Eventide H949 Harmonizer

MANUFACTURER'S SPECIFICATION
(preliminary)
Input characteristics: impedance nominal $10k\Omega$, maximum level +24dBm. Level for full dynamic range is from −10dBm to +24dBm.
Output characteristics: impedance nominal 150Ω. Suitable for driving 600Ω or greater at +18dBm. Electronically balanced.
Distortion: less than 0.15% at 1kHz, reference output level.
Dynamic range: greater than 96dB from clipping to noise floor.
Pitch variation: one octave up, two octaves down, continuously variable. 4-digit readout indicates precise ratio.
Delay: main output in pitch change mode, 0 to 100ms in 50ms steps. In delay mode, 0 to 393.7ms in 6.25ms steps. Delay only output, 0 to 393.7ms in 6.25ms steps.
Frequency response: at any delay, unity pitch ratio, 20Hz to 15kHz ±1dB. No degradation with increasing delay.
Dimensions: requires 3½ x 19in (889 x 482.6mm) panel space. Extends 11¾in (298.5mm) behind panel.
Power requirements: switchable between 115V ac, 50/60Hz, and 230V ac, 50/60Hz. Nominal power dissipation 45W.
Remote control: provision has been made for control by microcomputer using the IEEE standard interface buss (IEEE 488/1975). The *HK940* keyboard can be used to control the pitch ratio in discrete musical steps. Option 01 mono keyboard controls one harmoniser; option 06 polyphonic keyboard controls up to three harmonisers. An input is provided to phase lock the harmoniser to any synthesiser. A 3V peak-to-peak signal is required. The pitch may be varied by a control voltage input in the 0-5V to 0-15V range (internally selected).
Price: £1,295.
Manufacturer: Eventide Clockworks Inc, 265 West 54th Street, New York, NY 10019, USA.
UK: Feldon Audio Ltd, 126 Great Portland Street, London W1.

THE Eventide *H949 Harmonizer* is not only a pitch shifter but provides a number of other interesting facilities as well. The pitch changing facility itself covers as far as one octave up in pitch, or two octaves down with a choice of two algorithms—a feature which can save the situation where pitch shifting gives severe distortion on a particular programme material. In addition to wide range pitch shifting there is a facility called 'micro pitch shift' which allows the amount of shift about zero to be finely set.

A single input provides two outputs, one is always the input signal or a delayed version of the input signal with the delay selectable in 6.25ms increments, up to a nominal 393.75ms; while the other, called the main output, is also capable of simply being a delayed output but with a separate delay setting over the same range, or alternatively the special effects output.

When in the special effects mode the main output may also be delayed but only in increments of 50ms with a maximum delay capability of 350ms. The effects available in addition to pitch shifting and delay are a random delay feature, flanging, reversing the input signal in time and repeating a section of the input signal.

To widen the scope of the unit two feedback paths are provided, one from the delay output to the input and the other from the main output to the input. Independent feedback level controls have high and low frequency boost/cut controls common to the feedback path.

The unit has a 19in rack-mounting chassis only two units (3½in) high and the controls are divided into: an input section, feedback section and pitch ratio section, which has logically arranged pushbutton switches in horizontal array below these sections.

The input section contains the input level potentiometer, a locking pushbutton to activate the repeat function and four LED input level indicators. The top one is red and identified as 'limit', the three lower green LEDs identified as 'normal' and a yellow LED as 'present' which is illuminated in the presence of an input signal.

Within the equaliser section are two feedback level controls for the main and delay outputs, plus two further potentiometers for controlling high and low frequency equalisation in the feedback path, either boost or cut is available in both cases.

Turning to the harmoniser section a large knob driving a 270° potentiometer sets the pitch shift which is displayed on a 4-digit, 7-segment display in terms of pitch ratio. To the right of the pitch control three interlocked pushbuttons select the control mode which may be either manual via the front panel control, from an external keyboard or, from the sum of the manual control and an external control voltage.

This leaves the horizontal row of pushbutton switches at the bottom of the panel. To the left is a 'line' pushbutton which simply connects the input to the output, bypassing all the electronics—I'm not always happy about this arrangement because of possible difficulties with changes in loading.

Next a set of six locking pushbuttons for setting the delay time of the delay output. These insert 6.25ms, 12.5ms, 25ms, 50ms, 100ms and 200ms delay thus providing increments of 6.25ms up to a total of 393.75ms.

A 'function select' button follows which operates in conjunction with the next four buttons which are interlocked and each one has two functions depending upon the position of the function select button. When the function select button is depressed the unit is in the pitch change mode and the next four buttons select normal or external pitch change over the range one octave up to two octaves down, or micro pitch change, sharp or flat over about ±10% range. With the function select button out, the four further pushbuttons select the remaining functions: delay, random delay, flanging or reverse. The next button selects one of two algorithms in the pitch change mode with a further six buttons selecting the delay in the main output by the same means as the delay in the delay output. However, the 6.25ms, 12.5ms and 25ms buttons may only be used in the delay mode of the main channel.

The rear audio input and outputs are in the form of electronically balanced connections via XLR connectors. The mains power input is a standard IEC connector with a built-in voltage selector/power fuse, clearly identified. An 8-pin DIN connector provides for external keyboard control of pitch shift in discrete musical steps and the remaining connections are in barrier strip form.

A feature of the interface connections is that they allow a servo type tape recorder to have its speed controlled in a manner inverse to the pitch change. For instance: speech may be speeded up or slowed down while retaining the correct pitch. These servo outputs include the common servo frequencies of 19.2kHz, 9.6kHz and 60Hz (alternatively 50Hz). Additional connections include external control of pitch by a control voltage, bias output, reference frequency and a further single keyboard connection.

Space is provided for connection to a microprocessor via a standard IEEE buss. This interfaces all front panel controls except the input gain control and the feedback gain and equalisation controls.

All the electronics within are on good quality glass fibre printed circuit boards, interconnected by ribbon cables and connectors. Two large boards covering virtually the base area of the unit house a mass of integrated circuits with a further board supporting all the readily replaced front panel controls. As it's a prototype no servicing information is available and I hope the manufacturer will identify components in the production version. But even so for a prototype the component layout was very tidy and the standard of construction good.

Input and outputs
The electronically balanced input was found to have a common mode rejection ratio of

82 ▶

54dB at frequencies below 1kHz with the input impedance high at 530kΩ across the balanced input, or 260kΩ with one leg of the input grounded. This input impedance was maintained at any input gain setting with +2dBm required to illuminate the top LED level indicator at maximum gain and the maximum input level that could be handled was +21dBm which is quite satisfactory.

The main output and the delay output were similar in performance and both electronically balanced, the maximum output was +20dBm loaded into 600Ω from a source impedance of 150Ω in each leg of the outputs.

When loaded into 600Ω the maximum gain of the unit was +13dB at 1kHz, there being a straight wired connection between the input and both outputs in the bypass mode.

The peak sensing level display has a satisfactory peak hold time and a fast attack time which is vital for digital devices. However at 1kHz the red overload LED illuminated at 1dB above input clipping with the top green LED illuminating at only 1dB below clipping, which leaves a rather small margin of error. The lower green LEDs illuminated at 5dB and 15dB below clipping respectively. However it was found that the level display was highly sensitive to frequency and in this respect it was not satisfactory; furthermore, it was quite possible to run into severe output overload with no overload indication.

Frequency response and noise

The overall frequency response (**fig 1**) shows the top trace at maximum level in the pitch shift mode with zero pitch shift in operation. The middle trace is the same as the top but with the level 10dB lower; it can be seen that the anti-aliasing filter becomes highly effective at 15kHz with a substantially flat response down to 20Hz.

The lower trace depicts the frequency response at the same level but with x2 pitch shift in operation. Remembering that as far as the output is concerned the frequency scale must be multiplied by two, the output frequency response is ±2dB from 40Hz to 10kHz which is not too bad for this type of effects unit.

Noise in the output was independent of the gain control setting but differed between the main output and the delay output. Naturally the noise in the main output depended upon the function selected, so, the figures in **Table 1** show the worst case condition when referred to the maximum output level of +20dBm.

TABLE 1

MEASUREMENT METHOD	DYNAMIC RANGE DELAYED OUTPUT	DYNAMIC RANGE MAIN OUTPUT
Band limited 22Hz to 22kHz rms	90.0dB	96.0dB
A-weighted rms	93.5dB	96.0dB
CCIR-weighted rms	86.0dB	83.0dB
CCIR-weighted quasi-peak	82.0dB	80.0dB

These figures represent a very good performance and no significant mains frequency components were present in the output with all spurious clock frequency components at least 100dB below full output.

Distortion

The second and third harmonic distortion was investigated in all modes and found to be worst in the delay mode; this condition at +20dBm output was shown to be satisfactory as depicted in **fig 2**. As may be expected the level of distortion, particularly at low frequencies, fell with falling output level.

When pitch shifting it was possible to have many non-harmonically related tones in the output and these depended upon the algorithm selected and the nature of the signal being handled in addition to the degree of pitch shift.

It is quite impossible to put a number to these spurii which are common to all pitch shifting systems, but it is fair to say that most programme material can be pitch shifted for effects without any unpleasant results if care is taken.

The degree of pitch shift available was found to be 0.218 downwards, as far as 2.072 upwards, with the latter producing unpleasant subjective effects when the 2.000 mark was exceeded. The accuracy of pitch shift indication was excellent with the indication within one digit of the display.

In the micro pitch shift mode the available

84 ▶

FIG. 1
EVENTIDE H949
FREQUENCY RESPONSE

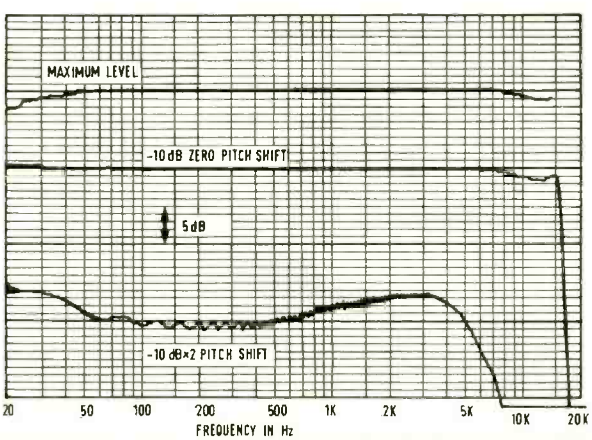


FIG.2
EVENTIDE H949
2nd & 3rd
HARMONIC AT 20dBm
DELAY MODE

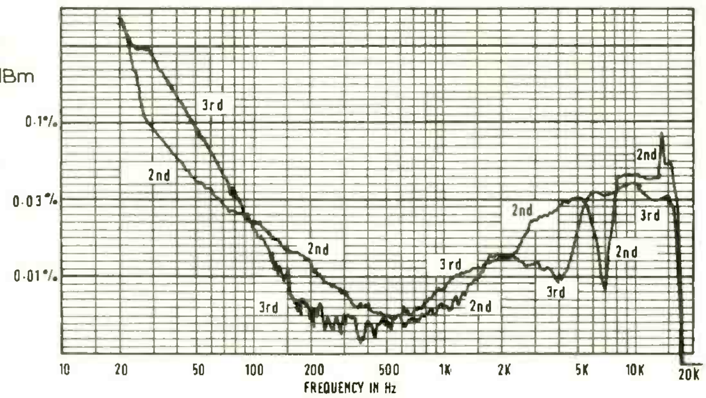
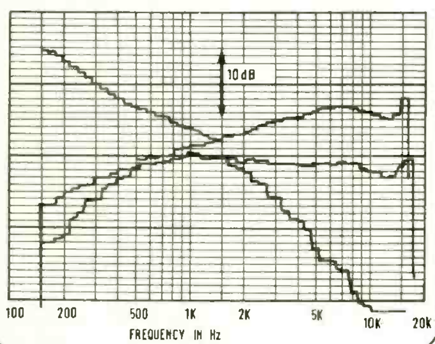


FIG.3 EVENTIDE H949 FEEDBACK EQUALISER



range of shifting was 1.09 down to 0.910 with the indication accuracy again within the readability of the digital display.

Likewise checking the accuracy of the delay steps showed the calibration accuracy was within 0.5% allowing for the 70μs inherent delay.

Regarding the feedback equalisers their frequency response at the mid position of the controls and the extreme positions is shown in **fig 3**, showing that there is more than ample cut and boost, but, the mid point of the bass equaliser is not satisfactory.

Subjective impressions

In the pitch shifting mode this unit can produce some very unusual and interesting effects but some care is needed if large pitch shifts (particularly upwards) are to be achieved without unwanted side effects. The incorporation of two pitch shift algorithms is particularly useful for overcoming such difficulties which are inherent in pitch shift systems.

The reverse facility is probably most useful on percussive sounds and it appears that this function takes blocks of input signal and reverses them in time block by block.

Similarly the repeat function works on 400ms long blocks of sound, repeating them for as long as required. This can be very effective when repeating short sequences of rhythm. Flanging was most natural and effective, as it was possible to stop the flanging at any frequency offset by pressing the external keyboard button; the flanging continues from this point when the button is released.

Summary

This prototype unit provides a new collection of effects at a most moderate cost and I feel it will be a very popular unit when it comes into production later this year.

Being a protytype there are of course a few shortcomings, but, I trust the manufacturer will put these to rights in the production version. **Hugh Ford**

MANUFACTURER'S COMMENT

1) Your criticism of the in-out switch is valid in that it is possible to cause changes in the loading of succeeding circuitry, but we feel the diagnostic advantage of such a hard wired isolating switch to be more valuable.
2) The IEEE 488 interface is now finalised and it provides remote control of all front panel operating modes, and complete control of delay in one sample increments rather than 6.25ms increments. Remote control of pitch ratio, and the ability to read and transmit to the controller the settings of all the above modes and controls. It will not contain the ability to read or modify the feedback level or in/out control.
3) The level indicator actually responds to the bit pattern on the A/D converter rather than input signal, and the level between clipping and 'normal' is fairly small, but will be increased on the production version. We suspect the reason the converter clipped before the overload, was because of misalignment of the DC offset pot. Because the indicator responds to audio level after processing, and because frequency sensitive processing does take place, the frequency dependence referred to in the review is unavoidable. We will try to adjust the various gains to assure that output clipping cannot take place without a front panel indication.
4) The feedback boost and cut filters have been redesigned in the production units. We believe the errors in the mid position to be caused by pots with incorrect tapers having been used in the prototype.
Richard Factor, Eventide Clockworks