

# EAW UX3600 Specifications



## Features and Benefits

- Single operational mode
- Preset EAW Greybox™ signal processing delivers EAW Focusing™ to optimize performance
- Comprehensive, intuitive front panel control makes computer control unnecessary for most adjustments
- 10 parametric EQs per input channel and per output channel
- EAWPilot™ software allows comprehensive computer control
- Filter parameters are compatible with EAW Resolution modeling and alignment software for optimized performance

## Description

The UX3600 provides a complete suite of state-of-the-art digital signal processing tools. Advanced capabilities and features set the UX3600 apart from other processors in its class. This high end processor controls sound systems with a variety of output channels and also delivers optimized processing of specific EAW loudspeakers.

Built-in System functions include EQ filter, delay, gain, crossovers, and polarity. However, these functions are implemented using custom-designed algorithms to optimize their usefulness when applied to loudspeakers. For example, unlike many bell filter designs, the UX3600's produce a flat frequency response when reciprocal cut and boost filters are overlaid. Also, to more accurately calculate signal delay distances, a temperature setting permits compensation for sound speed differences.

Loudspeaker preset processing is available for specific EAW loudspeakers and loudspeaker arrays. The UX3600 implement EAW's innovative Greybox™ settings, including EAW Focusing™, as part of the preset processing to correct loudspeaker anomalies that cannot be corrected with conventional digital processing. Advanced limiting maintains sound quality while achieving maximum output levels based on amplifier outputs. An ambient humidity setting provides appropriate "air loss" equalization based on listening distances. While the output settings in Greyboxes are locked to deliver optimal loudspeaker performance, the user still retains control of input gain, EQ, signal delay, and polarity for the entire loudspeaker. This mode's simplicity makes the UX3600 practical for entry level users and fast to operate for experienced users, providing a high degree of system consistency while retaining all necessary, user adjustable, alignment controls.

The UX3600 uses standard USB protocol and cabling for computer control using the custom EAWPilot software. Up to 50 user-defined presets can be save into device non-volatile memory. Two year warranty.

## Compliance



FC Part 15 EN 60065:2002, EN 55103-1:1997, EN 55103-2:1997, EN55103-1, EN55103-2, EN60065

## DIGITAL SIGNAL PROCESSOR, 3 INPUT X 6 OUTPUT

See NOTES TABULAR DATA for details

### PERFORMANCE

Operating Range	15 Hz to 22 kHz, +/- 0.25 dB
THD + Noise	<0.005%, 20 Hz to 20 kHz, +10 dBu
Channel Separation	80 dB, 20 Hz to 20 kHz
Dynamic Range	>110 dB, A-weighted, analog in to analog out

### INPUTS (3X)

<b>Analog Mode</b>	Type	Electronically balanced
	Connector	3x XLR female
	Impedance	20k ohm (balanced), 10k ohm (unbalanced)
	CMR	50 dB 30 Hz to 20 kHz
	Crosstalk	-110 dB, 1 kHz
	Maximum Level	24 dBu, 12.3 V

### Analog to Digital Converters (3x)

Resolution/Sampling	24 bit, 48 kHz
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### OUTPUTS (6X)

<b>Analog</b>	Type	Electronically balanced
	Connector	6x XLR male
	Impedance	<50 ohm
	Crosstalk	-110 dB, 1 kHz
	Maximum Level	24 dBu, 12.3 V @>600 ohm load
	Absolute Min	>50 ohm/20nf

### Digital to Analog Converters (6x)

Resolution/Sampling	24 bit, 48 kHz
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### DIGITAL PROCESSING

Processor	50 Mflop, 32 bit, 48kHz Sharc
Latency	1.6 ms

### COMMUNICATION

Type	USB (1.1)
Control Software	EAWPilot™

### AC MAINS (nominal)

Connector	IEC C14
Maximum Input Range	100V to 240V 50Hz to 60Hz
Load	<50VA
Temperature Range	32° F to 104° F / 0° C to 40° C

### ORDERING DATA

Description	Part Number
EAW UX3600 Signal Processor	2039199

### Supplied Accessories

AC Mains Cable	120V (6 ft) and 220V (1 m)
USB Cable	6 ft / 1.8 m

### Optional Accessories

None	
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## PERFORMANCE DATA

See *NOTES GRAPHIC DATA* for details

### CHANNEL FUNCTIONS (Ch A to Ch C and Ch 1 to Ch 6)

See *NOTES TABULAR DATA* for details

#### EQ FILTERS (10 filters for each input and output channel)

##### Parametric

Type	Symmetrical boost / cut
Frequency	20 Hz to 20 kHz, 1/24 octave steps
Gain	+/-15 dB, 0.1 dB steps
Bandwidth	0.2 to 2 octave, 0.1 octave steps
Q	0.25 to 64, 0.1 octave step

##### Low / High Shelf

Slope	6 dB / 12 dB
Frequency	20 Hz to 20 kHz, 1/24 octave steps

##### Low / High Pass

Slope	6 dB / 12 dB per octave
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##### Bypass

Bypass Filter	For each individual filter
Bypass EQ	For all EQ filters

#### CROSSOVER (each output channel)

##### Low Pass / High Pass

Slopes	Butterworth, Bessel: 6 dB to 48 dB per octave, 6 dB steps
Slopes	Linkwitz-Reilly: 12 dB to 48 dB per octave, 12 dB steps
Frequency	20 Hz to 20 kHz, 1/24 octave steps
Bypass	For each individual low and high pass filter

#### OTHER CHANNEL FUNCTIONS

Input Delay	0.00 to 1200 ms, 20.83 us steps
Output Delay	0.00 ms to 1200 ms, 20.83 us steps
Gain	+/-15 dB
Polarity	Normal/Inverted
Mute	Mute/Unmuted
<b>Source Select</b>	
In A to In C	2x Analog (two signals are summed)
Out 1 to Out 6	In A to In C
<b>Limiters (Out 1 to Out 6)</b>	
Threshold	-40 dBu to 20 dBu in 0.1 dB steps
Ratio	1:1 to 20:1 and Inf:1, integer steps
Attack	40 us to 1 ms, 10 us steps / 1 ms to 40 ms, 1 ms steps
Release	10 ms to 3 s, 10 ms steps
Knee	Hard/Soft
Bypass	Each output limiter

## CONTROLS AND INDICATORS

### FRONT PANEL

#### Meters

Input (3x)	4 segment LED, Clip = 0 dBFS = full scale on ADC
Output (6x)	2 segment LED, Clip = 0 dBFS = full scale on DAC
	2 segment LED, LIM = Limiter threshold

#### Buttons

Inputs A to C	Selects input channel for editing
Output 1 to 6	Selects output channel for editing
Input Mutes (3x)	Mutes the output of the input channel
Output Mutes (6x)	Mutes the output of the output channel
EQ	Equalization
DELAY	Signal Delay
LEVEL	Level and Polarity
MENU	Channel setup
	Crossover (X-Over)
	Limiter (Comp / Lim Setup)
	User Programs (Program)
	Global Functions (Utilities)

#### Other

Data Entry Encoder	4 up/down/left/right buttons navigate system menu
Display	Backlit LCD 122 pixel x 32 pixel graphic
USB Port	Connector type B

### REAR PANEL

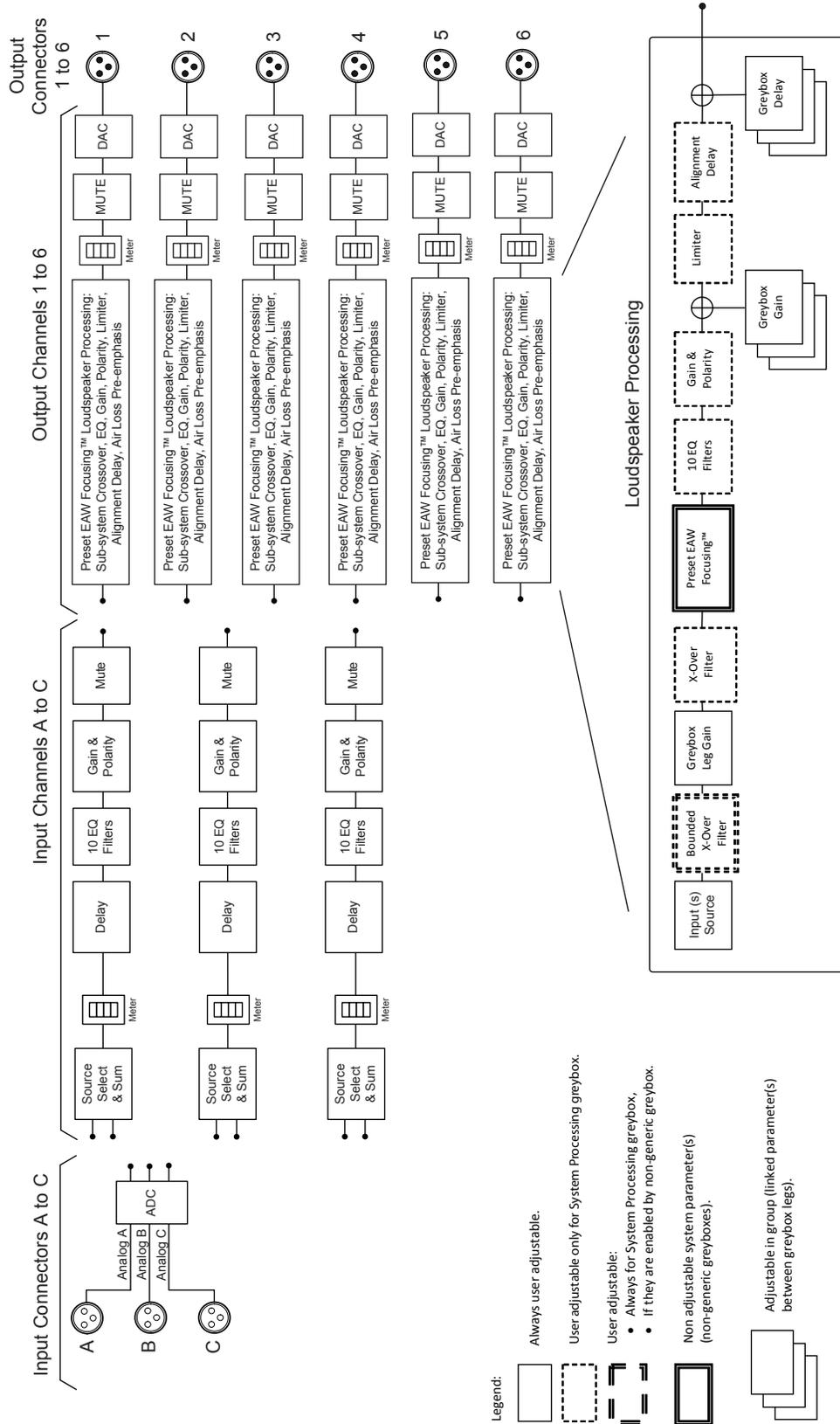
Power Switch	Turns AC mains on and off
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### GLOBAL FUNCTIONS

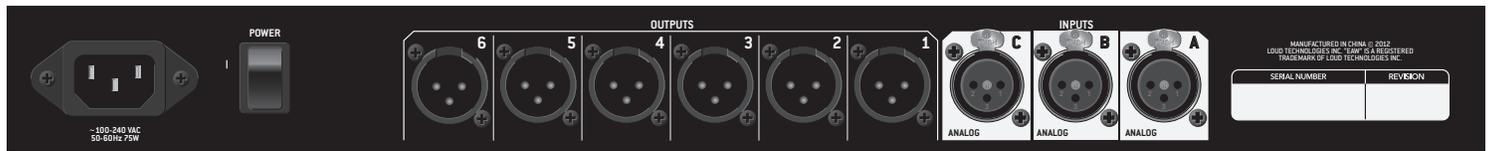
Units	Imperial/metric
Temperature	32 F to 114 F / 0 C to 40 C degrees, 1 F degree steps (used to convert delay time to distance)
Humidity	10% to 100% relative, 1% steps (functions only in loudspeaker optimization for the air loss pre-emphasis filter)
LCD Contrast	0 to 10 (relative scale)
Front Panel Lock	Password protects all functions except Mute buttons
Input Configuration	Analog
Programs	50 memories for user configuration
Memory Recall	<1 s, all parameters

## SIGNAL DIAGRAM (Loudspeaker Processor Mode)

### UX3600 BLOCK DIAGRAM



## REAR PANEL INPUTS / OUTPUTS



## UX3600 TECHNOLOGY

**UX3600 Design Challenges:** The goal for UX3600 digital signal processor was to provide a comprehensive set of digital processing tools in a 3 input by 6 output processor. In addition, the processor's hardware had to be able to implement EAW Focusing™ in order to provide preset, factory-optimized processing for specific EAW loudspeakers. This would include those intended for standalone use and those normally used in arrays. The preset processing allows unused input and output channels to have standard processing for other uses.

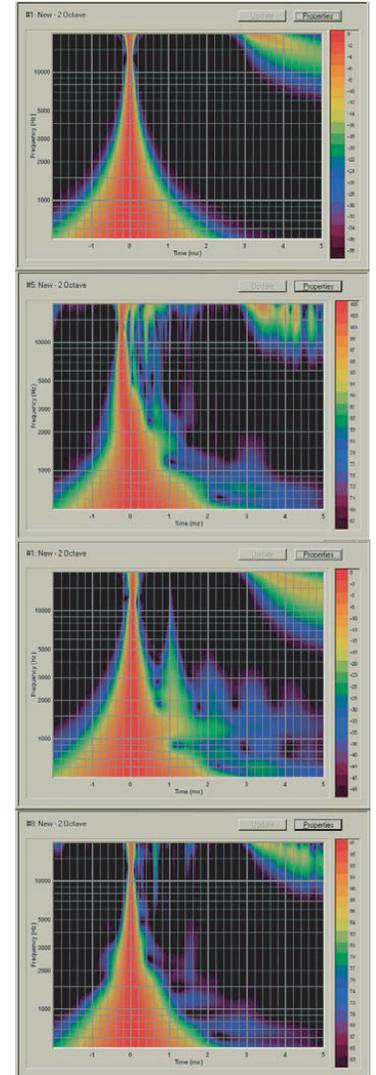
**EAW Focusing™:** Using innovative analysis tools and methods developed by EAW, specific, long-standing, loudspeaker problems were isolated and analyzed as to their solutions using DSP. However, the desired complex filter responses required accuracies grossly lacking in conventional DSP filters based on the Bilinear Z Transform (BZT). These filters sacrifice response accuracies in the upper audible octaves to avoid mathematical difficulties involving the Nyquist frequency. Using standard algorithms can result in filter response magnitude errors of over 15 dB, with equally flawed phase performance. Using FIR (Finite Impulse Response) filters would have resulted in latencies in excess of acceptability for real time use. EAW Focusing uses EAW-developed filter algorithms that avoid these issues while providing the exact, complex, filter responses required to correct the loudspeaker problems.

EAW Focusing is implemented in the UX3600 for many EAW loudspeaker models. Of particular note are EAW array loudspeakers. EAW Focusing not only optimizes the performance of the individual loudspeakers, but it also optimizes the integration of adjacent loudspeaker outputs and the off-axis performance of entire arrays.

**EAW Focusing Example:** Figure 1 shows the spectrogram of an ideal, point-source loudspeaker. Figures 2 and 3 show two different spectrograms of a 2-way loudspeaker optimized with conventional digital signal processing (DSP) and conventional measurements. In Figure 2 the time domain performance is emphasized. In Figure 3 the frequency domain performance is emphasized. In both cases there is significant energy to the right of the main energy spectrum compared to the ideal loudspeaker shown in Figure 1. These are all caused by inherent, mechanical properties of both the cone LF driver, the HF compression driver, and the HF horn itself. Although the frequency response (not shown) is nearly an ideal, flat line, these anomalies obviously exist in spite of the conventional processing. Because the usual measurements and corrective filtering lumped the undistorted signal and the anomalies together, the flat response is actually a combination of the energies from both. Anomalies like these are generally described as coloration and are responsible for why two, similar, flat-response loudspeakers can sound quite different.

In contrast, the result of applying EAW Focusing to this same loudspeaker is shown in Figure 4. The anomalies in both time and frequency are largely gone, making the spectrogram in Figure 4 look quite similar to that of the ideal loudspeaker in Figure 1. While the frequency response is also nearly an ideal, flat line, it is almost entirely a result of reproducing the energy from flat input signal.

**Summary:** EAW's engineering efforts resulted in the UX3600 digital signal processor which provides complete user control as well as factory-optimized, Guinness Focusing settings for EAW standalone and arrayed loudspeakers. Userfriendly, advanced, processing functions and plug-and-play audio/control networking facilitate its use for first-time DSP users, seasoned professional operators, system designers, and audio aficionados all over everywhere. In keeping with the performance of EAW loudspeakers, the UX3600's sonic performance is superb.



Figures 1 to 4 (top to bottom)

**SPECTROGRAMS:** EAW's proprietary spectrograms show the spectrum or frequency content of sound (vertical axis) and its variation in time (horizontal axis), the colors representing intensity. The width of the data reflects the size of the sliding time window applied to the data, which increases in size with lower frequency. The "data" in the upper right is simply a limitation of the spectrograph's mathematics and has no relevance.

## NOTES

### TABULAR DATA

1. Measurement/Data Processing Systems: Primary - FChart: proprietary EAW software; Audio Precision.
2. Measurements: Dual channel FFT; length: 32 768 samples; sample rate: 48 kHz; logarithmic sine wave sweep.
3. Measurement System Qualification (includes all uncertainties): Level: accuracy +/-0.05 dB 20 Hz to 20 kHz, precision +/-0.1 dB 20 Hz to 20 kHz, resolution 0.01 dB; Frequency: accuracy +/- 1 %, precision +/-0.1 Hz, resolution the larger of 1.5 Hz or 1/48 octave; Time: accuracy +/-10.4 μs, precision +/-0.5 μs, resolution 10.4 μs.
4. Volts/Amperes: Measured rms value of the signal or as noted.
5. Performance: Input, DSP (Digital Signal Processing), outputs, and ac mains characteristics.
6. Functions: Operating controls, function parameters, and indicators.

### GRAPHIC DATA

1. Graphs are plotted using raw data.
2. Frequency Response: Variation in output level with frequency for a constant input signal.
3. Phase Linearity: The difference in phase between the input signal and output, with signal processing latency removed.