tfpro P10 The Mighty Twin

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by Ted Fletcher

The Mighty Twin; why it happened and what it is.

Ted has been designing recording channels ever since the first days of the 'Alice' mixer company in 1969. This was the time of the very first private studios, when a few of the top engineers and musicians decided that they could make better sounding records than the 'establishment'. Ted designed a range of mixing desks for smaller recording studios and when Independent local radio came about in the UK in 1973, Ted was there with an economical 'on-air' mixer that became a standard, closely followed by the 'Alice 828' a small sound mixer that set the standards for the next 20 years.

But now all the mixers are copied from each other, and they are all made in the melting pot of China. Digital recording has at last become good enough for the most selective of engineers, and so the individuality, quality and life of recordings are down to the abilities of the engineer and the artist, with a little help from the very best equipment.

The Mighty Twin is the natural progression from Ted's preamp and compressor designs of the last few years, it is the very latest example of that 'very best equipment'; it is a combination of the greatest of microphone amplifiers, the sound and effect of classic optical compression but with new engineering for even greater flexibility, and classic 1980s style equalisers, all in a dual package; a package with weight and presence.

THE DESIGN PHILOSOPHY.

THE FRONT END AMPLIFIER.

While there are some input amplifiers that are fully 'solid state' and still sound good, it is a fact of life that there is a subtlety of sound, a clarity and warmth that comes through when a transformer is used. The P10 input stage starts with a transformer. This is not just any audio transformer; it is a UK manufactured transformer precision wound for the job, mounted in a mu-metal shield. Until very recently Ted has resisted using expensive transformers as good performance can be gained from capacitor coupled solid-state designs, but the latest generation of current mode amplifiers work brilliantly with the transformer, and avoid the problems associated with input coupling capacitors; they tend to make the sound slightly thin, and they are very prone to crackling and noise problems, particularly when used with phantom power.
The transformer becomes a part of the balanced current mode amplifier producing an input amplifier of extreme isolation with superb performance whether it is amplifying a ribbon microphone, or a directly plugged bass guitar.
As this system works so well, all inputs go via the transformer. In a conventional system this would not be possible without 'padding out' the input with resistors, which would increase noise levels, but with the input stage working in balanced current mode, the higher the input impedance, the lower the noise. All the inputs can have the same extended frequency response, minimal distortion and low noise as the microphone input, and have the advantage of hum rejection of a quality transformer.

BIG SOUNDS

There are a number of contributing factors to getting a truly 'big' sound. As well as the input transformer, which has to be of best quality and suitable for the job, there is also phase integrity, wide bandwidth, elimination of the wrong sorts of distortion and last but not least, a suitable overload margin.

PHASE…. Ted has always maintained that smooth phase response over the whole frequency range is of great importance. One of the ways of achieving this is to maintain good frequency response, and the circuitry of the P10 is ideal for this. A flat response from 8Hz right up to beyond 40KHz ensures that the phase response within the accepted audio band is kept controlled and smooth.

BANDWIDTH… Unless the filter is switched in, the low frequency response of the P10 is not only 'flat' down to 8Hz, but below that, the response falls gently. At the other end of the spectrum there is still good response up to 100KHz and here again the response falls smoothly.

DISTORTION. While all electronic designs of the 21st century publish wonderful distortion figures, it is not generally appreciated that while 0.1% distortion might be acceptable for 2nd order (2nd harmonic) distortion, if such distortion occurred on the musical signal evenly as 3rd order (third harmonic) distortion, then it would be obvious and unpleasant. The reality is that distortions as low as 0.005% can be audible. To make matters worse, there is another type of distortion that never features in the specifications; this is impulsive distortion and is particularly present in designs that use integrated circuits operating with gain. This is the type of distortion mainly responsible for the lack of 'body' in the sound of less expensive audio equipment; a sound that can never be improved with effects or EQ. Impulsive distortion is caused by the integrated circuit producing very slight momentary distortions, mainly during transient (sudden impulsive) signals. Within the design of the P10, integrated circuits are used, but they operate only as impedance converters or as very low gain devices to eliminate this sort of problem. To give even greater protection from impulsive distortion, the make-up amplifier after the compressor is actually a discrete transistor amplifier operating at high voltage in pure 'class A' mode. This type of amplifier has no 3rd order distortion at all, and should overload occur during transients, the distortion produced is pure 2nd order.

OVERLOAD MARGIN.

During the 1970s Ted designed a range of mixers for broadcast use by the Independent Broadcasting Authority and the BBC. One of the senior engineers at the IBA was of the opinion that all microphone

amplifiers should have an input overload margin of at least 30dB. (It is normally about 22dB). This was introduced into the spec, and consequently the amplifiers were easy to use and very flexible. The theory behind this is that speech is full of transient signals that one does not normally hear or notice, but if the input amplifier does not treat them properly then the sound will have a 'rough' quality. In practice this proved to be correct, and ever since then all Ted's microphone amplifier designs have had this very high overload capability.

THE VARI-PHASE' CIRCUIT.

Multiple microphone recording has never been easy. In the earlier days in the 1960s it was fashionable to separate everything possible so that the audio signals were kept apart from one another. One of the reasons for this was that when two microphones pick up the sound from a single instrument, it can cause problems with phase distortion.
More recently it is more natural sounding to have less separation and to use combinations of synthesised and real sounds, so this problem can be more acute. The P10 has a near unique circuit that can alter the phase of the microphone signal so that a better 'match' can be achieved when two microphones 'hear' the same source. The P10 'VARI-PHASE' can shift the phase of the microphone circuit all the way from 0 to 180 degrees in a single sweep, and then on again another 180 degrees by using the 'phase shift' button.

THE VU METER

The 'VU' meter (Volume Units) was originally developed in the Bell Laboratories in the late 1930s. It was an attempt to get a good indication of the perceived loudness of an audio signal. The VU does this quite well, but can cause difficulties for the inexperienced user because of the audio 'underread' that it gives; for a simple basic tone, like an organ note, the VU reads correctly. But for most signals, and particularly speech and certain music signals, the meter indicates a much lower level than is actually present. For speech signals, the

underread can be as much as 14dB. For most music signals it is around 6dB. And for piano it can be as high as 10dB.
The meter on the P10 is calibrated so that for a steady tone of 0dB, the meter will read '0'dB. The overload margin of the P10 immediately before the output gain control is better than 22dB, so it is safe to peak the meter to the '0' or even the '+3' mark with no distortion even when the signal has high transient content.

THE COMPRESSOR/LIMITER

The optical compressor has been around for 50 years now. The first ones were simple cadmium sulphide cells (that work as light-dependent resistors) with a flashlight bulb close to them, driven from an audio power amplifier. In the 60s designers in the USA developed variable mu tubes that produced the compression effect beautifully, but there was still a certain 'sound' to optical compressors, and following on from Fairchild in the USA, Ted designed a simple solid state version with controllable attack and release times, but it was not until 1993 that this design was produced commercially as the original 'Joemeek' compressor, the SC2. This type of compressor has pointed the way for all modern optical compressors, with their effective sound that is so preferable to all other types.
One of the continuing problems of these simple optical compressors was the limited range of compression ratios available; another was the repeatability of the compression effect.
After considerable extra design effort involving a new look at the way a true 'class A' amplifier can be integrated into the compressor, Ted has produced the P10 circuits, these are compressors where by careful design, a vast range of compression ratios can be achieved from near linear right up to 100 to 1 or even higher. Additionally, by using a different type of light cell, good standards of matching are possible between compressors.
Ted has also had to design new circuitry for compression metering, taking into account the extreme range of ratios.
The Edward.
The TFPRO P8 or 'Edward' utilised those earlier historic type designs, but additionally, Ted was able to emulate both the 1176 and the LA2A compressors by careful manipulation of the attack and release times and shapes. That valuable work has been carried

forward into the design of the P10.

THE EQUALISER

The TFPRO P9 or **'Ted's definitive equaliser'** is already a successful TFPRO product and is an ideal and valuable tool for record production mastering. To attempt to integrate a full 'class A' inductor tuned EQ into the P10 would have been prohibitively expensive and a bit like re-inventing the wheel! So Ted decided to use a 1970s style equaliser based on gyrator circuits. This type of equaliser has a distinctive sound, and because the bandwidths of the frequencies are reasonably tight, it can be a very useful EQ for tracking in record production.
On of the design features is that the frequencies intentionally 'overlap' so that unusual effects with phase can be achieved by boosting and cutting at very close frequencies.
The arrangement of the frequency bands is unusual; the LF section is not the usual 'shelving' type, it is a peaking equaliser with variable frequency, as are both of the mid frequency bands. The HF section has a shelving characteristic, but with a switch to select the turn-over frequency.

THE FRONT PANEL CONTROLS

INPUT SELECTOR selects the type of input. INSTRUMENT IN is via a jack socket on the front panel, LINE IN is via both an XLR socket and a TRS jack socket on the rear panel, Normal microphone input is for dynamic and ribbon microphones and is via the XLR Mic In socket on the rear panel, and the 'CAP MIC' input is via the same XLR connector.

0/180 is the main phase change button, giving phase reversal to all signals.

VARIPHASE is a rotary control giving variable phase shift from 0 degrees up to 180 degrees of the output of the first phase-shift amplifier. The phase shift is correct for frequencies above 250Hz, below that frequency the effect is progressively reduced.

FILTER is a push-button that introduces a precision 12dB per octave high-pass filter with a turnover frequency of 75Hz.
GAIN is a rotary control used to set the gain of the front-end amplifier system. The calibrations are in dB gain for the normal microphone input. If phantom power is selected ('CAP MICS') then the gain is reduced from the marked calibration by 15dB. When 'INSTRUMENT' is selected the gain difference is 20dB and for 'LINE' signals, the calibration '30' corresponds to unity gain.

THE METER is a dual purpose meter. Normally it reads the audio level immediately before the output gain control. When the 'READ GR' button is pressed it reads the compressor gain reduction, so the normal position of the needle is at '0'. The meter type is a VU meter. NOTE, this is NOT a peak meter.

LINK is a push-button that switches the compressor controls in channel 2 so that they are operated by the controls of channel 1. This is for stereo operation of the compressors. When this switch is pressed, the 'type', ratio, attack and release controls for channel 2 are inoperative.

RATIO is a rotary control that sets the compression ratio of the compressor. A ratio of 4:1 means that when the compressor is operating, an increase of volume level of 4dB at the input will cause an output level rise of 1dB. Ratios above 20:1 make the compressor operate as a limiter.

COMP is a push-button that engages the compression sidechain bringing the compressor into operation. A blue LED shows that the compressor is engaged.

ATTACK is a rotary control that controls the time that the compressor takes to operate.

RELEASE is a rotary control that sets the time taken for the compressor to return to normal gain after it has operated.

TYPE is a 4 way rotary switch to select the 'type' of compressor needed. The switch selects the TFPRO emulations of a simple VCA type compressor, the 1176 FET compressor, The LA2A (Urie) compressor and the 'Green Box' type.

LF is the low frequency equaliser control giving lift and cut at the selected frequency.

LF FREQ. is the control that sets the centre frequency for the LF control to lift or cut.

MF1 is the lower of the two middle frequency controls.

MF1 FREQ. sets the frequency for the MF1 control to lift and cut.

MF2 controls are similar.

HF is the lift and cut control for high frequencies.

HF LO is a push-button that lowers the frequency of operation of the HF control.

EQ IN/OUT is a push-button that engages the equaliser. When the button is in the 'out' position, the EQ is completely by-passed. When the button is pressed in, and orange LED shows that the EQ is active.

OUT GAIN is a rotary control setting the output level of the P10. When the control is set to '0' the output level corresponds to the indication on the VU meter (read the notes on VU meter.)
LED level indicators. A green LED shows when there is signal present in the channel (approx-26dB) A red LED shows when the internal levels of the P10 are within 6dB of clip; when an overload is possible.

POWER SWITCH. This is a conventional power on/off switch

THE REAR PANEL

All audio terminations on the P10 are balanced.

The standard connections for XLR connectors are:
Pin 1, screen or ground.
Pin 2, Audio 'hot' or +ve phase.
Pin 3, Audio 'cold' or ve phase.

The protocol for TRS Jacks is:
Tip, 'hot' or +ve phase.
Ring, 'cold' or ve phase.
Body, screen or ground.

Microphone input is by XLR connector only.

Line level inputs are on both XLR and TRS (tip, ring, and sleeve) connectors.

Insert send is the output of the mic amp and phase correction circuits. This is a balanced audio output on TRS jack.
Insert return is the feed direct to the input of the compressor. It is a balanced audio input on TRS jack.

Channel outputs are on TRS jack and on XLR male socket. The outputs are resistively buffered and can be connected to 2 destinations at once.

Also on the rear panel is a push-button labelled 'Ground lift'. Normally the chassis of the P10 is connected to audio ground. Under some conditions this may cause a hum due to the use of unbalanced connections or incorrect studio wiring. Disconnecting the chassis from ground can sometimes cure 'ground loop' hum, but it is essential that the chassis is connected to ground before trying this cure.

Power to the P10 is via an IEC connector. The fuse holder may be rotated to change the operating voltage from 230 to 115VAC.

HINTS AND TIPS ON USING THE P10

To avoid deafening 'clonks' make sure that gain controls are turned down before plugging and unplugging microphones. It is a good move to switch the input to 'LINE' before connecting mics.

When using capacitor (condenser) mics, allow a few minutes before turning up the input gain; they sometimes take some time to 'warm up'.

Start with the input gain low, switch the compressor and EQ out and then wind up the gain to get a good signal level, only start to use compression or EQ once you know the level is about right.

Make sure that the phase switch is not pressed and that the 'VARI-PHASE' is set to '0' at the start.

For human voice, a VU level of -3 peak is likely to be right. For music signals the VU needle can go into the red without harm. Use your ears rather than your eyes!

Use the compressor sparingly. Over compression cannot be undone!

Use the equaliser sparingly, remember that 10dB of lift reduces the overload margin by 10dB, so keep watching the volume level!

TECHNICAL SPECIFICATIONS

INPUTS

XLR input, transformer balanced, normal mode 800 ohm impedance to suit 200 ohm or lower microphones.
In phantom power mode, 2Kohm impedance and 48V phantom power.
Input level from -80dBu up to 0dBu.
LINE input on XLR and jack, floating transformer balanced 20Kohm

impedance. Input level -30dBu up to +28dBu.
Instrument input, front panel TRS jack, transformer balanced 20Kohm impedance. Input level -45dBu up to +8dBu.
All inputs are wideband current mode.

OVERLOAD MARGIN
Greater than 30dB on mic inputs.
Other inputs approx 25dB.

GAIN
Max system gain 85dB
Insert level nominally -10dBu.

NOISE
Line in normally 90dB below operating level.
Mic in 126dB below input at 50dB gain or more, measured RMS 20Hz to 20KHz across 200 ohm resistor. Self noise 132dB below input.

HARMONIC DISTORTION
Normally within 0.002% Harmonic distortion is affected by compressor settings (this is normal).

AMPLITUDE FREQUENCY RESPONSE
All inputs to output, +0 -1dB 10Hz to 38KHz (wideband response falls slowly beyond 40KHz to approx -6dB at 100KHz.)
High-pass filter 3dB down at 75Hz.

PHASE CONTROLS
180 degree phase shift via noiseless all-pass filter.
Full variable phase on frequencies above 250Hz via variable all-pass filter.

OUTPUTS
All outputs balanced 150 ohm source impedance nominally 0DBu.
Max output +24dBu.

COMPRESSOR
Class A amplifier operating in 'Wien Bridge mode' Variable ratio 1.2:1 up to approx 25:1.
Attack time variable from 0.5mS up to 20mS, not calibrated (varies with compression type).
Release time variable 250mS up to 2 seconds.

EQUALISER
Gyrator controlled LF, MF1 and MF2 bands with overlapping frequencies.
HF section shelving.
Max cut and boost 12dB.
Gyrator 'Q' (quality factor) value approx 1.5 for all frequencies.
By-pass switch with indicator.

POWER
5 watts.
IEC power cable.
Reversible fuseholder for 230VAC and 115VAC operation.

HOUSING
3U rack case with heavy gauge aluminium front panel.
Rear housing steel plate.
Depth 300mm.
Weight 2.8 kgs

REGULATIONS AND SAFETY.

The P10 is designed and built in the UK to conform to all safety regulations.
Within the EU the P10 easily meets the requirements for electrostatic and electromagnetic emissions and conforms to all safety requirements of the European Common Market: The 'CE' mark indicates compliance.

WARRANTY

Should the equipment fail in any way, please return it in its original

packaging to the supplier.
The unit will be inspected and repaired, and returned through the supplier.
The warranty covers all defects encountered in normal use and excludes physical damage and misuse.
The warranty period is for two years from the date of purchase.

DECLARATION OF CONFORMITY
This analogue audio equipment conforms to the standards required by the European Economic Community.
The EC Harmonised standards that have been applied are;
a) Electrical equipment (safety) Regulations 1994 (S.I. 1994/3260)
b) Electromagnetic Compatibility Directive (89/336/EEC) incorporating (S.I. 1992/2372)

This TFPRO equipment is designed and manufactured in the UK

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