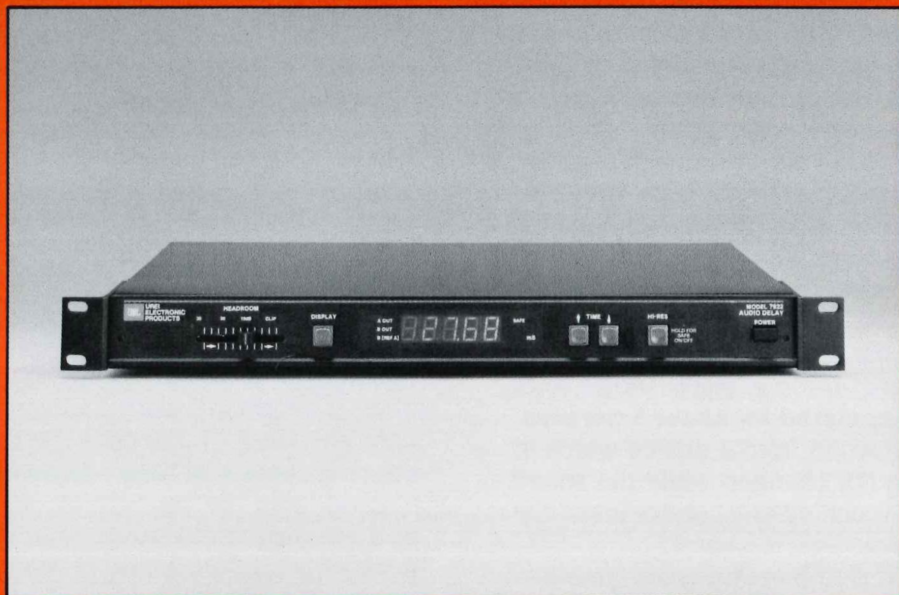
7922 AUDIO DELAY



FEATURES:

0 to 327 millisecond Audio Delay in 10 micro-second Steps; Equivalent to Approximately ⅛ inch Resolution for Over 350 Feet

One Input, two Independently Delayed Outputs

16 Bit Linear Conversion and Unique Low Noise Analog-to-Digital Converter for Greater Than 90 dB Dynamic Range

Linear Phase Anti-Aliasing Filter (±5°, 20 Hz to 20 kHz) for Superb Audio Quality and Sharp Imaging

Digital Oversampling Filter to Minimize Phase Distortion

Simplified Headroom Control and Indicator For Easily Optimized Signal-to-Noise Performance

B [Reference A] Delay Time Display and Reference Output For Precise Array Alignment (Compensates for Internal Propagation Time)

Non-Volatile Delay Time Memory; Unit Remembers Settings When Power is Off

Auto-Bypass Ensures Signal Flow if Power is Lost

The JBL/UREI Model 7922 Audio Delay is a high quality Digital Audio Delay with features that set it in a class by itself. Up to 327 milliseconds maximum delay is available on each of two independently adjustable outputs. This enables the 7922 to delay the sound to remote speakers in theatres, auditoriums, stadiums, large meeting rooms, or any large sound system where echoes must be avoided and image localization must remain at the main loudspeaker array. A significant improvement in sonic quality has been realized through advanced digital

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engineering, which includes the use of a linear phase anti-aliasing filter, 16-bit linear quantization, and digital oversampling. Equally important, this unit performs as well in the field as it does in the test lab, thanks to high-quality transformer-isolated outputs and JBL/UREI's unique single-control headroom adjustment system.

The 7922's very fine, 10 microsecond resolution further enables this Professional Digital Delay Device to be used for alignment of the acoustic centers of separate drivers within a single loudspeaker array. The precision and stability provided to perform this specialized function are also beneficial in remote loudspeaker delay applications.

The 7922 is contained in a handsome 1¾-inch high by 19 inch wide rack-mountable package. An LED Headroom display shows signal level as a function of dB below clipping level, while a 7 segment LED display indicates the precise delay time for either the A Output, the B Output, or the absolute difference between the two outputs (B [Reference A]). The displays automatically shut off in a 'sleep' mode when no control has been operated for about 5 minutes, thus minimizing distraction from a device which is set up once and then not changed while the sound system is operated. Touching any control instantly re-activates the display.

Before we committed to manufacturing a new, 'cutting edge' Audio Delay, we asked many leading acoustical consultants and sound system designers what features and performance criteria they considered essential. Higher resolution, a reference output, better sound quality, security protection, true balanced operation, and reliability ranked high on the list. Those comments — your requests — are now a reality in the JBL/UREI Model 7922.

REMOTE LOUDSPEAKER DELAY

If the audio sent to a rear 'fill' speaker array is delayed so that sound first arrives from the main array, then the listener will hear the program clearly and loudly with the second image localized up front. Sound travels at the rate of about 1.1 feet in 1 millisecond so the 7922's maximum 327 ms delay time works out to a distance of about 370 feet. Since the 'fill' speaker sound should arrive from 5 to 20 milliseconds after the main sound, the practical distance is more like 350 feet — which is adequate for sound systems in all but the largest of venues.

The 7922 has two independently adjustable outputs, so it can be used to delay sound to intermediate and more distant fill speakers. In normal mode, the delay is rapidly adjustable in 1 millisecond steps. In high resolution mode, very fine 10 microsecond steps permit the psychoacoustic effect to be fine tuned without having to move the rear speakers themselves. Expressed another way, the 7922 has 0.135 inch resolution, compared to about 14 inches in the typical Audio Delay. By bringing the arrival time of the speakers into perfect alignment, not only are gross echoes eliminated, but there can be a significant increase in perceived signal level to the rear audience listener, along with a major improvement in articulation and intelligibility. Here, the 7922's high resolution translates directly to a significant savings in sound system installation/setup time and money (just change a panel setting instead of moving the speakers).

ALIGNMENT OF DISSIMILAR DRIVERS WITHIN A SINGLE ARRAY

Any loudspeaker system which utilizes two or more types of drivers operating in different frequency bands will have a crossover region wherein different drivers reproduce the same signal. If the acoustic centers of these drivers are not precisely aligned, then the wavefronts they produce will arrive at some distant reference point at different times. Instead of constructively reinforcing one another, there will be varying degrees of destructive cancellation at different points on and off axis and at different frequencies. The resulting 'comb filter' or 'phasing' effect significantly degrades the sound. Obviously, the solution is to align the acoustic centers of the drivers. This 'obvious' solution is less obvious in its implementation.

Often, the drivers are physically 'locked' in place by constraints of the mounting, or, if the drivers can be moved, and one knows where the voice coils are located, the effective acoustic centers still may be incalculable. Physical manipulation of the drivers for the best sound in the prime listening area can be awkward, especially when the array is hung from a ceiling or other relatively inaccessible location. It is much easier to simply dial in a time offset with a suitable Audio Delay unit.

Most audio delay units are not suitable in this application because their resolution is inadequate. A unit with 1 millisecond resolution can, at best, offset the drivers in 13.5 inch increments. Units with 100 microsecond resolution yield 1.35 inch steps. Yet the required resolution to achieve a useful time correction should be within a few tenths of an inch. The 7922's 10 microsecond resolution corresponds to 0.135 inch (⅛ inch), and hence allows precise time corrections to be achieved. When this is done, the clarity and imaging improve significantly.

ALIGNMENT OF IDENTICAL DRIVERS ON DIFFERENT HORNS WITHIN A SINGLE ARRAY

What happens when different horns are used on the *same drivers* operating in the *same frequency range*? Consider a midrange array with short-throw horns for the near-field audience, and long-throw horns to cover the back of the audience. Both horns may be fitted with identical drivers, and, due to the physical requirements of the array and the need to avoid diffraction effects, the horn front edges may be aligned. This means the driver diaphragms will be at different distances from a point out in front of the horns where the wavefronts meet. Interaction in the overlapping coverage zone between the short and long throw horns will create a comb filter which again impairs sound quality. The answer is to excite the two sets of horns and drivers with separate amps, using the 7922's precision delay to align the wavefronts. The improved sound quality will be immediately clear to any listener in the zone of overlapping horn coverage.

LINEAR PHASE ANTI-ALIASING FILTER

In any digital audio delay, where the analog waveform is 'chopped' into thousands of samples per second and each sample is converted to a numerical value, there must be a steep input filter to restrict the processing of very high audio frequencies. Typically, for a 20 kHz bandwidth device, the sampling rate will be about 50 kHz. If input frequencies approaching 25 kHz enter the analog-to-digital converter and subsequent digital processing circuitry, they will create an unnatural distortion known as 'aliasing.' In aliasing, there is a critical input frequency above which the output will contain a 'reflected' frequency component that is at a lower frequency within the passband of the device. Even at low levels, these aliasing components are very noticeable, and hence must be avoided. One solution is to use a much higher sampling frequency, which requires a lot more digital signal processing power and more memory (which is very costly). The practical solution is to use very steep 'brick wall' anti-aliasing filters. These filters have slope rates of from 24 dB to 150 dB per octave, and thus prevent the higher freqency audio signals from getting into the digital processor. But, while even a good filter is cheaper than the higher sampling rate it avoids, the filter generally creates a new problem: severe phase shift in the audible passband.

Conventional anti-aliasing filters introduce up to 1,000° of phase non-linearity at 10 kHz. The resulting degradation of the audio quality, particularly the imaging, is perhaps one of the major reason why critical listeners have objected to digitized audio of any type. To counteract this problem, JBL/UREI engineers redesigned that anti-aliasing filter. It wasn't easy, but we created filters that cause no more than ±5° of deviation from non-linear phase in the audio passband from 20 Hz to 20 kHz. The audible improvement is noteworthy.

DIGITAL OVERSAMPLING

Following the digital-to-analog converter, the processor's 'clock' frequency, which is evident as small 'steps' in the waveform, must be removed by a filter similar to the input anti-aliasing filter. Normally, these 'reconstruction filters' also must be steep, in order to keep any of the 50 kHz clock from bleeding into the 20 kHz audio signal passband. JBL/UREI avoids the need for a steep reconstruction filter, and thus eliminates potential phase distortion, by performing special digital signal processing known as 'oversampling.'

The "oversampling" filter interpolates digital samples and produces an effective sample rate of 200 kHz. This digital signal processing uses a linear phase low pass filter, with special attention given to minimizing quantitative noise in the D/A process. The reconstruction filter required after each output's D/A converter has, due to the 200 kHz sampling frequency, a much more gentle slope rate which produces negligible phase distortion in the audio band.

LOW NOISE A/D CONVERSION

If not properly implemented, the analog-to-digital converter can be a source of distortion and noise. We devised a special circuit that loads the A/D converter's output with all zeroes (digital absolute silence) whenever the input level drops to below −90 dBu. This ensures the lowest possible noise.

INPUT AND OUTPUTS

The 7922 is equipped with barrier strips for permanent installations where maximum connector security and lowest long-term contact resistance are essential. It is also equipped with XL connectors for a secure, yet easily removed, low resistance interface. The electronically balanced input circuit exhibits the same gain whether an unbalanced or balanced source is connected. High quality, carefully matched components maintain a high Common Mode Rejection Ratio (CMRR).

SECURITY

Once the delay for a sound system has been set up, you don't want anyone to casually change the settings. For this reason, the 7922 is equipped with a simple front-panel 'SAFE' control that prevents inadvertent tampering. For still greater protection, all front panel controls can be guarded by an optional Model SC-6 plexiglas security cover.

HEADROOM CONTROL

Traditionally the adjustment of controls affecting headroom has either been a hit-or-miss proposition or one that required test equipment and time. No more. We have designed a single control that allows you to quickly and easily optimize the signal level through the Audio Delay for best signal-to-noise and headroom in *your* system. A peak reading LED display is calibrated in 10 dB steps from 30 dB to 0 dB (clipping) to give you an immediate visual indication of peak signal level and remaining headroom.

The circuit works like this: A variable gain amplifier (a VCA) is inserted at the input, before the A/D converter, and another pair of VCAs are installed after the D/A converter just ahead of each output. The Headroom control affects all three VCAs, thereby simultaneously increasing the input gain and decreasing the output gain. Because the output VCAs not only attenuate the signal from the delay section, but also attenuate the noise output from that section as well, the output VCAs further improve output noise performance.

PROFESSIONAL FEATURES & CONSTRUCTION

The 7922 Audio Delay is likely to be installed in a permanent, fixed sound system and touring sound systems, so we built it to the most stringent standards, utilizing a heavy-duty steel chassis, an extruded aluminum front panel, and rugged, removeable aluminum rack mounting ears. The rack mount ears can be placed in either of two positions, allowing flush mounting or mounting with the controls recessed behind the front of the rack (the security cover will then be flush with the rack front). There are no internal connectors to oxidize or get knocked loose in transit. In the event of power failure (or when the unit is shut off), a relay automatically bypasses the delay circuitry, connecting the input to the outputs via gold, bifurcated contacts. Upon power up, the relay doesn't engage until all transients have dissipated. The front panel graphics not only look good, they also won't rub off under hard usage beacuse they are printed on the back side of a protective polycarbonate overlay. Locking washers and/or chemical screw holding methods are used on all threaded fasteners.

Rear panel view.

ARCHITECTS AND ENGINEERS SPECIFICATIONS

The digital delay device shall be capable of delaying an audio input signal to each of two, independently adjustable outputs. The minimun output time setting shall be 150 microseconds, and the time delay at each output shall be capable of adjustment in 10 microsecond steps (in a high resolution mode) or 1 millisecond steps (in a normal mode) for any time from 150 microseconds up to a maximum delay of 327 milliseconds. In addition, a B [Reference A] difference mode shall be provided for delay settings starting at zero microseconds in 10 microsecond steps. The unit shall utilize 16 bit linear conversion with a 50 kHz sampling rate, and shall have a dynamic range of greater than 90 dB (15.7 kHz noise bandwidth, 600 ohm load). The input shall be electronically balanced, capable of accepting signal levels up to +22 dBu (0 dBu ref. 0.775 V RMS), and shall have an impedance of 20 k ohm with a balanced source or 10 k ohm with an unbalanced source). The gain shall be unity, ±1 dB, whether the input source is balanced or unbalanced. A single Headroom Control shall simultaneously adjust the input gain and the output gain to maintain optimum headroom and signal-to-noise performance throughout the internal digital processing circuitry with any nominal signal level from −20 dBu to +4 dBu, and a 4-segment LED display shall indicate the remaining headroom for any combination of input signal level and Headroom control setting. The outputs shall be balanced and floating, transformer isolated, capable to +22 dBm maximum output level (0 dBm ref. 1 mW) at less than 0.5% total harmonic distortion into a 600 ohm load. Frequency response shall be flat, ±1 dB, from 20 Hz to 20 kHz, and phase shift shall be less than ±5 degrees over this same passband. The digital delay device shall include a digital display that can be set to indicate the delay time of the 'A' output, the 'B' output, or the relative difference in delay time of the 'B' output referenced to the 'A' output. A pair of front panel 'Time Set' pushbuttons and a 'High Resolution' pushbutton shall permit the user to adjust the output delay for the 'A' or 'B' channel in normal (1 millisecond step) or high resolution (10 microsecond step) mode. A 'Safe' mode shall be available, wherein the time setting pushbuttons have no affect. An optional security cover also shall be available to protect front panel controls from tampering. The LED displays shall automatically turn off when no front panel controls have been operated for approximately 5 minutes, and the displays shall automatically re-activate when any control is subsequently operated. Input and output connections shall be made via rear panel XL connectors or barrier strips. The chassis shall be black painted steel, and the front panel fashioned of extruded aluminum with a polycarbonate overlay to protect control and indicator nomenclature. The unit shall meas-ure 44.5 x 483 mm (1¾ x 19 inches) with rack mount ears attached, and shall occupy one EIA rack space. The rack ears shall be adjustable so that front panel controls can either protrude or be flush in the rack. Depth behind the front panel shall be 324.6 mm (13¼ inches) with the rack ears flush, or 368.3 mm (14½ inches) with the rack ears forward. The unit shall operate from 105 to 125 VAC, 60 Hz mains and shall draw no more than 30 watts. The acceptable operating environment shall be from 0°C to +50°C, and storage environment shall be from −20°C to +60°C. The unit shall be a JBL / UREI Model 7922 Audio Delay.

SPECIFICATIONS

ELECTRICAL:	
Input:	Differential amplifier. May be used balanced or unbalanced, bridging. Impedance 20 k ohm balanced, 10 k ohm unbalanced.
Maximum Input Level:	+22 dBu.*
Output (Both Channels):	Floating, transformer isolated.
Output Level:	+22 dBm max. into 600 ohm load.
Frequency Response:	±1 dB, 20 Hz to 20 kHz.
Gain:	Unity, ±1 dB.
Dynamic Range:	Greater than 90 dB. (15.7 kHz noise bandwidth with 600 ohm load.)
Phase Shift:	Less than ±5 degrees, 20 Hz to 20 kHz.
Delay Range:	0 to 327 milliseconds.
Delay Resolution:	10 microseconds per step (High resolution mode). 1 millisecond per step (Low resolution mode).
Digital Conversion:	16 bit, linear.
Sampling Rate:	50 kHz.
Power Requirements:	105 to 125 VAC, 60 Hz, 30 W max.
Environment:	Operating, 0°C to +50°C. Storage, -20°C to +60°C.
INDICATORS:	
Headroom:	One red LED labeled 'Clip' and three green LEDs labeled '10,' '20,' '30' to indicate signal level below clipping.
Delay Time:	Three digit LED display in normal resolution mode. Five digit display in high resolution mode.
Display Modes:	A channel delay time. B channel delay time. B channel delay time referenced to A.
CONTROLS:	
Headroom:	Single knob adjusts internal input and output gain for best signal-to-noise ratio.
DISPLAY Pushbutton:	Steps through three display modes: A output delay time B output delay time B [reference A] delay time
TIME SET Pushbuttons:	Change time setting of either A or B output.
HI-RES Pushbutton:	Selects Hi-Resolution (10 microsecond) or normal resolution (1 millisecond) for display and control. Also, holding HI-RES button in for 5 seconds locks out the TIME SET buttons for security. Hold down HI-RES again to release security.
POWER:	Front-panel mounted ON/OFF switch.
CONNECTIONS (Rear Panel):	
Input:	Female XL and barrier strip.
Outputs:	One Male XL and barrier strip connection for each output (A & B).
Ground:	Audio ground to chassis ground removable strap
PHYSICAL:	
Front Panel:	22.3 x 483 mm (1¾ x 19 in) EIA rack mount.
Depth Behind Panel:	324.6 mm (13¼ in) with rack ears flush, 368.3 mm (14½ in) with rack ears forward.
Finish:	Aluminum extruded rack ears, Polycarbonate overlaid front panel, Chassis is black painted steel.
Net Weight:	3.9 kg (8.5 lb).
Shipping Weight:	4.5 kg (10 lb).
ACCESSORY:	SC-6 Security Cover

*0 dBu = 0.775 V RMS 0 dBm = 1 mW.

JBL / UREI continually engages in research related to product improvement. New materials, production methods and design refinements are introduced into existing products without notice as a routine expression of that philosophy. For this reason, JBL/UREI reserves the right to change any current product in some respect from its published description, but it will always equal or exceed the original design specifications unless otherwise stated.

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