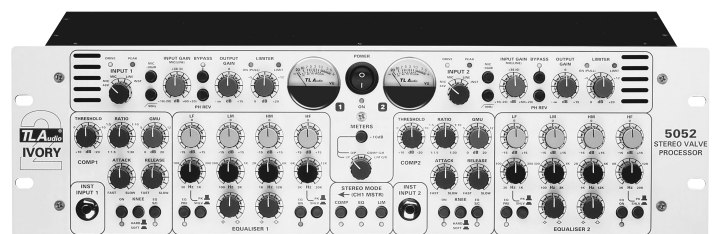


Ivory 2 Series

5052



STEREO VALVE PROCESSOR

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1. INTRODUCTION

Congratulations on purchasing the Ivory 5052 Stereo Valve Processor by TL Audio!

The Ivory 2 Series consists of a range of hybrid valve signal processors, which utilise low noise solid state electronics in conjunction with classic valve circuitry to produce audio processing units offering very high quality signal paths with the unique valve audio character. The units offer comprehensive control facilities, whilst remaining straightforward to operate, and represent excellent value for money.

The 5052 is a dual channel processor with mic, line and instrument inputs, and features a compressor, four band equaliser and peak limiter on both channels. The compressor, equaliser and limiter stages can be operated in 'dual mono' or 'linked stereo' modes, making the 5052 equally powerful for both mono/stereo tracking and final stereo mix processing. A pair of illuminated VU meters monitor the input level, output level, compressor gain reduction and limiter gain reduction for both channels. The optional DO-2 digital output card allows 24-bit analogue to digital conversion via an RCA phono type SPDIF output, with selectable 44.1 or 48 kHz sample rates and the option to clock the converter to an external word clock source.

The block diagram of the 5052 is shown in Figure 1. A solid state, electronically balanced input amplifier is used to achieve state of the art performance with very low noise, low distortion and wide bandwidth. An ECC83/12AX7A triode valve stage (run from a stabilised 150v DC supply) is used as a second stage voltage amplifier, to obtain the classic valve sound and gradual overdrive characteristics. The preamp stage also features 48V phantom power, phase reverse, 30dB pad and a high pass 90Hz filter. A Drive LED gives a visual indication of the signal level through valve stages (and thus the amount of 'warming' taking place) while a Peak LED warns that clipping is about to occur, and this monitors the signal at key points in the audio path.

The compressor section offers fully variable control of threshold, ratio, attack/release and gain make-up, and both hard and soft knee options are available. A 'Hold' facility reduces LF distortion, and a side chain insert point is provided for frequency conscious compression. Like all other TL Audio compressors, the gain control element of the 5052 compressor stage is based around a special transconductance amplifier, which avoids the use of VCAs and helps

contribute to the smooth, open sound of the unit, along with a triode valve stage that forms part of the gain make-up circuit.

A four band valve EQ stage is included, providing up to 15dB of cut or boost on each band. The LF and HF bands provide swept frequency controls with a choice of shelving or peaking response. The LM and HM bands are fully parametric with variable control of gain, frequency and bandwidth. The 'EQ PRE' switch places the EQ ahead of the compressor in the signal path, while the 'EQ S/C' switch places the EQ entirely in the compressor sidechain circuit, for frequency conscious compression. A triode valve stage is located just after the four EQ filter stages.

A peak output limiter is included, and this is located after the output level control. This limiter has a high ratio and a fast attack time, making it ideal for prevention of overload in digital recorders. A threshold control is provided, along with an LED indication of when the limiter threshold is exceeded.

Mic and line inputs are provided on electronically balanced XLR connectors, and the line input is duplicated on an unbalanced mono 0.25" jack connector. Balanced and unbalanced line outputs are provided (on XLR and jack respectively) and these can be used simultaneously. The operating level of the line input can be shifted from -10dB to +4dB (unbalanced) or +4dB to +18dB (balanced) via a rear panel switch, allowing the 5051 to accept very high levels - such as those from a digital recorder. A pair of high impedance front panel instrument inputs are also provided, thus allowing guitars, basses and keyboards to feed directly into the 5052, removing the need for a separate DI box. Finally, balanced insert points (located after the preamp stage but before the processing stages) and compressor sidechain insert points are featured. The former allows other external units to be patched into the 5052 signal path, while the latter permits frequency conscious compression to be performed by patching in an external equaliser.

Please read this manual fully before installing or operating the 5052.

2. PRECAUTIONS

The Ivory 5052 requires very little installation, but like all electrical equipment, care must be taken to ensure reliable, safe operation. The following points should always be observed:

- All mains wiring should be installed and checked by a qualified electrician,
- Ensure the correct operating voltage is indicated on the rear panel before connecting to the mains supply,
- Never operate the unit with any cover removed,
- Do not expose to rain or moisture, as this may present an electric shock hazard,
- Replace the fuse with the correct type and rating only.

Warning: This equipment must be earthed.

FIG 2: FRONT PANEL

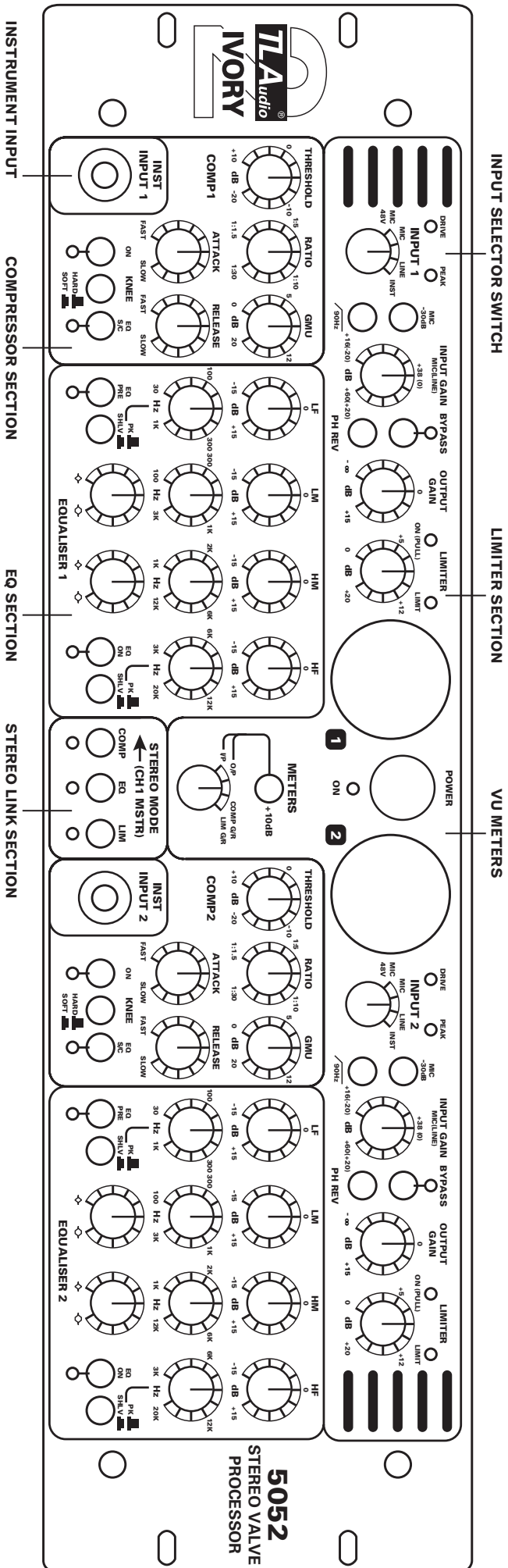
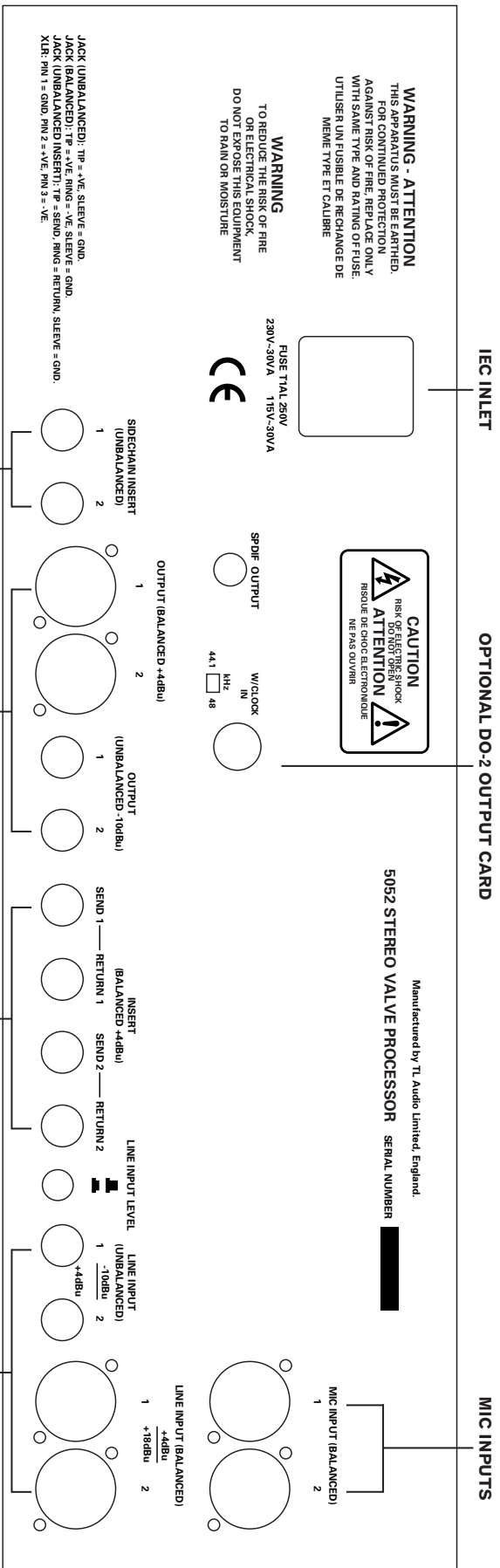


FIG 3: REAR PANEL



3. INSTALLATION

3.1 AC Mains Supply.

The unit is fitted with an internationally approved 3 pin IEC connector. A mating socket with power cord is provided with the unit, wired as follows:

Brown: Live.

Blue: Neutral.

Green/Yellow: Earth (Ground).

All mains wiring should be performed by a qualified electrician with all power switched off, and the earth connection must be used.

Before connecting the unit to the supply, check that the unit is set for the correct mains voltage. The unit is internally set for 110-120V 60Hz or 220-240V 50Hz operation, and should only be changed by an authorised service centre. The mains fuse required is 20mm anti-surge, 1AT rated at 250V. If it is ever necessary to replace the fuse, only the same type and rating must be used. The power consumption of the equipment is 30VA.

Warning: attempted operation on the wrong voltage setting, or with an incorrect fuse, will invalidate the warranty.

3.2 Microphone Input.

The microphone input is via 3 pin female XLR connector, suitable for balanced or unbalanced microphones. The mating connector should be appropriately wired as follows for balanced or unbalanced operation:

Balanced inputs:

- Pin 1 = Ground (screen).

- Pin 2 = Signal Phase (also known as "+" or "hot").

- Pin 3 = Signal Non-Phase (“-” or “cold”).

Unbalanced inputs:

- Pin 1 = Ground (screen)

- Pin 2 = Signal Phase (“+” or “hot”).

- Pin 3 = Signal Ground.

3.3 Balanced Line Input.

The 5052 has a 3 pin XLR socket on the rear panel, which will accept both balanced and unbalanced line level inputs. The mating connector should be appropriately wired as follows:

Balanced inputs:

- Screen = Ground,

- Tip = Signal Phase (“+” or “hot”),

- Ring = Signal Non-Phase (“-” or “cold”).

Unbalanced inputs:

- Screen = Ground,

- Tip = Signal Phase (“+” or “hot”),

- Ring = Ground.

When using unbalanced signals, the signal ground may be obtained by linking pins 1 and 3 in the mating XLR connector. If this connection is not made, a loss in level may result.

3.4 Unbalanced Line Input.

An unbalanced line level input at a nominal level of -10dBu is also provided, on a 0.25" mono jack socket. The mating plugs should be wired as follows:

- Tip = Signal Phase ("+" or "hot").
- Screen = Ground.

3.5 Instrument Input.

Each channel has a 0.25" jack socket on the front panel (see Figure 2). A 2 pin (mono) jack plug is required, which should be wired as follows:

- Tip = Signal Phase ("+" or "hot"),
- Screen = Ground.

Good quality screened cable should be used, particularly for microphone or low level sources, to prevent hum or noise pickup.

3.6 Insert Points.

The insert points are interfaced via a pair of 3 pin, 0.25" jack sockets on the rear of the unit. These points (not to be confused with sidechain insert points – see section 3.7) are located after the initial preamp stage but before the compressor stage, and allow an external device to be patched into the 5052 signal path.

One socket is marked 'Send' (which connects to the external device's input) while the other is marked 'Return' (which connects to the external device's output).

The pin connections for both Send and Return are:

- Sleeve = Ground,
- Tip = Signal Phase ("+" or "hot"),
- Ring = Signal Non-Phase ("-” or "cold").

The insertion points are balanced, and operate at a nominal level of +4dBu.

3.7 Sidechain Insert Points.

The sidechain insert points allow access to the 5052 compressor sidechain (i.e. for frequency conscious compression – see section 4.20) and are interfaced via a 3 pin, 0.25" switched jack socket on the rear of the unit. The pin connections are:

- Sleeve = Ground,
- Tip = Send,
- Ring = Return.

The sidechain insert point is unbalanced, and operates at a nominal level of -2dBu . If used as an additional send only (e.g. as a send to a tape machine or monitor mixing desk), the Tip and Ring should be wired together, to preserve the signal path through the insertion point. If connecting the 5052 sidechain insert to a patchbay it is important to ensure that the 'send' and 'return' are connected together when not in use – if the send/return circuit is left open then this can cause incorrect operation of the compressor stage.

3.8 Line Outputs.

The line outputs are provided on a balanced XLR connector, and also via an unbalanced _" jack. Balanced operation is always preferable to maintain maximum headroom and signal to noise ratio, but can only be used if the following equipment is also capable of balanced operation:

Balanced outputs:

- Screen = Ground,
- Tip = Signal Phase ("+" or "hot"),
- Ring = Signal Non-Phase ("- " or "cold").

Unbalanced outputs:

- Screen = Ground,
- Tip = Signal Phase (“+” or “hot”),
- Ring = Ground.

3.9 Nominal Operating Level.

A switch on the rear panel allows the line inputs to be matched to equipment at a nominal operating level of +4dBu or -10dBu for unbalanced signals, and +18dBu or +4dBu for balanced signals. Most professional equipment requires +4dBu (approximately 1.2V rms), but some small mixing consoles, portable tape recorders or domestic audio equipment require -10dBu (approximately 225mV rms). Digital multitrack recorders have very high signal levels due to the digital scale of 0dBFS usually matching +18dBu in the analogue domain. The 5052 is able to match the operating level of such recorders, using the +18dBu setting. If the operating level is not known, the switch should be set to the position which results in the best signal to noise ratio and least noise while being able to handle high signal levels.

3.10 Ventilation.

The 5052 generates a small amount of heat internally, mainly due to the valve heater. This heat should be allowed to dissipate by convection through the side grills, which must not be obstructed. Do not locate the 5052 where it will be subject to external heating, for example, in the hot air flow from a power amplifier or on a radiator.

3.11 Rear Panel.

The rear panel connectors are identified in Fig.3. Make sure that all settings, mains and audio connections have been made as described above before attempting to operate the equipment.

4. OPERATION

4.1 Input Stage.

The Input Gain control sets the level of the mic, line or instrument input into the 5052 preamp stage. The signal source is selected by the 4-position switch adjacent to the Input Gain control. Input choices are Mic 48V (for condenser microphones that require phantom power), Mic (for dynamic or most ribbon microphones), Line and Instrument. The Instrument input allows high impedance instruments such as guitars or a bass etc to connect directly into the 5052 and eliminates the need for a Direct Injection Box. A wide range of signals can be fed into the 5052, and the Input Gain control also allows the valve stages to be driven to a variable degree. After the preamp stage the signal passes through a triode valve stage positioned between the input circuit and the compression stage. Increasing the input gain pushes more signal level into the valve, thus generating more harmonic distortion and creating that special “valve sound”. This is indicated by the yellow ‘Drive’ LED which will glow more intensely as the level increases. At the same time the output level can be turned down to preserve the same level at the output, so a choice of sounds is available. For a more pronounced valve sound, turn up the input gain and reduce the output gain, and vice versa for a cleaner sound. Don’t be afraid to push the 5052 hard!

As well as driving the valves harder, increasing the Input Gain control setting will also have a pronounced effect on the amount of compression as the threshold will remain constant as the input level increases. If the input gain is adjusted, the threshold can be adjusted accordingly to maintain a similar amount of compression.

4.2 Microphone Input.

When using the 5052 with a microphone source, care should be taken not to apply too much gain at the input. Start with the input gain control set to minimum, and the output master at the mid-point (12 o’clock position). The input gain can then be gradually increased until the VU meter registers about 0VU on normal signal level, when set to read ‘I/P’. The master output level should then be adjusted to produce the required output.

CAUTION: Never switch phantom power on or off, or plug / unplug a microphone with phantom power applied unless the output level control is turned down. Failure to do so may result in a thump in your monitor loudspeakers or PA system.

4.3 Line Input.

A line level signal should already be at about the correct operating level, but this may be checked by monitoring the level with the VU meter set to 'I/P'. The input gain should be adjusted until the meter reads about 0VU at normal audio level.

4.4 Instrument Input.

The front panel instrument input socket is suitable for low level sources such as hi impedance microphones, pick ups or passive guitars, and higher level sources such as active guitars and keyboards. To cater for this wide variety of sources, the 5052 has a large amount of gain available, and care should be taken to avoid applying excessive input gain with a high level source.

4.5 30dB Pad Switch.

Occasionally - when using sensitive condenser microphones - the source signal may be too loud for the input preamp. In this situation, to avoid any overloading or distortion of the mic preamp stage, the 30dB pad can be used to reduce the input gain to a more manageable level. The 30dB pad only applies to the microphone input.

4.6 90Hz Filter.

The high pass filter switch restricts the low frequency response of the preamp, to effectively remove rumble or LF noise from the signal. The filter can be useful in restricting "popping" on vocals or even low frequencies caused by contact with microphone stands or microphone cables. Popping is an undesirable thump that is caused by close-miking certain spoken or sung letters, namely "P" or "B". These particular letters cause a sudden expulsion of air that can result in an audible thump. As this thump has a lot of low frequency content the high pass filter can help to reduce the problem, as can using a pop filter (a device usually made out of nylon material similar to stockings) suspended in front of the microphone.

4.7 Phase Reverse.

The phase reverse switch allows correction of a phase error, which may have occurred in microphone wiring or placement. Phase errors can be due to two

microphones picking up the same signal at the same time. An example of this problem is when recording a snare drum with one microphone on the top snare skin and a second microphone on the underside skin of the snare drum. Because the two microphones are picking up the same signal at the same time, phase cancellation can occur. By inverting the phase of one of the microphones this problem can be rectified. A phase mis-match will probably manifest itself as an apparent loss of bass content when two microphone signals are mixed together or fed to a stereo pair of loudspeakers. If an error is suspected, it is a simple operation to check by phase reversing each channel in turn. The phase reverse is active on mic, line and instrument inputs.

4.8 Drive and Peak LEDs.

The yellow Drive LED provides a visual indication of the signal level through the valve stages, and therefore the extent of “warming” or valve character being introduced. The drive LED will gradually illuminate as the input level or gain is increased, over the range +6dB to +16dB.

The red Peak LED operates as a conventional warning that clipping is about to occur. The operating level of the entire signal chain is monitored, and the LED illuminates when an internal level of around 19dB is reached. Normal operation would be to set the input gain so that the drive LED is regularly illuminating, with occasional lighting of the red Peak LED on transients.

If the input and output gain controls are set to their centre (0dB) positions, the Peak LED will illuminate some 3dB after the Drive LED has reached its full intensity. However, it is possible to add gain further down the chain (i.e. output level gain), which will cause the Peak LED to illuminate at a lower level of Drive. This situation implies that a high level of “clean” signal is present, without driving the valves hard.

4.9 Output Gain.

This controls the level at the 5052 outputs. The nominal level is 0dB at the centre detented position. This control effectively acts like an output fader, and is very useful when recording direct to tape or hard disc through the 5052. You may find that some digital recorders require a good deal of input level in order to register a 0dB reading on their meters (+18dBu analogue usually matches 0dBFS in the digital scale). This is normal, since many digital recorders are designed to preserve headroom and keep the signal well below the 0dB clip point - thus preventing the recorder distorting. The 5052 provides a further 15dB of gain at the output fader to

drive digital recorders. It is important to distinguish the difference between the output gain knob and the gain make-up knob in the compressor section. The gain make-up control is only active when the compressor is switched on. The output gain control is always active but will have no effect on the compression characteristics of the signal.

4.10 Bypass.

The Bypass switch is a convenient way of bypassing the compressor, EQ and limiter sections on either channel so that the original and processed signals can be easily compared. This avoids having to individually bypass each of these sections to hear the original signal. Please note that all status LEDs remain active in these sections even when bypassed, and the meters will continue to register any gain reduction occurring.

4.11 What is Compression?

Compression is an essential but often misunderstood process in modern recording. Put simply, compression reduces the difference between the loudest and the quietest levels of an audio signal. This is known as reducing the “dynamic range” of that signal and is a powerful tool for an engineer helping to avoid overloading, distortion problems as well as raising the level of the quieter parts of the audio signal. Before the introduction of compressors the only way this could be achieved was by “gain riding”, whereby an engineer would control the fader manually in order to try and anticipate very large levels (which might distort the signal) or very low levels (which may get lost in noise). The introduction of compression devices meant that this process could be controlled automatically, allowing the engineer to get on with more productive jobs!

Many instruments and voices have a very wide dynamic range that need to be controlled. A singer, for instance, may be singing quietly one moment and very loudly the next, and unless compression is applied the vocal won't “sit” correctly in the mix, in addition to the problems of distortion on loud passages and noise on quiet ones. Compressors effectively turn down the loud bits and turn up the quiet bits, to achieve a more even and controllable level.

Compressors are often judged by their ability to control the dynamics without creating noticeable audible side effects. Heavy compression can cause the signal to pump or breathe with the onset and release of the compression. Some compressor designs can dull the signal and lose the top end of the signal. The 5052 compressor

design, as with other TL Audio compressors, uses a technology based around a transconductance amplifier rather than a VCA design. This transconductance amplifier design is known for being able to retain the full frequency range and natural character of the audio signal, even when compressing the signal quite heavily. The Ivory 5052 is also capable of more severe compression based around the optional Hard Knee mode if this is desired.

There are other benefits of compression as well as just controlling the peaks and raising the quiet parts, applied properly, it can add punch and excitement to music, as well as fattening up sounds and creating a more professional sounding recording. With the 5052, you have the added benefit of valve stages in the signal path, which create a warmth and presence just not obtainable with solid state or digital products.

4.12 Why Valve Compression?

Valve compression yields a particularly special sound which has become very sought after, particularly with the widespread use of digital products. The reason valve equipment sounds special is due to two things: harmonic distortion and natural compression. When the signal through a valve is increased, it tends to generate a particular type of subtle and desirable distortion, called “second harmonic” distortion. This has the effect of thickening and warming the sound, and the more the level you feed to the valve stages, the more of this harmonic distortion will be produced. You should be able to hear this effect as you increase the Input Gain on the 5052.

Secondly, valves will tend to naturally compress an audio signal, again particularly as the signal level is increased. This itself also contributes to the warmth produced by the 5052.

4.13 Overview of Compressor Operation.

To operate the 5052 compressor successfully an understanding of each control will help to obtain the best results. If you are unfamiliar with the effect of compression it may help to adjust the controls to extreme settings and listen to the sonic effect. Compressors can either be used to enhance the signal and control the dynamic range with as little sonic effect as possible, or they can be used more severely specifically for effect.

4.14 Threshold.

The Threshold is the point measured in decibels that any compression comes into operation. The Threshold control is variable from +10dB in the most anticlockwise position to -20dB at the most clockwise position. Any signal below the Threshold passes through the unit unaffected; while signals above the Threshold are reduced in gain (and are thus 'compressed'). This does depend on the Soft or Hard knee to some extent as the Soft knee is more gradual around the Threshold point.

Unlike some compressors, the Threshold control on the 5052 starts at a 'plus' value in the counter-clockwise position, and decreases to a 'minus' value as you rotate the control clockwise. The reason for this is as you turn the Threshold control on the 5052 clockwise (i.e. towards the negative region) then the degree of compression will increase. We think this is logical, whereas the common method of turning the control 'down' to achieve more compression is not - but beware, some other compressors may work in the opposite direction!

4.15 Ratio.

Once the input signal has crossed the threshold, the degree of gain reduction is determined by the Ratio control. The Ratio control is calibrated in decibels and is simply the change in output level that results from a given change in input level. An uncompressed signal will have a 1:1 compression ratio - every 1dB change in input level results in the same 1dB change in output level. A compression ratio of 1:3, for instance, means that a 3dB change in input level will only give a 1dB change in output level. For more severe compression, simply turn up the Ratio control.

The 5052 offers a wide range of ratios from 1:1.5 (very gentle compression) through to 1:30 (very severe compression).

4.16 Attack and Release.

The Attack time sets how quickly the compression is applied once the threshold has been exceeded, and the Release time sets how quickly the compression is released (and the signal returns to normal) once the signal drops back below the threshold. The 5052 allows fully variable control of Attack and Release times between 'Slow' and 'Fast'. For Attack this covers the range 0.4mS to 40mS, while the Release control spans 40mS to 4S. There is an element of automatic operation of the Attack

and Release on the 5052 - for instance, should a very short transient occur the time constants tend to become shorter, to prevent a slow release leaving a "hole" in the signal after the transient. Also, a fast release setting will be extended by a slow attack setting. Due to this automatic nature of the time constants, the controls are simply labelled 'Slow' to 'Fast'.

The speed of the Attack and Release should in general be able to work with the tempo of the signal. For example if the signal is a snare drum, by monitoring the gain reduction it is possible to set the Release to allow the compression to fully recover (i.e. the gain reduction needle will settle back to 0dB) before the next snare beat. This prevents the second snare beat being reduced in level in comparison to the first. One side-effect of having an incorrect release setting is distortion on low frequency signals, which can particularly occur when using a fast release setting on bass heavy signals - the compressor is forced in and out of compression during one cycle of the waveform, and distortion results. The 5052 has a built-in "Hold" facility which delays the onset of release for approximately 10mS after the input signal falls below the threshold. If distortion is still experienced, a slower release time should be used.

4.17 Knee.

The Knee switch enables the 5052 compressor to be operated in two different modes - soft knee or hard knee. Soft knee mode offers a gentle compression curve around the threshold point, and is traditionally employed to yield a more subtle, musical type of compression effect. The hard knee setting causes the full compression ratio to be applied immediately the signal has passed the threshold point, so tends to produce more pronounced and severe compression.

4.18 Gain Make Up.

While the subjective sound quality of the signal can be improved by compression, the overall signal level will be reduced when gain reduction is taking place. The Gain Make Up control is designed to boost the compressed signal by between 0 and 20dB, in order to bring back the level to the same loudness as the uncompressed signal. Without this control, comparing the original and compressed signals becomes difficult, since there would be a level drop each time the compressor is switched in: therefore it is normal to adjust the Gain Make Up control so that when the 'compressor on' switch is activated, the audio signal remains constant in level.

Unlike the Output Level control, the Gain Make Up control is active *only when the 'compressor on' switch is engaged*. Once the Gain Make Up has been adjusted, use the Output Level control to set the overall output level of the 5052.

4.19 Compressor On.

This switch enables or disables the compressor stage, thus allowing an A/B comparison to be made between the original untreated signal and the compressed signal. Any gain make up applied to the signal only becomes active when the "Compressor On" is enabled. An associated status LED indicates when the compressor is active. The Gain Reduction on the VU meter will monitor the level of compression regardless of the compressor stage being active or non active.

4.20 'EQ S/C' & Frequency Conscious Compression.

The 'EQ S/C' switch places the entire 5052 EQ section into the compressor sidechain circuit, allowing frequency conscious compression to be performed. Once the EQ S/C switch is activated, any frequency boosted on the equaliser section will effectively lower the compression threshold at that frequency. To de-ess, for instance, activate the EQ S/C switch and try boosting the sibilant frequency (normally 3-5kHz) using a narrow bandwidth setting on the 5052 EQ section. This should make the 5052 more sensitive to the boosted frequency, thus compressing it to a greater degree and reducing the sibilance effect.

Please note that the EQ S/C switch overrides the setting of the EQ Pre switch in the compressor section.

If you wish to use an external EQ unit to assist with frequency conscious compression, then the provision of rear panel sidechain insert points on the 5052 allows this. Proceed by patching an equaliser (normally a parametric or graphic type) into the sidechain (the insert point works on a send-and-return principle whereby the insert 'send' connects to the equaliser input, and the equaliser output connects to the insert 'return', thus completing the circuit). Once connected, any frequency boosted on the equaliser will effectively lower the compression threshold at that frequency. Working this way frees up the EQ section within the 5052, allowing it to be used for normal EQ duties.

4.21 Meter.

The 5052 is equipped with a pair of illuminated VU meters. The associated Meter switch enables this meter to monitor each one of four parameters. When switched to 'I/P', the meter reads the audio input level just after the preamp stage. When switched to 'O/P' the meter reads the audio output level, and is calibrated to read 0VU when a +4dBu signal is reached at the main XLR line outputs of the 5052 (or -10dBu at the unbalanced line outputs). The meter when switched to 'Comp G/R' (Gain Reduction) indicates the amount of compression occurring. If the signal is below the threshold, the meter will indicate 0dB: i.e. no gain reduction. As the signal passes through the threshold, the meter will start to indicate the gain reduction at the compressor stage (this will be a *negative* value, so the meter will move to the left, away from 0VU). Note that this reading won't include any extra gain make-up applied.

The final setting on the Meter switch is marked 'Lim G/R', and this separately monitors the amount of gain reduction that the limiter circuit is applying.

A separate '+10dB' switch is provided which reduces the meter reading by 10dB below the actual audio signal level, and this applies to both input and output level metering. This is useful when the 5052 is being fed with large input levels - or is producing large output levels - and the meter needle would otherwise be pressing constantly at its end stop. This situation is normal, particularly if the 5052 is interfacing with digital recorders or mixers where large signal levels are regularly encountered.

4.22 Equalisation.

The 5052 equaliser section has four bands, each with continuously variable cut/boost and frequency controls. The LM and HM bands also have a variable Q (or bandwidth) control that means these two bands are fully parametric in operation (see section 4.25).

Before switching the EQ into circuit, it is advisable to set the cut/boost controls to their centre, or flat, position. Each channel of EQ is brought into circuit with the overall "EQ-ON" push switch positioned to the far right of the unit. "EQ-ON" is signalled by a green LED.

The four bands of EQ per channel are labelled as LF (low frequency), LM and HM (low and high mid frequency) and HF (high frequency). However, the frequency variation available on each band allows overlapping of two, or more, bands into the same frequency range. This arrangement allows maximum flexibility, permitting

reduction of a narrow band whilst simultaneously boosting an overlapping broader band of frequencies, for example. Used individually or in combination, the filter bands allow comprehensive equalisation of any audio signal.

4.23 Cut/Boost Control.

Each of the four EQ bands has a +/- 15dB gain control that is used to apply cut or boost at the frequency selected by the Frequency control. Often, EQ-ing is as much about cutting frequencies as it is about boosting them, and removing excessive bass or mid frequencies from signals to help them sit better in the mix is a common recording technique. On the other hand TL Audio EQs are renowned for being very musical - and very forgiving - so you may find yourself being able to apply a lot more boost than other EQs, and the sound will still retain its quality and not be destroyed - as is often the case on cheap or poorly designed EQ sections, particularly when pushed hard.

4.24 Frequency Control.

This control sets the Frequency on each band that is to be cut or boosted. The frequencies are continuously variable over the following ranges:

LF band:	30Hz to 1kHz
LM band:	50Hz to 1.5kHz
HM band:	1kHz to 12kHz
HF band:	3kHz to 20kHz

4.25 Q (Bandwidth) Control.

A bandwidth control is provided on the LM and HM bands of the 5052, making these filters fully parametric. The "Q" of the filter is a measure of the shape of the frequency response curve, and is closely related to the 'bandwidth' or range of frequencies controlled by the filter. A narrow bandwidth (high Q) setting means that a tight band of frequencies either side of the selected centre frequency is affected by any cut or boost actions, while a wide bandwidth (low Q) affects a much broader band of frequencies.

As a result, high Q settings (about 7 at the maximum setting on the 5052, which roughly corresponds to a bandwidth of 0.15 octaves) are generally used for audio correction or effects - for instance when a single troublesome frequency needs to be

removed without upsetting the rest of the frequency spectrum, or a single frequency needs to be boosted to create a particular unique sound. A proven technique for identifying frequencies is to set the Q control to its minimum (narrow) position, apply a reasonable amount of boost in that EQ band, and then sweep the Frequency control around until the problem (or desired) frequency is found (since boosting will make the frequency prominent). Cut (rather than boost) can then be applied if the frequency is troublesome, and if necessary the Q can be widened to take out a broader band of frequencies.

Intermediate Q settings, say 1 to 3, are generally used to enhance or reduce a broader range of frequencies, typically to make an instrument or vocal stand out - or recede - into the mix. Finally, low Q values (down to about 0.7 on the 5052, which roughly corresponds to 1.4 octaves) provide gentler contouring, or “sweetening” of the response.

4.26 Shelving EQ.

The LF and HF sections offer either ‘peaking’ or ‘shelving’ equalisation by means of a dedicated switch. In peaking mode the EQ band will cut or boost a fixed band of frequencies either side of the selected centre frequency (a little like a parametric section but without a variable Q control) – in this mode the Q is fixed at 1.5. In shelving mode the band will process any frequencies below the selected frequency (in the LF band) or above the selected frequency (in the HF band) with a slope of 12dB per octave. Shelving EQ can be used to “roll-off” low or high end frequencies or to boost all frequencies above or below the one selected by the variable frequency control. This can be effective for “warming up” the low frequencies or adding brightness or “air” to the top end frequencies.

4.27 EQ On.

The ‘EQ On’ switch allows the EQ section to be activated or bypassed for comparison of the equalised and original, unprocessed signal.

4.28 EQ Pre.

The ‘EQ Pre’ switch places the equaliser ahead of the compressor section in the signal path. This enables more flexibility in sound, since the compressor will become more sensitive to any frequencies that have been boosted by the EQ section. The effect may be compared by toggling the EQ Pre switch.

4.29 Output Limiter.

The 5052 output limiter provides responsive yet natural control of output peaks - making it ideal for preventing overload in digital recorders. Digital recorders will distort in a very unmusical way when overloaded to any degree, so an output limiter is an ideal way of preventing over-modulation occurring. A limiter is a more extreme form of compression, and the 5052 limiter is programmed with a very high ratio, short attack time and longer release time, resulting in a fast response "brick wall" action. This means that once the output level of the 5052 has reached that of the limiter threshold, it will be clamped at this maximum level no matter how much more input signal is applied to the 5052.

A threshold control allows the limiter threshold to be varied between 0 and 20dB (this indicates the actual level on the main balanced XLR line output), and the limiter is activated by pulling out the threshold control (since this control incorporates a push-pull switch). A green status LED indicates when the limiter is activated. A second red 'Limit' LED illuminates when the threshold has been crossed and the limiter is actively providing some gain reduction.

The limiter is positioned 'post' the output level control, so the user can be confident that any increase in output gain will be precisely controlled.

A OUV output reading on the 5052 should read approximately -14dB on a digital recorder input (using +4dB connections) although this will vary with different recorders. This means that most digital recorders will require +18dBu of analogue input level to generate a 0dBfs full scale reading. If the 5052 limiter threshold is set to approximately +17dB this would be just below the point where the digital recorder may overload, so would be a useful threshold to limit the output of the 5052. Bringing the threshold down further towards 0dB - so that the limiter is forced into working harder - enables the limiter to be used as a more creative effects device, for example on drums or drum loops.

4.30 Stereo Link Modes.

A key part of the 5052's power lies in the ability to independently use the compressor, EQ and limiter sections in 'dual mono' or 'stereo linked' mode. A bank of three switches located in the bottom centre section of the 5052 allow the user to separately link the compressor, EQ and limiter stages, at which point channel A's

controls automatically become the 'master' for that stage, and channel B becomes the 'slave'. This means that when processing a stereo source or stereo mix, channel A's controls can be used to process both left and right signal paths simultaneously – providing very quick and accurately matched stereo processing. Channel B's controls for those 3 processing stages effectively become inactive, meaning that only one set of adjustments is required to process both signal paths. Please note that only the compressor, EQ and limiter switches and controls are governed by the stereo link mode – other sections such as input/output gain, preamp settings etc remain independent – even in stereo mode – to allow separate gain balancing. To summarise this, here is a table of controls that are not part of the stereo link facility, and thus will remain independent and unlinked even in stereo mode:

Input source selector switch
Input gain control
30dB pad
Phase reverse
90Hz filter
System bypass switch
Output gain control

Note that by providing separate linking switches for compressor, EQ and limiter modes it is possible to link some of these processing stages but not others.

In dual mono mode, the 5052 can process two separate signals (such as a vocal on channel A and a bass guitar on channel B) and provide completely independent control of each.

Stereo linked compression and limiting is essential to avoid imbalances in the stereo image (known as “dips”) to appear on one side of a stereo signal, if the signal exceeds the threshold on that side only. If a compressor/limiter has not been stereo linked, the “dipping” of one channel can sound very obvious and unnatural. In linked mode, if either signal crosses the threshold setting, both channels will react together and will be compressed by the same amount.

4.31 Optional DO-2 Digital Output Card.

The 5052 is designed to accept the optional DO-2 24 bit digital A to D converter card to allow easy interfacing with devices such as sound cards and digital recorders.

The card feeds the converted output signals of channels 1 and 2 to the SPDIF phono output. The sample rate is switchable between either 44.1kHz or to 48kHz, and the card can be clocked to an external digital source via the BNC wordclock input. When clocking the DO-2 to an external source the sample rate setting on the DO-2 needs to be set to match the external sample rate, otherwise correct locking may not occur and audible clicking may appear on the digital output. In terms of gain, the DO-2 will generate a signal level of 0dBfs in the digital domain when +18dBu of output level is generated at the balanced line output of the 5052.

5. GETTING STARTED

5.1 Connections.

There are various ways that the 5052 can be connected into your audio system. The four most common are:

- a) As a vocal or instrument front end
- b) Connected to a channel insert point on a mixing desk
- c) Connected to a group or master insert point on a mixing desk
- d) Connected in-line from the mixer's master outputs to the 2-track recorder

To use the 5052 as a front end, connect the output of the 5052 directly to the line (not mic) input of your console, recorder or sound card. A common mistake is to plug the XLR line output into the XLR mic input of a console. This will cause the console mic inputs to overload very easily and may result in a loss of quality. Once the output is connected, simply feed your microphone into the rear panel XLR mic input (with the +48V engaged if the mic is a condenser type), or feed your instrument into the front panel jack input on the 5052. Recording direct to the multitrack recorder (thus bypassing the console) is a common technique these days as it keeps the signal path short, and of the highest quality. No unnecessary console stages are passed through, thus maintaining quality.

Many mixers have sockets called 'insert points', which allow processors such as dynamics devices and EQs to be patched in-line into the mixer signal path at various points. The mixer's channel insert point usually 'sends' the input signal out directly after the mixer's preamp stage - allowing connection to the line input of the 5052 - and then returns the processed signal from the line output of the 5052 back into the mixer at the same point in the signal path. This is commonly achieved using a special insert cable (sometimes known as a 'Y' lead or split lead - usually a stereo 0.25" jack

connector at one end split into two mono jack or XLR connectors - one for send and one for return). The most likely positions that insert points are located on a mixer are in the channel, group and stereo master sections. Patching the 5052 into the channel insert point means that any signal passing through that channel will pass directly through the 5052. Compressing a vocal already recorded on your multitrack recorder, for instance, can be achieved by connecting the multitrack outputs to the channel tape return inputs of your console, then connecting one channel of the 5052 into the relevant console channel insert point. The off-tape signal will then be fed into the 5052's line input via the mixer insert 'send' connection. The line output of the 5052 connects back to the insert 'return' connection, thus returning the processed signal to the mixer and ensuring continuous signal flow.

Group insert points are used to process sub-grouped signals such as drums or backing vocals. It's common to mix an entire drum kit to a stereo group, and then use a pair of group faders to control the overall level, rather than having to adjust each individual drum level. If you then wish to compress the overall stereo kit signal, you can connect a stereo linked 5052 to the relevant group insert points, using the same 'send and return' technique as the channel insert.

Having processed individual tracks while recording, it is common to apply some compression to the stereo mix while mastering it to 2 track tape, DAT or CDR. Doing this will help fatten the sound further and control levels. Like the channel and groups, the stereo L/R mix buss will normally have a pair of insert points to facilitate this. If not, the 5052 can be connected in-line with the mixer's main stereo outputs, ahead of the master 2 track recorder. The latter method may be preferable as this allows the processors to be connected with balanced connectors (the insert points are usually unbalanced). Connecting the 5052 to the main insert points does however allow the 5052 processing to be monitored as the processor is looped into the output stage of the mixer. If the 5052 is connected in-line, to hear the results of the processing the 2 track mastering machine needs to be monitored. This is possible by connecting the mastering recorder to the mixer's 2-track return inputs and monitoring these returns on the mixer.

The optional DO-2 digital output will allow a stereo high quality A/D conversion at 24-bit from the 5052 on a coaxial SPDIF output. The DO-2 can feed directly into digital recorders such as Digital Multitrackers, Hard Disk Recorders, DAT Recorders, Minidisc and CD-Recorders, bypassing any A-D conversion stages on the way. When connecting the DO-2's SPDIF output it is advisable to use cables less than 5 metres in length and of high quality. The digital output can be used simultaneously with the 5052's analogue outputs.

5.2 In Use ~ Mono tracking mode

Having connected the 5052 - checking that the input operating level switch is at the most suitable setting (see section 3.9) - it's time to put it into action! Here's a simple step by step guide, starting with single channel processing of a mic signal.

1. We'll assume that a condenser microphone is connected to the channel 1 5052 mic input, and the +48V phantom power is engaged by choosing the 'MIC +48V' setting of the input selector switch. The first stage is to set up the gains of the 5052. With the compressor, EQ and limiter stages switched out, start with the 5052 input gain at minimum and output gain at 0dB.
2. Gradually bring up the 5052 input gain until the Drive LED illuminates with the chosen source material - this should also generate a healthy reading on the 5052 VU (when set to 'I/P'). Then switch to meter 'O/P' and check that around 0VU is being produced, and adjust the 5052 output level control if necessary. To get the most benefit from the valve stages it's normal to keep the Drive LED glowing most of the time, with the Peak LED illuminating on transient peaks only. For a cleaner sound you can back off the input gain and thus reduce the drive to the valve stages, which will cause the Drive LED to glow much less frequently.
3. Try engaging the 30dB pad and 90Hz filter switches. You should notice that the pad greatly reduces the mic gain and the filter reduces the LF response of the mic (try those 'B' and 'P' letter sounds as outlined in section 4.6).
4. Now depress the Compressor 'On' switch on channel 1, and select the Meter switch to read Gain Reduction ('Comp G/R'). Using the compressor's controls you will need to adjust settings to suit the source you are listening to. A good starting point is set the Attack and Release to 'Fast', Ratio to 1:3, Knee to 'Soft', and Threshold to +20dB.
5. As you start turning the Threshold control clockwise towards 0dB, the 5052 meter should now register that some gain reduction is taking place. Aim to get around a maximum 3-4dB of gain reduction occurring as a starting point, by lowering the Threshold further if necessary. You should also notice that increasing the Ratio setting causes more gain reduction to occur.

6. When gain reduction is taking place, you should notice that the output level is reduced. By switching the compressor 'in' and 'out', you can compare the levels and the subjective sound quality of the original and compressed signals. With the compressor active, use the Gain Make-Up control to set the level so that when disabling the compressor, there is no level drop. This way you can A/B the original and compressed signals without the levels changing.
7. At this stage if you are unfamiliar with compression you should experiment with each control to see how it affects the sound. Until you are familiar with using compressors it can be difficult to hear these changes as a good compressor will retain the natural sound of the source signal. If in doubt aim to use compression gently as it can be difficult to compensate for over-compression. On the other hand there are no rules, so if extreme settings get you the effect you are after, the choice is yours. Let your ears be the guide.
8. With the EQ set 'flat' (i.e. no cut or boost applied to any of the four EQ bands, engage the channel 1 EQ section and experiment with applying some EQ. You should find that the LF and HF bands are useful for adding some low end 'warmth' and high end 'air', while the mid bands are more useful at removing boxiness or nasal tones from mic sources. However, since there is a large amount of overlap between the mids and the low/high bands, you can use the LM band to accurately EQ frequencies down to 50Hz - so very tight parametric control of the bass end is possible. In the same way frequencies up to 12kHz can be processed by the HM band. Unlike other EQs, you should find that you can apply quite drastic amounts of EQ boost and the unit will still retain its musicality - and not introduce unwanted harshness.
9. Now engage the limiter on channel 1, by pulling out the threshold control. Also switch the meter to the 'Limiter G/R' setting. Start with the limiter threshold set to the maximum +20dB setting, and then gradually bring down the threshold towards 0dB. You should notice that the 'Limit' LED starts to illuminate, showing that some limiter action is taking place, and the amount of limiter gain reduction will be indicated by the VU meter in a similar way that compressor gain reduction is displayed. The limiter threshold can either be set to prevent overload in the piece of equipment following the 5052 in the signal path (typically if it is a HD recorder or CD burner) or it can be used for more severe deliberate effect - see section 4.29.

5.3 In Use - Stereo mixdown mode.

Now we'll assume that the 5052 is connected via its line inputs/outputs to the master insert points of a mixer (see 'Connections' section 5.1). Set the input selector switches on both channels to 'LINE', and activate each of the 3 stereo mode switches (Comp, EQ and Limiter) so that channel 1 can be used as the master for both L and R signals of the stereo mix. Note that only the compressor, EQ and limiter controls work as masters - preamp and in/out gain settings remain independent - see section 4.30.

1. The first stage is to set up the gains of the 5052. With the compressor, EQ and limiter stages switched out, start with the 5052 input gains at minimum and output gains at 0dB.
2. Gradually bring up the 5052 input gains until the Drive LED illuminates with the chosen source material - this should also generate a healthy reading on the 5052 VU (when set to 'I/P'). Then switch to meter 'O/P' and check that around 0VU is being produced, and adjust the 5052 output level controls if necessary. To get the most benefit from the valve stages it's normal to keep the Drive LED glowing most of the time, with the Peak LED illuminating on transient peaks only. For a cleaner sound you can back off the input gain and thus reduce the drive to the valve stages, which will cause the Drive LED to glow much less frequently.

Now follow steps 4 to 9 in section 5.2.

6. FREQUENTLY ASKED QUESTIONS (FAQ)

Q: What type of valves (tubes) are used in the 5052?

A: The 5052 employs 3 dual triode valves, and the type used are Sovtek ECC83 (12AX7WA). A dual triode valve is one that provides two independent valve stages within one glass tube, so while the 5052 has three valves this actually means it has six valve stages: one each in the preamp, compressor and EQ sections, repeated for both channels.

You can in theory use any brand of 12AX7WA in the 5052, but you may find subtle difference in audio quality (and price) between various brands. We

have found the Sovtek brand to offer a good balance of quality and affordability.

Q: What is the life expectancy of the valves in my TL Audio unit, and how do I know when the valves need replacing?

A: We would normally estimate around 4 years, but much will depend on the way the units are used.

Leaving units powered up permanently will certainly reduce the valve life - we would recommend that the unit is switched off at the end of a session (although our use of dc heater supplies reduces the fatigue on the valve heaters). Traditionally, as a valve starts to deteriorate with age, its HF performance will be affected, so that there is a perceived loss of brightness, and the sound tends to become "woolly". TL Audio products make extensive use of localised internal feedback circuitry to ensure that if a valve's characteristics change with time then the performance of the unit is not compromised - thus extending the useful life of the valve.

Q: When replacing valves, is any calibration necessary?

A: Small gain changes may occur when replacing valves, and ideally you should have a technician re-calibrate the unit - this involves checking signal levels at various test points and adjusting if necessary. However, these gain changes are often small - so many users simply change valves and carry on as before. It is advisable to replace all the valves in a unit at the same time, particularly on stereo units where valves of different brands and ages may cause a difference in tone between channels.

Q: When I connect the output of my 5052 to the input of my mixer, I get lots of noise and distortion. Why is this?

A: The most common mistake when connecting up our products is to feed the balanced XLR line output of the TL Audio unit into the XLR microphone input of a mixer. Although both mic and line connections often use XLR sockets, they have different impedances and signal levels. A line level signal fed into a mic input will cause it to distort very easily, since a line output

provides a much higher signal level than a microphone does. Also, mic inputs are of low impedance whereas a line output is generally high impedance: this mismatch often results in poor frequency response and increased noise levels. So always ensure that the line output of any TL Audio product is connected to the line input of any mixer or recorder.

Q: Why are there two types of insert point on my 5052?

A: The sidechain insert point is specifically designed to allow frequency conscious compression such as de-essing. This is achieved by patching an equaliser (normally a graphic or parametric type such as the Ivory 5013) into the sidechain insert point (the tip or "send" of the insert point goes to the equaliser input, while the ring or "return" is connected to the equaliser output). Boosting any frequency on the equaliser then effectively lowers the compression threshold for that particular frequency of signal, so that it gets compressed to a greater degree than other frequencies. Thus boosting the frequency that corresponds to the sibilance in a vocal performance will allow de-essing to take place.

The standard insert points on the 5052 are more like conventional mixer insert points and allow an external piece of equipment to be connected between the 5052 preamp stage and the compressor, EQ and limiter sections. This could be a gate, filter or de-esser, or even another compressor or EQ unit.

Q: When using the 5052 compressor section I sometimes experience some LF distortion when using a fast release time, which disappears as I make the release time longer. Why is this?

A: This effect tends to be experienced on signals with significant LF content, when some gain reduction is taking place. The reason it occurs is a result of the way compression works: As a simple example, imagine you feed a low frequency source signal such as a 20Hz sine wave into the 5052, whose release time is set at its fastest position (40mS). The time it takes for a 20Hz signal to complete one whole cycle is 50mS. Since the fastest release time on the 5052 is 40mS, this means that once the source signal has fallen back below the compression threshold, its gain will return to normal in 40mS, which is a shorter time than the cycle time of the source signal. As a result, the envelope of the compressed signal is distorted as it is forced in and out of compression within a single cycle. Increasing the release time to a figure

greater than 50mS prevents this problem, which is why adjusting to a slower release setting will cause the distortion to disappear. For the same reason, source signals with little or no LF content will not be subject to this effect.

Additionally on the 5052 a 'Hold' circuit is designed into the compressor stage. This delays the onset of the release by around 10mS, and as a result the above effect is minimized.

7. SPECIFICATIONS

Line Inputs:

Balanced via XLR, switchable +4/+18dBu nominal level.

Unbalanced via Jack, switchable -10/+4dBu nominal level.

Gain range +/-20dB.

Maximum input level +26dBu, balanced and unbalanced inputs.

Mic Inputs:

Balanced via XLR.

Selectable +48V Phantom Power.

Gain range +16dB to +60dB (-14dB to +30dB with pad).

Input noise (EIN, 22Hz-22kHz, 60dB gain, 150 ohm termination): -127dBu.

Instrument Input:

Gain range 0dB to +40dB.

Input impedance 1Mohm.

Maximum input level +8dBu.

High Pass Filter:

-3dB at 90Hz.

-12dB per octave.

Effective on all inputs.

Phase Reverse:

Effective on all inputs.

Insertion Point:

Balanced, via normalised TRS send and return jacks.
+4dBu nominal level.

Bypass Switch:

Bypasses Compressor, Equaliser and Limiter sections.

Compressor:

Transconductance amplifier / valve configuration, with dual envelope feed-forward

and feed-back sidechain.

Threshold continuously variable, +10dBu to -20dBu.

Ratio 1:1.5 to 1:30.

Attack continuously variable, programme dependant, typically 0.4ms to 40ms.

Release continuously variable, programme dependant, typically 40ms to 4s.

Gain make-up: 0 to +20dB.

Switchable Hard/Soft Knee.

Sidechain insertion point, unbalanced via TRS jack, nominal level -2dBu.

Equaliser:

Four band, switchable Pre/Post compressor and into compressor sidechain.

LF: +/-15dB, 30Hz to 1kHz.

Peak/Shelve response.

Q = 1.5.

LM: +/-15dB, 50Hz to 1k5Hz.

Q variable 0.7 to 7.

HM: +/-15dB, 1kHz to 12kHz.

Q variable 0.7 to 7.

HF: +/-15dB, 3kHz to 20kHz.
Peak/Shelve response.
Q = 1.5.

Output/Limiter:

Output fader centre detented at 0dB.
Maximum gain +15dB.
Limiter post fader, threshold 0dBu to +20dBu.
Balanced output via XLR, +4dBu nominal level.
Unbalanced output via jack, -10dBu nominal level.

Stereo Operation:

Independent selection of Compressor, Equaliser and Limiter for stereo.
All selected controls stereo ganged, Channel 1 operates as master.

Metering:

Variable intensity "Drive" LED, illuminating over the range +6 to +16dBu.
"Peak" LED at +19dBu, monitoring the signal chain at multiple points.
Twin illuminated VU meters, switchable to Input Level, Output Level,
Compressor gain reduction and Limiter gain reduction.
10dB meter pad switch, effective when monitoring Input and Output levels.

Frequency Response:

Line: +0, -1dB, 10Hz to 40kHz.
Mic: +0, -2dB, 20Hz to 40kHz, @ 38dB gain.
Instrument: +0, -1dB, 10Hz to 40kHz @ 20dB gain.

Distortion:

THD+N typically 0.1% @ 0dBu, 10Hz to 40kHz.
THD+N typically 0.5% @ +12dBu, 10Hz to 40kHz.

Distortion is predominately second harmonic, increasing with

valve signal level and constant over the frequency range.

Valves:

Three off 12AX7A twin triodes, positioned at the Input stage,
Compressor and Equaliser.
Ceramic bases with precious metal contacts.
Stabilised HT and heater supplies.

Power Consumption:

30VA.
Internally set for 230V or 115V AC operation.

Dimensions:

19" rack mounting, 3U high.
483mm (19") W , x 133mm (5.25") H , x 250mm (10") D behind front panel.

Shipping Weight:

9kgs.

8. SERVICE

Should the 5052 require service, it must be taken or posted to an authorised dealer with a description of the fault. Please retain the original packing for possible future use, and ensure the unit is suitably protected during transit. The manufacturer cannot accept responsibility for damage caused during transportation.

The 5052 is supported by a limited warranty for a period of one year from the date of purchase. During this period, any faults due to defective materials or workmanship will be repaired free of charge. The warranty excludes damage caused by deliberate or accidental misuse, tampering, operation on the incorrect mains voltage, or without the correct type and value of fuse fitted. It is the user's responsibility to ensure fitness for purpose in any particular application. The warranty is limited to the original purchase price of the equipment, and excludes any consequential damage or loss. When claiming service under warranty, proof of purchase date must be included with the equipment for repair.

Please record the following details, and retain proof of purchase:

Serial Number.....

Date purchased.....

Dealer.....

FIG 1: BLOCK DIAGRAM

