Technical Notes Vol. 2, No. 3

Applications for the JBL/UREI 7922 Digital Audio Delay

Introduction:

Digital Delay devices have been around since the mid 1970's. Initially, their high cost and limited capabilities restricted the use of such devices to the highest quality professional sound reinforcement systems, where the signal fed to remote "fill" speakers was delayed in order that people at the rear of the audience would not hear an "echo" from the main, front speaker array, and would, instead, perceive sound to be localized at the front of the Auditorium. More than a decade later, this "remote speaker" application remains the primary one for digital delays but, as the available tools and sophistication of the industry have advanced , other delay applications have developed.

Certain audio delay devices have been designed solely or primarily to create special effects; such "effects" signal processors are outside the scope of this paper. The more advanced applications with which we are concerned include: time correction for non-equal arrival time of the acoustic wavefront from dissimilar drivers on either side of a frequency dividing network within a given loudspeaker system, and correction for non-equal arrival time of the wavefronts in the area of overlapping coverage from similar drivers which are mounted on dissimilar horns in a single array. The latter application, in particular, requires very fine delay time resolution as well as the effective elimination of propagation delays from the device's input to its output.

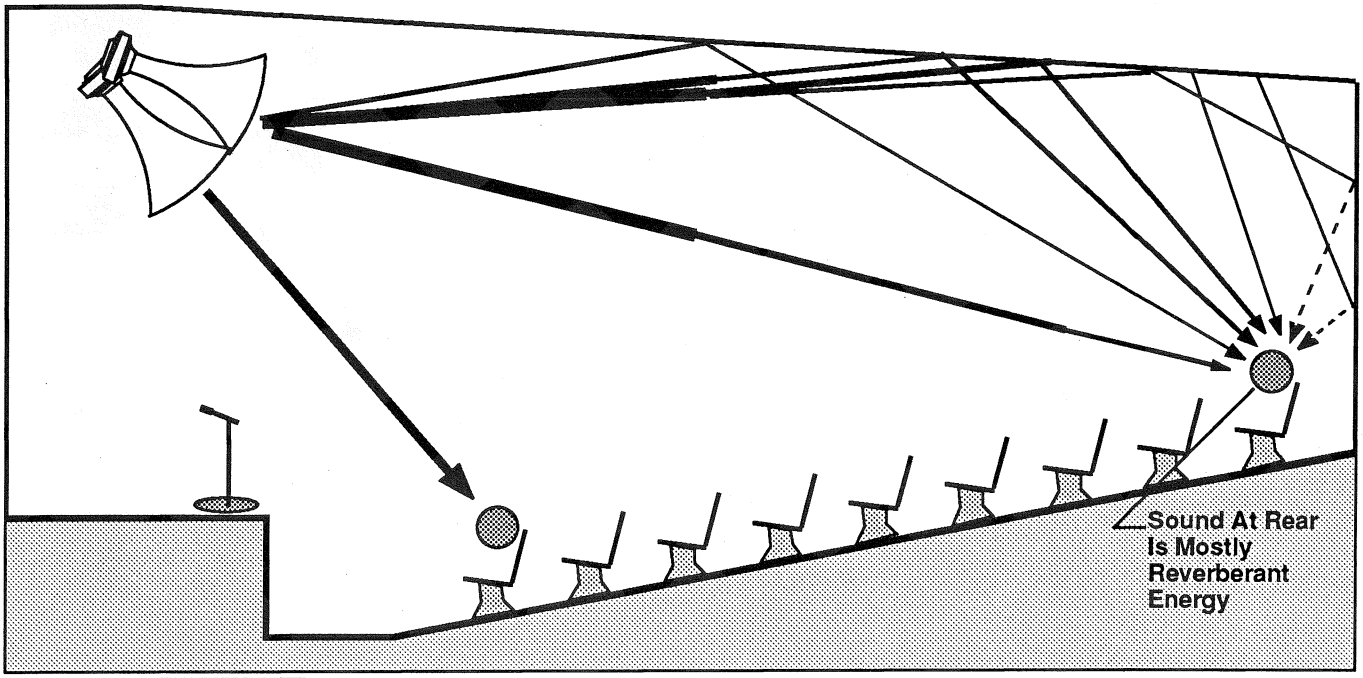
This technical note explains the use of the JBL/UREI 7922 Digital Audio Delay, and its salient features, as they pertain to the aforementioned applications.

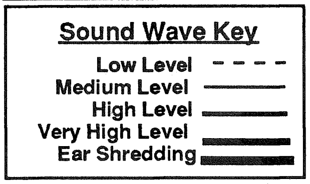
Why Time Delay Is Used on Remote "Fill" Speakers:

Let's examine why a typical sound reinforcement system for a theatre, auditorium or church will end up with one or more remote "fill" speaker, and why the fill speaker(s) benefits from time delay. We begin with a single primary loudspeaker system up front, the "main speaker array." Unfortunately, even though this array is mounted very high, near the ceiling, it is almost impossible to achieve the desired intelligibility at seats in the rear of the area without (a) causing deafness in the first 10 rows, and/or (b) causing very loud, but completely unintelligible noise in the rear of the area due to the highly reverberant nature of the sound field back there. Clearly, the best solution is to place another loudspeaker (or several) somewhere toward the rear of the audience (see Fig 1).

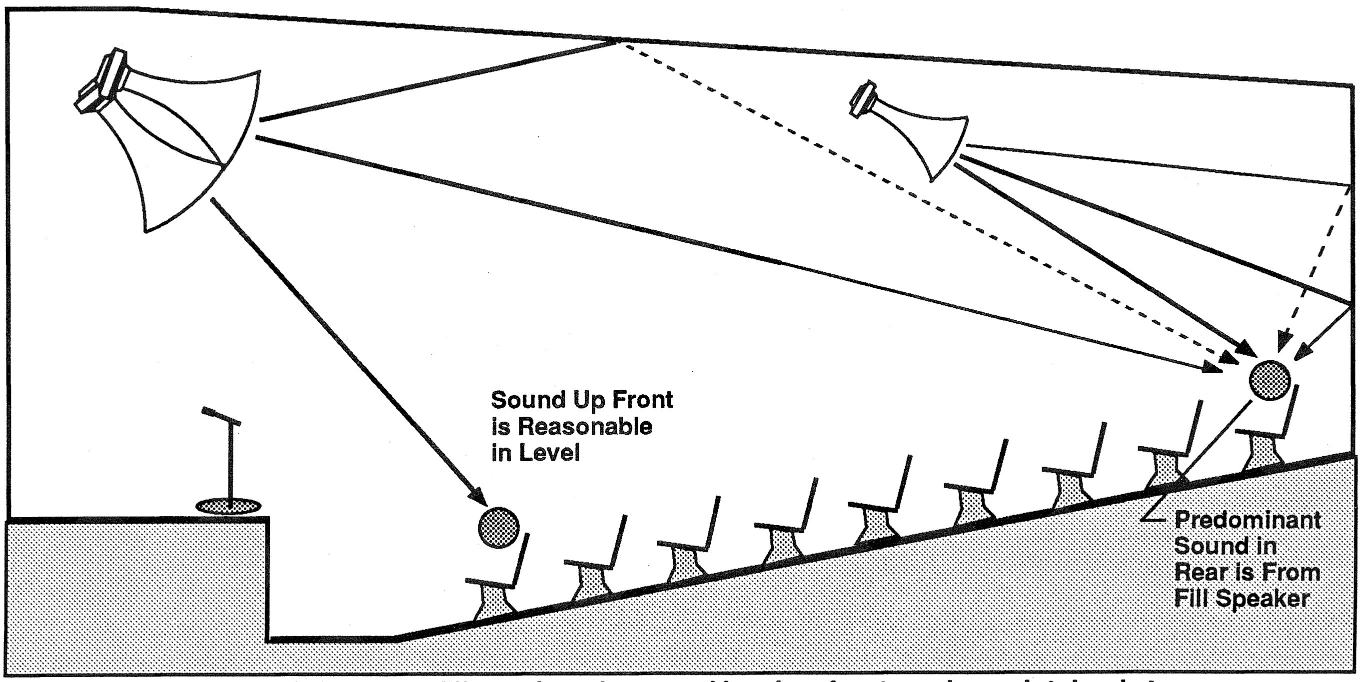
Unfortunately, when the rear speaker is simply installed as an extension of the main speaker, its sound arrives at those rear seats well in advance of the front speaker's sound. Also, since the fill speaker is much louder (due to proximity), the audience now hears a very confusing echo arriving from the main speaker after the sound arrives from the fill speaker. What's worse, they perceive the sound source to come not from the front of the venue, but from the rear fill speaker. The solution is to delay sound to the rear fill speaker.

The psychoacoustic result of delaying sound to the rear speaker, if done properly, is to "fuse" the sound into a single image which is localized up front. The rear fill speaker contributes the greater percentage of sound level to the rear audience's ears, yet they





A. With a single front cluster, as the sound level is increased to cover the rear seats, it becomes uncomfortabley loud up front, while reverberant energy prevents a significant improvement in Rear Area intelligibility.



B. With a rear fill speaker, the sound level up front can be maintained at a reasonable level, and sound at the rear can be increased to a level where intelligibility is improved without excess reverberation.

Figure 1. Why Rear Fill Speakers Are Used In Large Sound Systems

perceive the sound to come from the front. This precedence effect, described by Haas[1], is best realized when the sound from the rear fill speaker is delayed so that it arrives from 10 to 20 milliseconds after the sound from the main speaker system. Moreover, the sound from that rear fill speaker can be as much as twice as loud (10 dB) as the front speaker, yet the delay will create masking which prevents the listener from perceiving the rear speaker as a separate sound source.

1. Helmut Haas, 'The Influence of a Single Echo on the Audibility of Speech,' Journal of the Audio Engineering Society, March, 1972. Reprinted from doctoral dissertation submitted to the University of Gottingen, Germany, under the title, 'Uber den Einfluss des Einfachechos auf die Horsamkeit von Sprache' in December of 1949.

How To Align A Remote (Rear Fill) Speaker:

Basically, the process is very simple. Set up a sound system with a separate amplifier driving the rear fill speaker, and install a digital audio delay ahead of that rear amplifier. Then determine how much later the sound from the front "main" speaker arrives at some 'ideal ' (or at least average) listening position in comparison to sound arriving from the rear fill speaker, and dial in that amount of delay plus an additional 10 to 20 milliseconds. The trick is determining the correct delay setting. (See Figure 2.)

One can calculate the distance from speakers to a given seat using simple geometric manipulation, though in the 'real world' the speakers may not be installed precisely where they were specified to be installed on paper. Then, again, you could run a tape measure, but that is not necessarily practical, either. Besides, there is a very insidious factor that is seldom

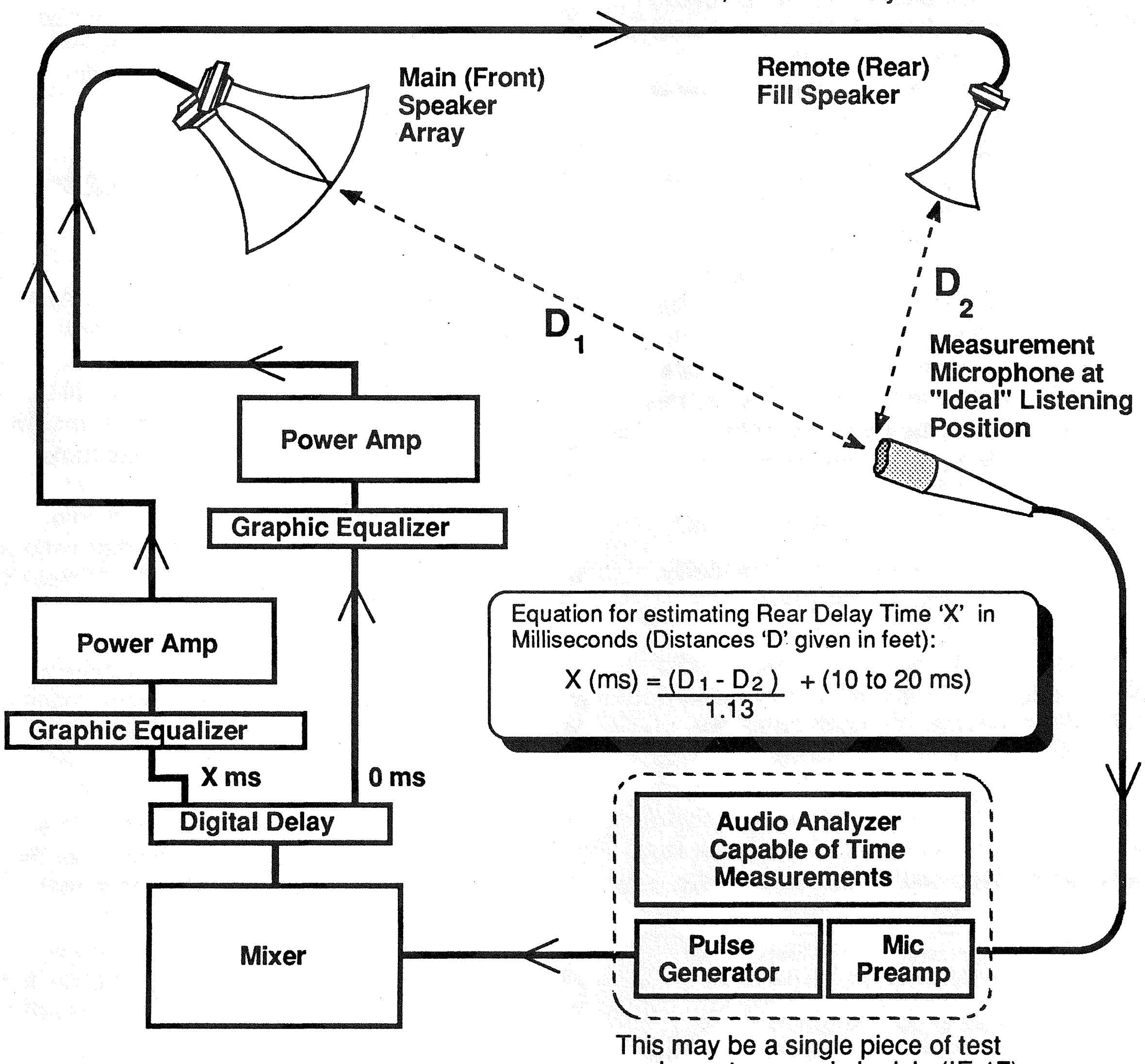


Figure 2. **Use of A Digital Audio Delay For Fusion of The Image in A Sound System With A Rear Fill Speaker**

discussed in the literature; the actual path which sound travels from speaker to listener is not usually a straight line. Due to temperature gradients indoors (as well as wind gradients out of doors), the sound may take some undefined curved path. Therefore, for optimum tuning, it is preferable to measure the difference in arrival times, using calculated times only as an initial guide to delay setup.

How can the difference in arrival time be measured? First, connect the sound system normally (with the Digital Audio Delay Bypassed or out of circuit). Verify correct polarity through the system. Adjust all, crossovers, equalizers, and amplifiers for the best possible sound. Then you're ready to make measurements and time adjustments.

Commercially built test equipment made by several companies is available, including the Techron TEF (Time/Energy/Frequency) analyzer from Crown, the Ivie IE-17 Sound Analyzer, and various dual-channel FFT (Fast Fourier Transform) analyzers.[2] With such equipment, it is possible to inject an impulse (a 'click') into the amplifier that drives the main front speaker(s) and read out the time interval between the moment the impulse is generated, and the moment it arrives at a measurement microphone at our ideal listening position (actually, you'll want to evaluate several positions before you learn which is typical). Then, with the time delay unit bypassed or physically removed from the circuit, inject an impulse into the amp that drives the rear fill speaker(s), and measure the time interval between impulse generation and arrival at the test mic. Simple subtraction gives you a value which you can use as the basis for initial setup of the Digital Audio Delay's output; add 10 to 20 milliseconds to this value, however. By the way, the measured time difference between the two uncorrected speakers should be something like .885 milliseconds per foot distance in the sound paths from the two speakers.[3]

Once you have dialed in the estimated time delay, it will be necessary to do some listening tests to determine the optimum delay. Initially, try using a repetitive clicking sound. You can use the output of a phase checker (though you're not using the measurement capability of that device). You can also have someone tap a drum stick on a wood block at a stage microphone. Just get a sharp leading edge waveform that you can listen to, and then experiment with slightly different delay settings until you hear one precise, loud 'click' instead of a smeared double click or a too-fat 'clock'. When the correct delay is achieved, the sound should appear to come from the main, front speaker, and there may be an increase in apparent loudness. Finally, play a variety of music through the system (unless it is strictly for speech reinforcement, in which case you'll want to 'talk' the system). Select music with a lot of transient 'attack' since a continuously bowed violin or a long organ note will give you no real means to evaluate the delay setting. The optimum delay time will almost certainly be from 5 to 30 milliseconds greater than the actual wavefront arrival time difference between the speakers, but practical experience shows that the range of 10 to 20 ms 'extra' is most common. When you've 'got it right' be sure to check a few other seats, and then make a note of the settings for ease of recovery in the event the equipment is later misused or removed for service.

2. TEF is a trademark of Techron (a division of Crown International), and Techron is a trademark of Crown, International.

3. The speed of sound in dry air, at sea level, at 59°F, 29.92 inches of mercury atmospheric pressure, is 1128 feet per second, or 1.13 feet per millisecond or 0.885 milliseconds per foot. However, sound travels more slowly in less dense air. This means that above-standard temperature, humidity or altitude, as well as below-standard atmospheric pressure, will slow down the sound and throw off time delay values based on simple calculations.

NOTE: Some people have suggested using an oscilloscope to visually identify the 'spikes' associated with arrival of the impulse sound from the main and rear fill speakers. However, this is a very uncertain method. It's difficult to get a sharp enough leading edge to see what you're doing. Based on our conversations with several acoustical consultants, we think the approach presented here is more realistic and practically achievable.

Details of the JBL/UREI 7922 in Rear Fill Applications:

The 7922 has two modes for delay time adjustment, high resolution and normal. In the normal mode, the delay time is adjustable in 1 millisecond increments, which correspond to a distance of about 1.13 feet. This should be sufficient for most rear fill applications, and the normal display mode makes it easier to slew through the available 327 ms range to the desired delay time. However, very fine increments of 10 microseconds are available in the high resolution mode, and you may wish to experiment with this mode for 'perfectionist' sound system setup.

The maximum delay time of 327 milliseconds corresponds to a distance (standard atmosphere) of 364 feet. However, if one allows some 20 milliseconds for additional delay to the rear fill speaker (for best Haas precedence effect), then the maximum distance differential between acoustic paths is about 346 feet. Bear in mind that this distance is *not the distance between the two sets of speakers*; rather it represents *the difference in acoustic paths between the speakers and the listener.* There are few installations where the rear speakers even come close to this maximum distance, so the 7922 should be useable almost everywhere.

The JBL/UREI 7922 has two independent delay outputs, and therefore it can be used to delay the sound to two different sets of rear fill speakers. For example, one rear fill may be in the middle of a venue, and the other under the balcony. In these cases, the procedure for setup is about the same; each rear fill channel is adjusted independently, against the front channel. Then the entire system should be turned on and checked again as the fill speaker delay(s) may require 'touch up' of the delay time in the event any transient smearing or echo develops due to overlap of delay zones.

Time Correction of Drivers on Either Side of A Crossover:

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 audio signal quality.

What if the drivers are brought into proper alignment so that the wavefronts from the high and low frequency drivers (in the crossover region) simultaneously arrive at a reference point in front of the speakers? Objectively, you can look at an oscilloscope, TEF analyzer, or the like, and see the dual-peaked display of an impulse merge into one peak. Subjectively, the sound becomes much more distinct. Imaging is significantly improved, and there may well be an apparent increase in sound level of several dB. Do you need to do this? Well, anyone who has aligned a speaker system in this manner will tell you they don't want to go back to listening to a non-time corrected system. The average listener can hear the difference when time correction is switched in and out, and can readily tell you which is the better sound.

Given that you want such a correction, how do you achieve it? The apparent solution is to align the acoustic centers of the drivers. This is easier said than done. Generally, the drivers are physically 'locked' in place by constraints of the mounting. If the drivers can be moved, physical manipulation may be awkward, especially when you're dealing with an array which is hung from a ceiling or other relatively inaccessible location. Even if you have an 'ideal' workbench situation, where you can physically offset the drivers with great accuracy, and you actually know where the voice coils are located, the effective acoustic centers still may be incalculable. Another factor... not all the time error is caused by the difference in acoustic centers between the drivers; there is 'phase shift' associated with every high pass and low pass filter, and typically that really consists of a combination of group delay and non-linear phase errors. At the crossover frequency, one filter may be leading by 90 degrees, and the other lagging by 90 degrees, so that the two drivers are 180 degrees out of phase. This amounts to a polarity reversal (though it is not), and some people will flip the polarity of one driver to correct this problem. Unfortunately, the polarity reversal only 'corrects' for one frequency and is incorrect for all other frequencies. Neither does it correct for the typical offset in acoustic centers of the drivers. It just confuses things, so what do you use as the basis for alignment? It is much easier to simply dial in an appropriate time offset to one of the two drivers with a suitable Audio Delay unit.

Most Audio Delay units are not suitable in this application because they do not have adequate resolution. Consider that a typical Audio Delay with 1 millisecond resolution can only be adjusted in increments corresponding to 13.5 inches (1.13 feet). Yet the necessary correction for proper alignment may be just fractions of an inch. The JBL/UREI 7922 can be adjusted in 10 microsecond increments, which corresponds to 0.135 inches (about 1/8 inch). This fine resolution is sufficient to obtain the necessary time correction for optimum speaker system performance.

(NOTE: We discuss a 2-way system here, but 3-way or larger systems can be handled similarly, correcting one transition at a time.)

For the actual alignment technique, refer to Figure 3. Before you attempt to make any time correction, you should do several things. First, set up the system as best you can without any time delay. Just get the sound as good as you can by setting up the relative level for the two sections and making any necessary crossover, graphic or parametric equalizer adjustments . Then measure the speaker system and determine from which of the drivers the sound arrives first; this is the driver which will have to be delayed.[4] You can make this determination by first pulsing the system with both drivers active and observing the analyzer. Then shut off the signal to one of the drivers, pulse the system again, and watch the analyzer to see which spike disappeared. Now install the time delay (or switch it out of bypass mode) ahead of the amplifier which powers that 'first arriving' driver.

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4. If you can measure the actual time offset at this point, be sure it is at least 390 microseconds. This is equal to the internal propagation delay of the JBL/UREI 7922 (caused by group delay in the A/D and D/A converters, and propagation delay in the digital circuitry), and corresponds to about a 5-1/4-inch offset. If there is less than 5-1/4-inches equivalent offset, then you won't be able to obtain a correction using the setup shown in Figure 3. Instead, try to move the 'later' arriving driver back just enough that its sound does arrive more than 150 µs after the other driver, or you may need to use the special setup illustrated by Figure 4.

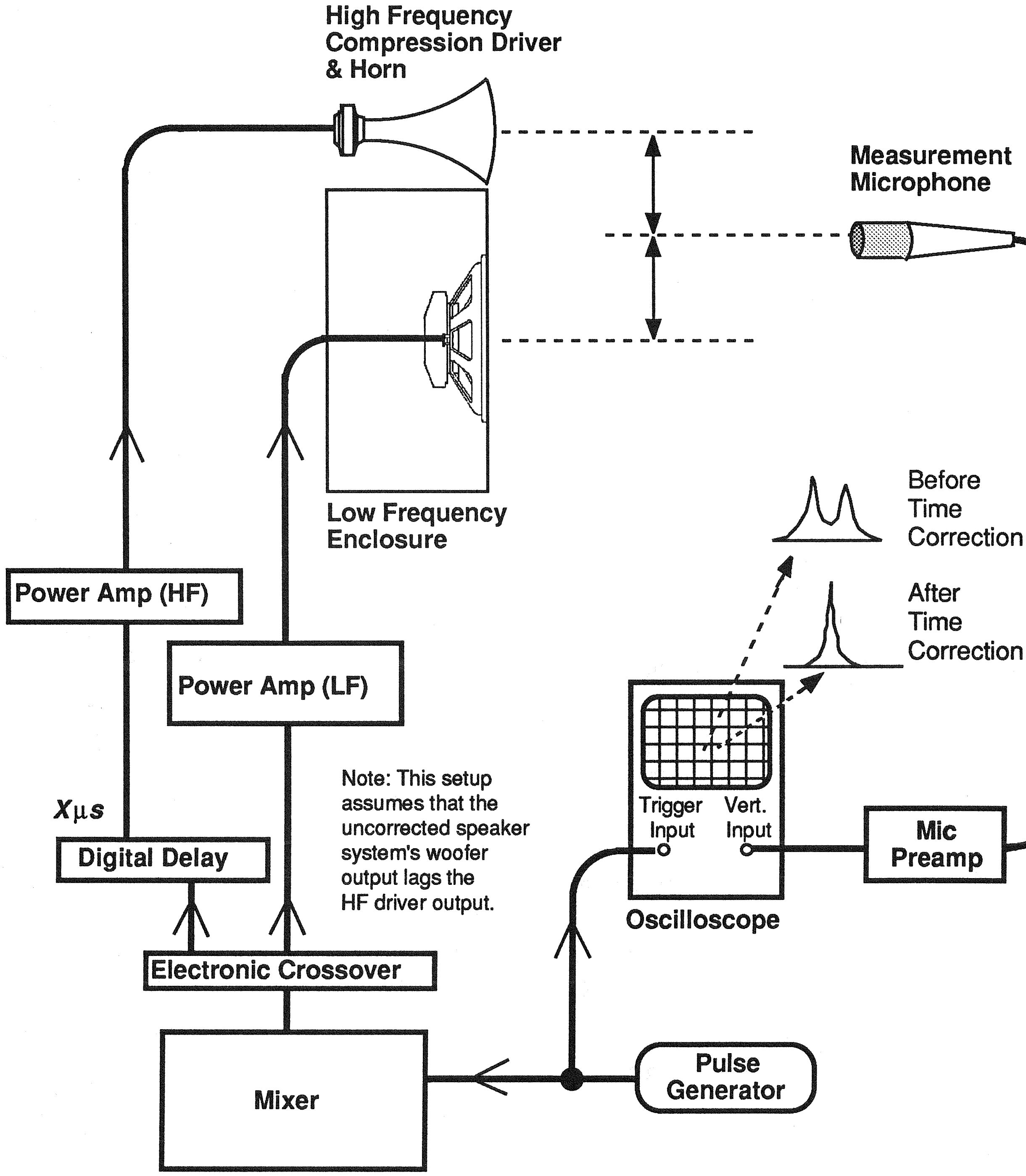


Figure 3. **Use of Digital Audio Delay For Time Correction of Drivers In A 2-Way Loudspeaker System With A Typical Electronic Crossover Network**

With the setup in Figure 3 (and Figure 4) we show an oscilloscope as the measurement tool. While we recommended against the scope in the rear fill speaker setup, this tool can be used here because the proximity to the sound source means that a sharply defined wavefront may be present. You can still substitute a TEF or FFT type analyzer if you have one. Assuming that the drivers for the two bands on either side of the crossover point are mounted in a single enclosure, you can place the measurement microphone about 4 feet in front of the enclosure, centered between the drivers. If you're dealing with an array of high frequency drivers, and a non co-located low frequency bin, then place the measurement mic at a sufficient distance out in front, where the sound fields from the drivers have merged... perhaps 10 feet or more. Now you're ready to begin the adjustment.

With the Digital Audio Delay in its high resolution mode, dial in delay while you watch to see the impulses come closer together on the scope or analyzer screen. As the correct fusion is achieved, you should hear a noticeable change in the sound. Now try playing music or talking into the system, and dial the

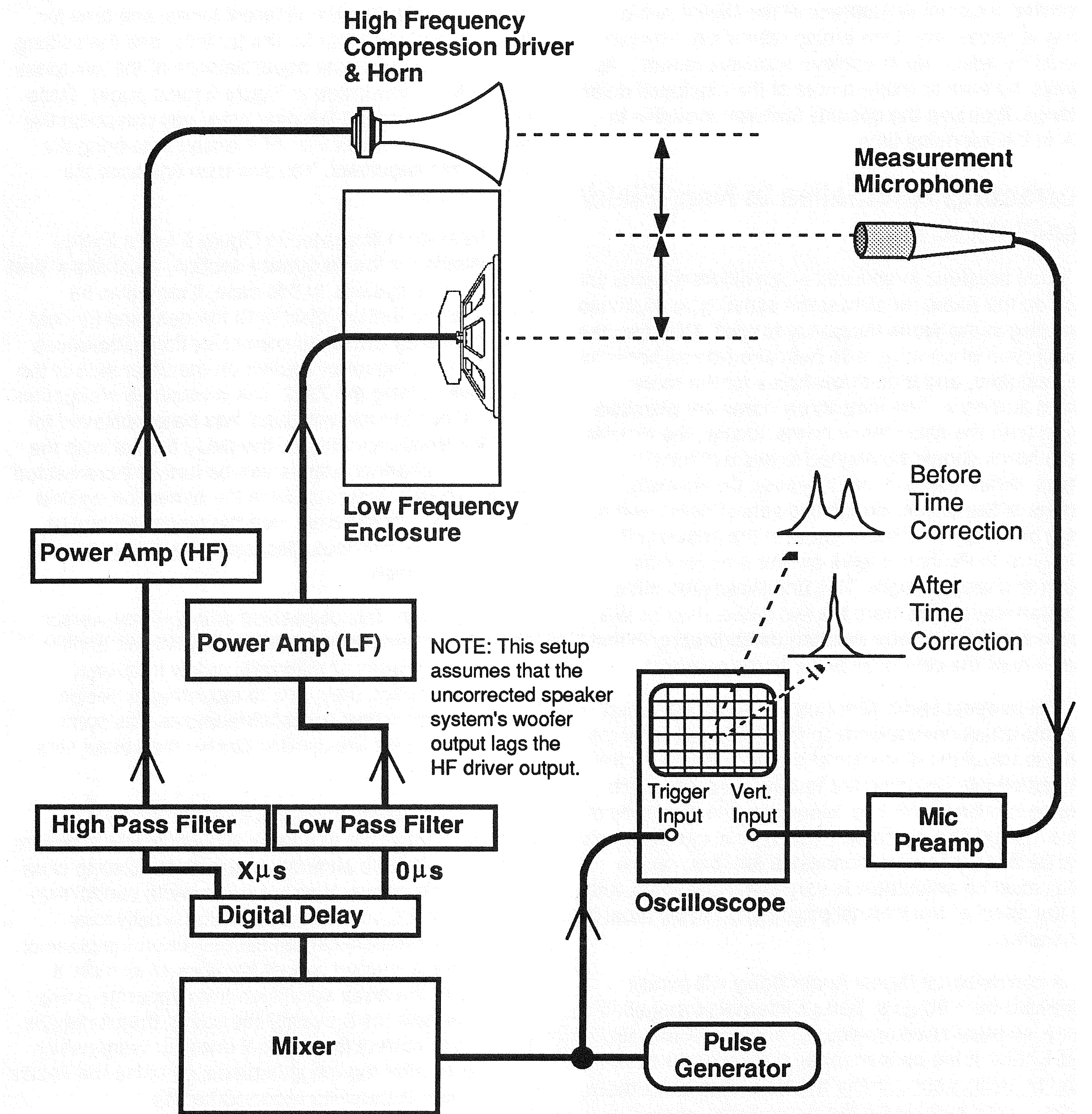


Figure 4. **Use of Digital Audio Delay With A Reference Output For Time Correction of Drivers In A 2-Way Loudspeaker System With Discrete High Pass and Low Pass Filters in The Crossover Network**

time offset back and forth by one or two increments to make sure the delay setting is optimum. This adjustment is really far less subjective, and more critical, than the adjustment of a rear fill speaker.

If you happen to have separate high and low pass filters, rather than a single input electronic crossover, then you can connect the 7922 Digital Audio Delay as shown in Figure 4. The benefit here is that one delay output can be set to zero delay, but it will actually have some 150 μs delay from the internal propagation time of the unit. The other delay output then can be used to dial in the precise delay, as in the setup in Figure 3. The difference, however, is that inter-driver time differences down to a 1/8 inch minimum can be corrected for in Figure 4, instead of the 2-inch minimum difference available with the setup illustrated in Figure 3.

Once you have corrected a system for time offset between drivers, you may find that the system equalization, and the relative levels, must be fine-tuned. Remember that any change in crossover filter settings or graphic equalizer settings may alter the group delay such that a further adjustment of the Digital Audio Delay is necessary. One tuning reiteration, however, should be adequate to achieve excellent results. As always, be sure to make a note of the displayed delay settings, then use the security features available to lock in the set delay time.

Correcting Anomalies in Near-Field/ Far-Field Arrays:

What happens in an array when different horns are used on the same (or almost the same) type of drivers operating in the same frequency range? Consider the typical case of an array with both short-throw horns for the near field, and long-throw horns for the more distant audience. The long-throw horns are physically longer than the short-throw horns. Ideally, the mouths of the horns should be aligned to avoid diffraction effects. When this is done, however, the acoustic centers of the drivers on the two sets of horns end up being offset by an amount equal to the path length difference in the horns (give or take a bit for differences in phasing plugs). This undesired time offset between wavefronts from the two types of horns will create significant, deep-notched comb filtering in that zone where the coverage of the horns overlaps.

The deepest comb filter notches will occur when the time offset corresponds to the time it takes for the wave to travel the diameter of the horn mouth. If the time offset can be corrected to near zero, the comb filtering can be minimized. Moreover, the directivity of the array may be improved. This type of correction can only be done when two conditions are met: (a) the delay must be adjustable in very fine increments, and (b) the effect of the internal propagation delay must be eliminated.

A conventional Digital Audio Delay will exhibit anywhere from 50 µs to 500 µs internal propagation delay, as mentioned previously, and so will the JBL/ UREI 7922. If the conventional, single-output device is used to delay sound to the drivers on the short-throw horns, it may provide far too much delay, even at a zero delay output setting, due to that propagation delay. The JBI/UREI 7922, however, can be set up so that one output (the 'A' channel) is at zero delay, and the other output (the 'B' channel) is at as little as 10 s. When the display is set for 'B [Ref A]' mode, the display will show the actual difference in delay between the two outputs. This effectively eliminates the propagation delay and permits correction for distances of a little as 1/8 inch.

The only 'trick' in aligning this type of array is that the sound must be measured in situ, with the measurement mic (and a good reference set of human ears) located in the middle of the zone of overlapping coverage between short and long throw horns. Pinpointing this zone may take some experimentation with turning on and off one set of drivers and then the other... or injecting two different tones, one tone for the near and the other for the far field, and then sitting where you hear a near equal balance of the two tones. The setup is illustrated in Figure 5 (next page). Once you're in the correct listening area, you can pulse the system and use a TEF or FFT analyzer to bring the pulses into alignment. You can then fine-tune the results by ear.

The system illustrated in Figure 5 has a further complication, a low frequency section... just like a 'real world' sound system. In this case, it may also be necessary to further offset both the near and far field high frequency drivers to correct for time differences with the low frequency section on the other side of the crossover. Using the 7922, this is relatively straightforward. Once the correct 'delta' has been achieved for the high frequency drivers, the delay time of both the 'A' and 'B' channel outputs can be further incremented (by an equal value) to provide the correction against the low end. In this case, use the basic techniques outlined in the previous discussion for 2-way speaker system alignment.

Incidentally, this discussion of long-throw versus short-throw horns on high frequency drivers applies equally to an array of midrange or low frequency enclosures which may, due to mounting or design, have wavefront time arrival differences. The same basic techniques are used to correct the sound here, as well.

One final variation on this theme involves the correction for errors in a large array of identical drivers and horns. If such an array is not constructed to close physical tolerances, then the beamwidth control may not be as planned, and/or the sound quality may suffer. The problem can be caused when the plane of the drivers is warped by just fractions of an inch. If portions of the array are driven from separate power amp channels (as is usually the case), then it may be possible to correct for physical errors or warp (which may occur after mounting in place) by using the 7922's two outputs in the delta mode (B [ref A]).

Obstacles To Sound Quality in A Digital Audio Delay:

To pass excellent, not merely acceptable, audio, a Digital Audio Delay must have several characteristics, including: wide bandwidth, low phase shift, low noise, and low distortion.

Wide bandwidth in a digital audio device is expensive because the sampling rate must be more than twice the highest frequency in the passband. The lowest sampling frequency for a 20 kHz device is 40 kHz, and the 7922 uses 50 kHz to achieve a 20 kHz passband. However, it is essential that no energy be

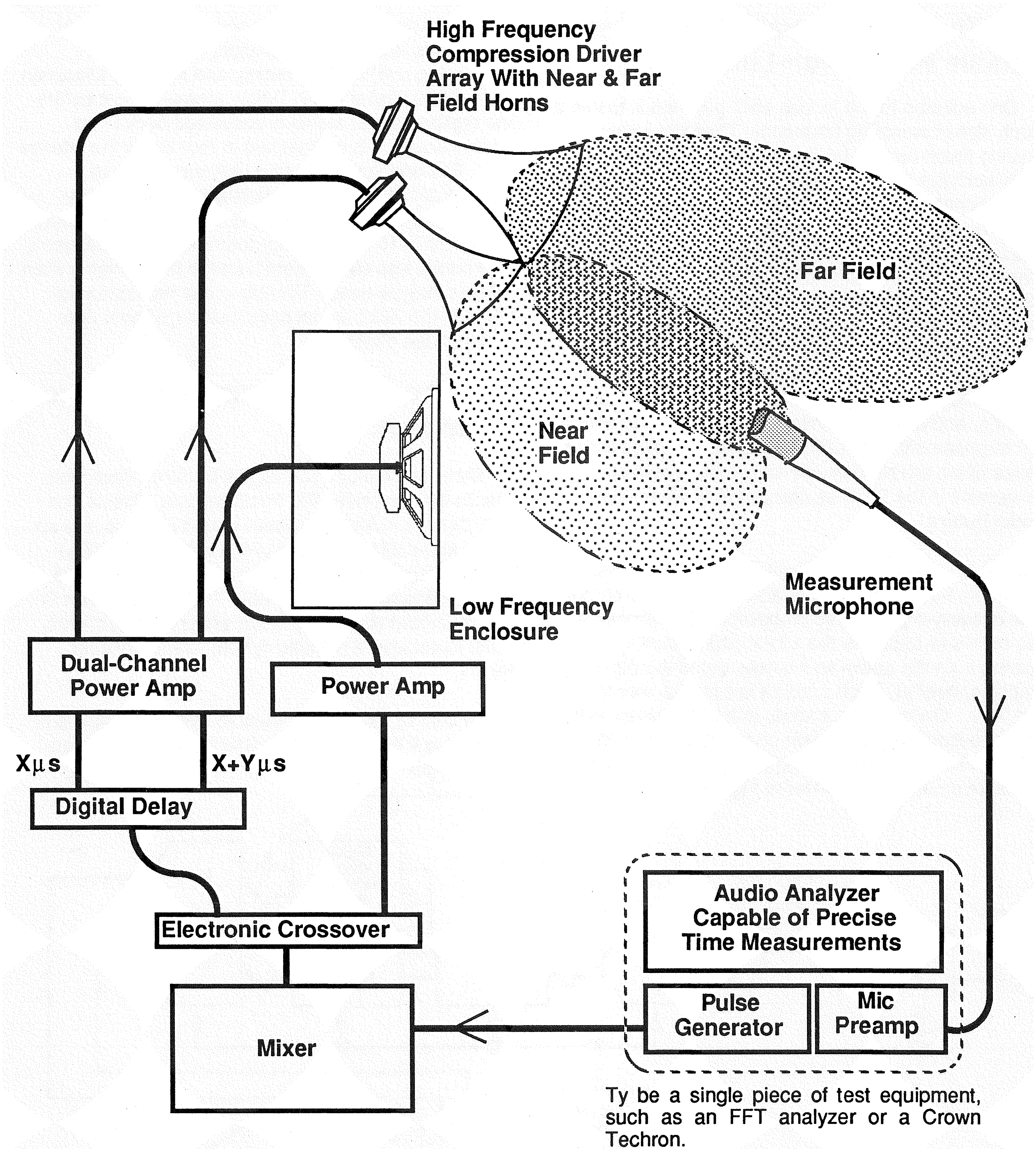


Figure 5. **TIme Correction of An Array With Identical Drivers on Different Length Horns**

present above half that sampling frequency, or an unpleasant phenomenon known as 'aliasing' will occur. Aliasing is a mirror-like reflection of higher frequency energy down into lower frequency ranges where it becomes audible. To prevent aliasing, very steep filters are used to define the upper frequency limit of the digital audio device. These filters have slopes of from 60 dB per octave to as much as 150 dB per octave. And, typically, they can create as much as 1,000 degrees of phase non-linearity at 10 kHz! Thus, even as the filter avoids aliasing, the resultant severe phase distortion causes noticeable degradation of the audio signal quality.

How The 7922 Virtually Eliminates Audible Phase Non-Linearity:

One solution to the 'phase shift' problem is to use a much higher sampling frequency. Then the anti-aliasing filters can be (a) less steep, and (b) further away from the audio passband so that they cause less audible phase non-linearity. Unfortunately, higher sampling frequencies require much more elaborate analog-to-digital converters, and much more digital processing and memory, all of which cost a lot more money. It is simply not cost effective or economically feasible to offer such devices for commercial use. Instead, JBL/UREI redesigned the anti-aliasing filter. We devised a linear phase filter that passes 20 kHz with no more than 1 dB of attenuation, yet prevents aliasing, and does this while introducing no more than ±5° of phase shift from 20 Hz to 20 kHz. These linear phase filters are responsible for a major sonic improvement in the 7922 as compared to other Digital Audio Delays.

The anti-aliasing filter is but one source of phase distortion, however. There is also a filter at the output side of every digital audio processor. The purpose of this filter is to separate the 'clock' (the sampling pulses) from the audio as it comes out of the digital-to-analog converter. Such a circuit is called a 'reconstruction filter.' Once again, it tends to be very steep, with a lot of associated non-linear phase distortion. JBL/UREI took a different approach here, and it's called 'digital oversampling.' (Refer to Figure 6.)

The basic input sampling rate of 50 kHz is what determines A/D converter design, CPU (central processing unit) design, timing, and memory allocation in the 7922 Digital Audio Delay. However, just before the digitized audio signal is converted back to an analog signal, it is re-sampled at four times the internal rate. This enables the output D/A converters to operate at 200 kHz. The resulting clock frequency component which must be removed by the reconstruction filters is 10 times higher than the audio passband, and hence less steep filters tuned to much higher than 20 kHz may be used. Therefore, the reconstruction filters in the 7922 do not contribute significant non-linear phase distortion.

How the 7922 Minimizes Noise and Distortion:

Many so-called '16 bit' digital audio devices yield results that are more like 14 bit devices. That is, the quantization noise and overall dynamic range end up being about 85 dB instead of the better than 90 dB which should be available with a properly designed and implemented 16 bit device. The high quality A/D converter and true 16-bit linear encoding enable the 7922 to achieve a realistic dynamic range of over 90 dB.

Of course, 90 dB of available dynamic range is of little value if the operator incorrectly sets the system

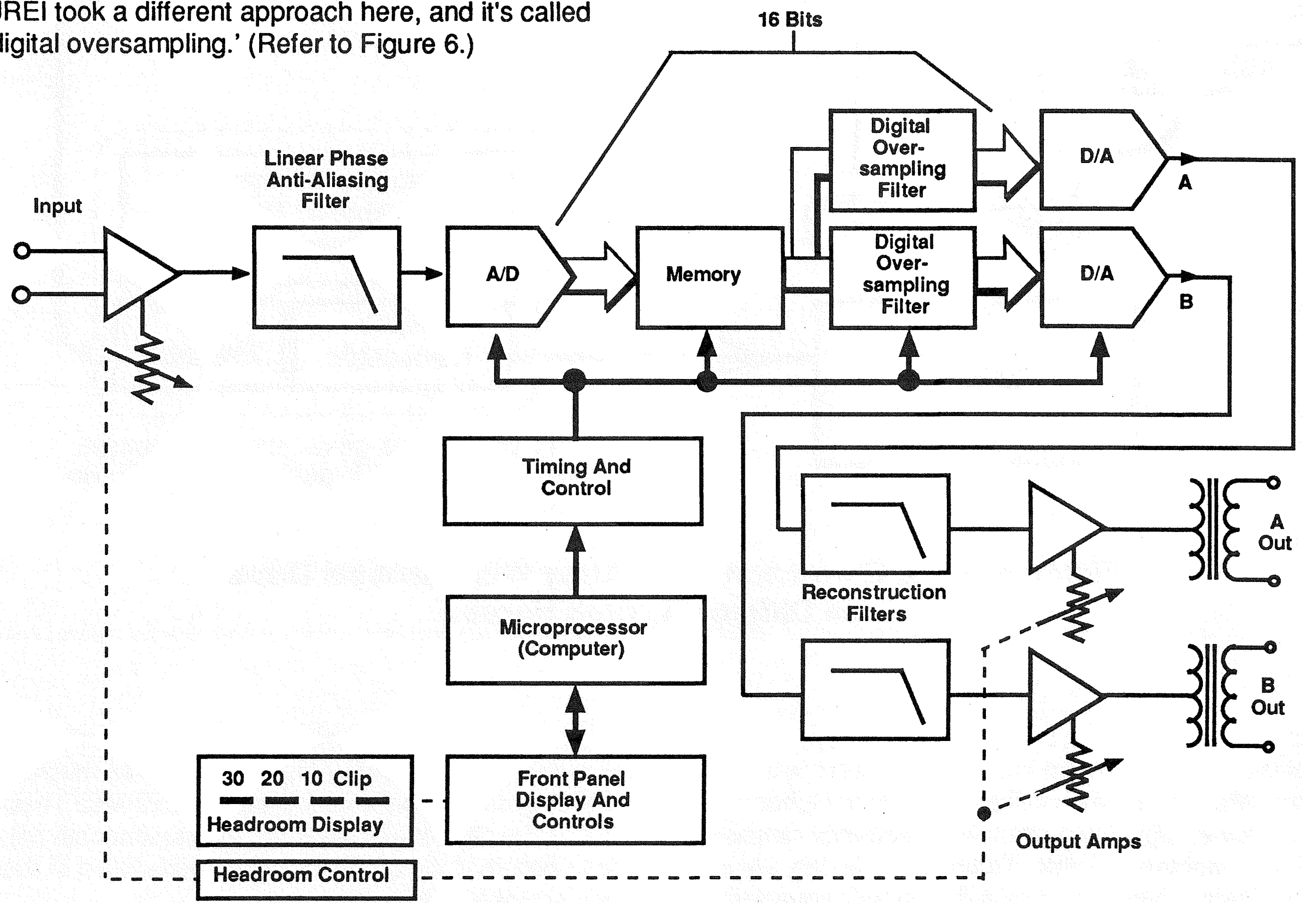


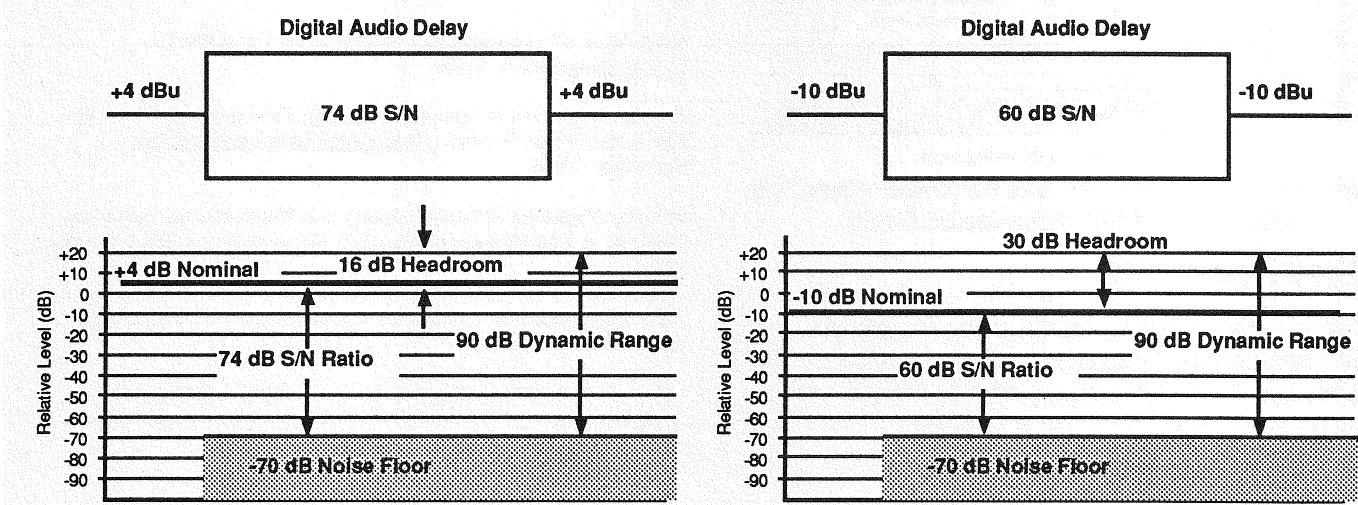
Figure 6. Block Diagram of JBL/UREI 7922

levels. Proper gain structure is essential if one is to realize optimum signal-to-noise performance and lowest distortion.

The difference between the program level and the maximum undistorted output is called headroom and will vary in different systems because of program material variations. For example, a large amount of headroom is normally required in a live recording studio situation where program level may change rapidly over a short period of time, but where compression or limiting is not yet appropriate (or even desirable). Conversely, an audio system which plays primarily pre-recorded program material with much more tightly controlled dynamic range may operate well with reduced headroom. In many systems it is possible to determine the amount of headroom necessary for good operation. If, in such a situation, it were possible to optimize the signal level for each piece of equipment in the audio chain, it would be possible to make a trade-off of any excess headroom for improved signal-to-noise ratio. Unfortunately, this is not always easy to do. Set the input level too low, and the signal will be unnecessarily close to the internal noise floor; consequently, turning up the output level may provide the correct nominal operating level, but that extra gain will also amplify the internal noise. Set the input level too high, and you will reduce the headroom sufficiently to create clipping distortion or overload the A/D converter during program peaks; when you turn down the output gain to obtain the proper nominal operating level, that distortion will still be present. A Digital Audio Delay in a sound system should be operated at as high a signal level as possible to keep the signal-to-noise ratio high, but with 10-20 dB of headroom to account for the crest factor in the program material. The JBL/UREI 7922 is equipped with a novel and extremely convenient method for accomplishing headroom adjustment.

JBL/UREI engineers 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.

To understand how this works refer to Figure 7 which shows the noise/headroom performance of a typical Digital Audio Delay with -70 dBm output noise and maximum output of +20 dBm (the 7922 is capable of +22 dBm output, but we've rounded numbers so the following explanation is easier to grasp). In Figure 7A the Delay is being driven by a signal level of +4 dBu (Ref. 0 dBu = 0.775 V). The signal-to-noise is 74 dB (70 + 4=74) and the headroom is 16 dB (+20-{+4}=16). In Figure 7B the same Delay is being driven by a signal level of -10 dBu. The signal-to-noise has degraded by 14 dB to 60 dB (70+{-10}=60) and the headroom has increased by 14 dB to 30 dB (+20 -{-10}=30). Occasionally 30 dB of headroom is appropriate but in many situations it is excessive and we would prefer to trade off excess headroom for better noise performance. Figure 8 shows what happens in the JBL/UREI headroom circuit for different levels. Note that variable gain amplifiers (actually Voltage Controlled Amplifiers) are inserted before and after the Delay. In the case of a +4 dBu input signal, the gains of the three VCAs are set to unity and as shown in Figure 8A. The noise and headroom numbers are unchanged from the example in Figure 7A. In the case of the -10 dBu signal however, the gain of the input VCA is raised by 14 dB and the gains of the output VCAs are reduced by the same amount. The signal level actually seen by the delay processing section has now increased back to the +4 dBu level and the signal to noise and headroom numbers at the output of the Delay have been restored to their previous values as in Figure 8A. The decreased gain (relative attenuation) in the output VCAs then returns the signal to the original -10 dBu level.



A. At +4 dBu Nominal Operating Level **B. At -10 dBu Nominal Operating Level**

***Figure 7.* Headroom and S/N Performance in a Typical Digital Audio Delay.**

Now, of course, all of this gain adjustment could be performed external to the Digital Audio Delay itself, but that would require additional amplifiers, pads and wiring with the attendant increase in circuit complexity and possible reduction in system reliability. It is much more convenient and cost-effective to include the facility right in the unit. A single front panel linear slide pot serves as gain control for the one input and two output stages with unity to +20 dB of gain and unity to -20 dB of attenuation, respectively. As the control is moved, the gain through the Delay does not apparently change to the outside world.

Traditionally the adjustment of controls affecting headroom has either been a hit-or-miss proposition or one that required test equipment and time. Neither method was optimum. The 7922's single control and readout facilitate precise headroom adjustment in less time than it takes to explain or read.

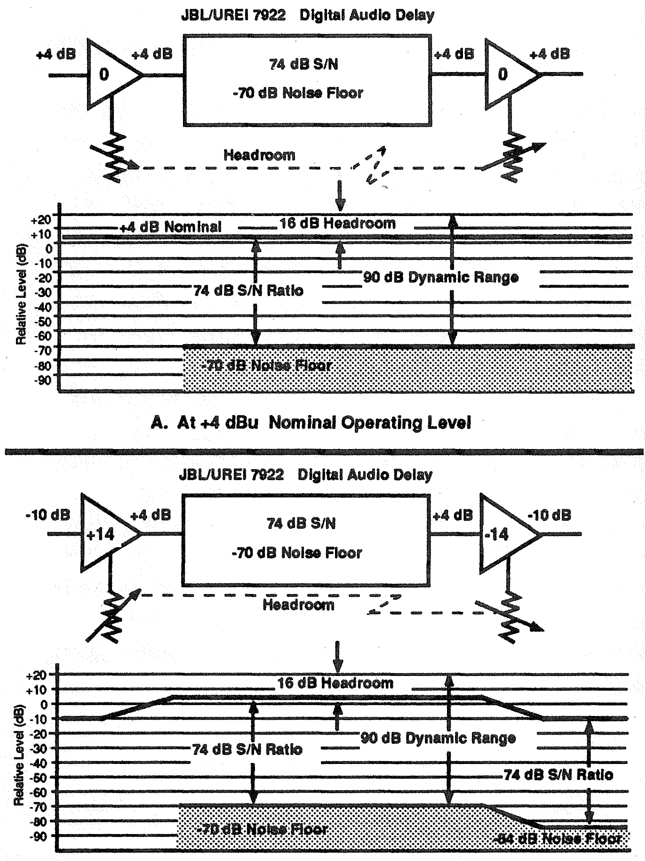


Figure 8. **Headroom and S/N Performance in The JBL/UREI 7922 Digital Audio Delay.**

Other Features:

After carefully setting up the delay values, you don't want anyone to change them unintentionally. For this reason, the 7922 is equipped with a 'Safe' mode which electronically locks out the time adjustment pushbuttons. A plexiglass security cover is available as an option to preclude any front panel adjustments.

The input circuitry to the 7922 is electronically balanced, and the circuit is designed so that the gain does not change whether the input source is balanced or unbalanced. The output circuitry includes high quality transformers, designed and built by UREI, which establish floating outputs for complete grounding isolation.

An LED digital display can be set to display the delay time for the 'A' output, the 'B' output, or the difference between the two ('B [Ref. A]'). With the unit set to normal resolution mode, the time adjustment pushbuttons will increment the delay time, and display will show the time, in milliseconds from 0 up to 327 maximum. In high resolution mode, the time setting and display resolution change to tens of microseconds, up to the same 327 ms maximum delay time. All delay time settings are saved in internal memory, even during power down.

In the event of a problem, shutting off the power to the 7922 automatically bypasses all active circuitry, creating a hard-wired shunt from input to outputs, thereby assuring signal continuity. The same relay used for the bypass also prevents spurious signals from reaching the outputs when the unit is first powered up.

Additional References:

'Reference Output,' from the Syn-Aud-Con Newsletter, Summer, 1986, Page 13, Synergetic Audio Concepts, P.O. Box 669, San Juan Capistrano, CA 92693.

'Adjusting Acoustic Response,' from the Syn-Aud-Con Newsletter, Winter, 1986, Page 10.

'Time Alignment of Sound Reinforcement Equipment,' by Pat Maloney, Recording Engineer/Producer, December, 1980, Page 52.

'Time-Aligned™ Loudspeaker Systems,' by Dean Austin, db Magazine, March, 1979.

'Impulse Alignment of Loudspeakers and Microphones, Part I' by Tom Lubin & Don Pearson, Recording Engineer/Producer, December, 1978.

'Impulse Alignment of Loudspeakers and Microphones, Part II' by Gary Leo & Don Pearson, Recording Engineer/Producer, February, 1979.

'Time Alignment™ in Loudspeakers,' by Edward M. Long, Audio, Vol. 61, No. 8, August, 1977.

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