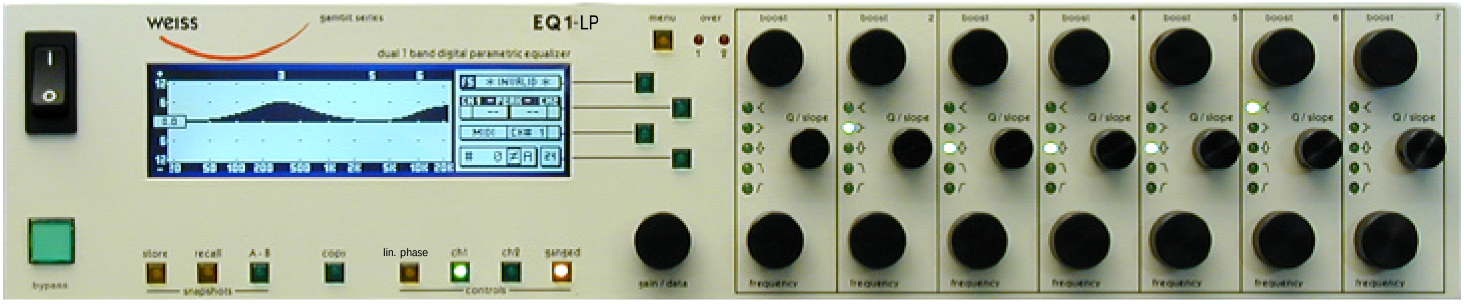
gambit series EQ1-LP

linear phase dual seven band digital parametric equalizer, 24Bit / 96kHz



The Weiss Gambit Series EQ1-LP

EQ1-LP: The EQ1-MK2 with linear phase response

The EQ1-LP has the same parameter set as the EQ1-MK2 and is therefore fully snapshot and MIDI control compatible. Sampling frequencies supported are 44.1, 48, 88.2 and 96kHz.
The impressive list of features of the EQ1-MK2 is now further extended with a linear phase response. This means that the delay induced by processing is now constant across the whole spectrum, unconstrained by eq settings - this is not the case with standard equalizers, where signal delay varies with frequency, with the length of the delay depending on the amplitude response.
The sound, or character, of an equalizer has been said to be influenced by phase response (for example John Watkinson: ...much of the audible difference between EQs comes down to the phase response., Studio Sound 9.97). The EQ1-LP is therefore the ideal tool for corrective amplitude adjustment, without the unwanted phase distortion added by standard equalizers.
The linear phase feature can be turned off, resulting in the exact same sound as the EQ1-MK2. So for creative sound designing, there is the benefit of two different equalizers in one machine.

For further information see the back of this brochure as well as the basic EQ1 and EQ1-MK2 leaflets.

"I think that the double-sampling EQ1-LP is the finest sounding equalizer that has ever been produced, whether analog or digital."

Alan Silverman
Arf! Digital

"You've never used an equalizer with this pure a sound... have to try it."

Bob Katz,
Digital Domain

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EQ1-LP

Background To The Linear-Phase Equalizer

There are two frequency dependent systems (filters): IIR (infinite impulse response) and FIR (finite impulse response). Both of these can be implemented either in the digital or the analogue domain, as discussed below.

IIR filters

Digital IIR filters can be derived from their analogue IIR counterpart, thus transforming the analogue filter parameters (Q, frequency, gain, etc.) into the digital domain. This is the procedure for digital EQ design favoured by Weiss Engineering and most manufacturers, because it maintains the well understood parameters from the analogue equalizers. IIR filters have certain properties:

- An infinite impulse response, hence the name (in practice, the impulse response will eventually decay below the noise floor).
- Some kind of phase response. Practical criteria dictate a so called minimum-phase response (which is linked to the amplitude response), though this not necessarily need to be so, and many other phase responses are possible (see below).

Linear phase IIR filters

Contrary to common belief, non-minimum phase IIR systems are realizable, this usually involves a minimum-phase section followed by a linear all-pass section with any arbitrary phase response. Thus, a linear-phase system would have a minimum-phase system followed by an all-pass system designed to have a phase response which, when added to the minimum-phase, will result in a linear phase response. This can (theoretically) be done in the analogue as well as the digital domain. However, there are certain practical problems which make this method difficult, if not impossible.

FIR filters

There is no direct way to transform analogue equalizer parameters to digital FIR systems, though there are algorithms that try to emulate a specific frequency response on a FIR system. These yield quite exact copies of the amplitude (and perhaps even the phase) response. FIR filters also have certain properties:

- A finite impulse response, resulting in a fixed length output when excited by an input pulse.
- Most design algorithms produce a linear phase response (just as IIR designs result in minimum phase respones).
- A delay.

So linear phase response is almost a by-product of FIR filters. But IIR filters are favoured over FIR filters in audio equalizers because of several reasons:

- The delay of the FIR systems is usually not acceptable for audio processing.
- The parameters of analogue equalizers are easier maintained in IIR than FIR filters.
- There is a considerably larger hardware expense involved for tuneable FIR audio equalizer than for the IIR equivalent.
- Digital signal processing for audio band FIR filters is computationally more intensive than for IIR filters.

FIR filters are used mainly for applications requiring extremely narrow transition bandwidths combined with no effects (i.e. phase shifts) in the pass band. Examples (in audio applications) are interpolation filters for sample rate conversion and band-split filters for crossovers.
In order to avoid the technical and commercial drawbacks of FIR systems for audio equalizers, the Weiss Gambit EQ1-LP uses yet another scheme for linear phase response based on the following property of IIR filters: if one processes a sample with any IIR system, then time-reverses this sample and processes it again with the same IIR system, one will effectively have cancelled out the phase response of the IIR system, while squaring the amplitude response. The solution lies in the time-reversed (non-causal) filter of this algorithm. So effectively, the EQ1-LP is a time machine, sending the audio signal backwards through time...
On a side note, this algorithm was experimentally implemented by Weiss Engineering in 1995, but only now is current DSP hardware powerful enough to realize the seven band 96kHz requirements of the EQ1-LP.