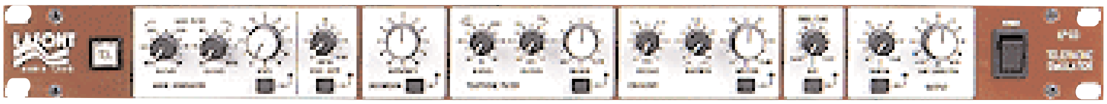
Lafont Audio Labs LP-23

Less mainstream but no less useful than compressors and EQs, the telephone simulator is an essential tool in many postproduction studios. **Rob James** dials up Lafont's LP-23

MANY AUDIO PROCESSORS, 'fairy dust' as some of us refer to them, have applications in sound for picture and radio drama - as well as music. Equally, a fair number are really only appropriate for music. Much rarer are processors with no aspirations towards music, but of these telephone simulators are a good example.



There have been off the shelf telephone simulators before the Lafont LP-23 but these have usually amounted to little more than band limiting filters. More complex designs have mostly been limited to one-off, custom built devices or programs for digital effects boxes which can be difficult and time consuming to set up. The LP-23 sets out to provide a single (mono) channel of processing to simulate the effects of telephony and various forms of radio transmission. To this end the unit offers a great deal more than simple low-pass and high-pass filters.

The LP-23 is constructed in the ubiquitous lU-high rackmount format. Connections are ridiculously simple, XLR connectors for signal in and out plus a balanced jack for a separate noise output, an IEC mains connector and no more. There are no MIDI, no time code and no computer connections of any sort. The front panel is finished in a restrained dark. claret with silver areas denoting the various processing blocks. Knobs come in two flavours, large with light grey caps and small, black all over. All the buttons except one are mechanically latching types with red LED. indicators adjacent. The remaining button is a square, internally-illuminated type labelled TEL which silently and instantly switches the unit in and out of circuit. The overall impression is of restrained and simple design.

There are two signal chains in the unit - a main audio chain supplemented by a noise chain. The processed noise may either be output separately or used in conjunction with the audio chain. In addition to the TEL switch each block of processing may be individually switched in and out of circuit

The first block in the audio chain is the Telephone Filter. This has the usual high-pass and low-pass filters - 35Hz-2kHz and 18kH-800Hz, respectively with an 18dB per octave slope. This section also contains a Balance control that sets the ratio between direct and filtered signal, Next in line is Distortion: which is controlled by one BALANCE knob. Turning this left or right of centre adds or subtracts a percentage of the input signal with a rich harmonic content. Importantly, this distortion appears to be independent of the main signal level. From here the signal passes to a Frequency FADE block. This simulates the fading associated with single side-band radios, superheterodyne frequency fading and other transmission effects that involve fading of signal. The fade rate is controlled by a VLFO (very low frequency oscillator). The rate is variable from 2s-15s. The last block in the audio only path is a single-band parametric equaliser (400Hz-7kHz) with a maximum boost of 166dB and variable Q (0.6-3). The noise chain is fed from a clock that drives a noise generator via a shaping filter to provide a pink noise output. A Level control varies this output from infinity to -7dBu. The noise chain has independent high-pass and low-pass filters with similar characteristics to the others apart from slope.

After this point the two chains are mixed. A Voice Over control decides whether the mixing will be constant or if the noise will duck when audio is present. The ducking is variable from 0 to -20dB. The combined signal goes through a Squelch section which is, in effect, a steep slope noise gate with a threshold control variable from 0 to -15dB. The final control covers Gain Correction and allows up to 10dB of boost or cut enabling the level of the effected signal to be matched to the direct.

Jean-Pierre Lafont has done his homework well. The choice of ingredients is nicely judged, balancing useful features with simplicity of use. The independent audio and noise filters are particularly useful. The LP-23 delivers the goods with a variety of convincing simulations of the degradation to be expected from telephone and radio transmission. Part of the key to success is under-standing that what is required is not a perfect recreation of what would actually happen to a signal, rather it is what the audience expects to hear. Simply sending a signal through a telephone or radio chain is often unconvincing, not to mention time consuming to set up. It might be thought memories and automation control would be desirable if not essential in a unit of this type. In fact, this would be to ignore the way in which most film mixing is done. The accent is on speed and ease of set up. A brilliant effect which takes half an hour to achieve and longer to automate is no good. A thoroughly convincing effect that can be set up in moments and recorded on a spare pre-mix track is just the job. And this is exactly what this unit provides. ■

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