THE DIGITAL DELAY ADVANTAGE

A guide to using Digital Delays

Synchronize loudspeakers
Eliminate comb filter distortion
Align acoustic image

Contents

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Special thanks to Hans Drobilitsch of Hans Drobilitsch Audio GmbH. (Wollersdorf, Austria) for his invaluable technical advice.

Why Digital Delays?
The most intelligible sound occurs when two people speak face to face. The sound is loud and dry and the direction of the sound aligns with the speaker. It stands to reason that the most intelligible sound systems are the ones that come closest to emulating face to face communication. If this is your goal, a digital delay is essential to your sound system.

Until recently, a digital delay's cost was prohibitive for the average user. Only high-end applications could justify the cost. But recent drops in component prices now put the benefits of digital delays within affordable reach of every user.

There are three distinct applications for digital delays. The first and most important is **synchronization of the loudspeakers** to control excess reverberation and echo. Secondly, digital delays help **control comb filter distortion,** and finally, digital delays are useful for **aligning the acoustic image** so the direction of the sound seems to come from the performer rather than the loudspeaker.

This guide goes beyond the typical operating manual that explains only the front and back panel adjustments. Instead, we discuss the basic acoustical concepts needed to get the most out of your digital delay and present examples of several practical applications.

Loudspeaker Synchronization

Sound travels at about 1,130 feet per second in air, or about 1 millisecond per foot. On the other hand, electronic signals travel almost one million times faster through your sound system to the loudspeakers — effectively instantaneous. The main task of digital delays is to synchronize multiple loudspeakers so the sound traveling different distances through air arrives at the listener's ears at about the same time. Synchronizing the loudspeakers reduces reverberation and echoes for improved intelligibility.

How to Synchronize Your Signals
There are several powerful tools available for precisely measuring the time a loudspeaker signal takes to arrive at a certain point in the audience. Most of these tools are very sophisticated and tend to be quite expensive. Fortunately, simpler tools are sufficient for most applications.

In the 1930s, engineers synchronized the low and high frequency speakers in movie theaters by feeding a sharp click through the system. They moved the speakers until they could only hear a single sharp click coming from both speakers. You can use this same method with a common child's toy called a clicker. Pressing the thin metal strip makes a loud sharp click. A clicker is especially useful when synchronizing the direct sound from the performer with the sound from the loudspeakers.

Alternatively, you can use a phase checker especially for synchronizing the signals of two loudspeakers (either LF and HF or two full range systems), since most of the phase checkers include a click generator and receiver. Phase checkers are quite affordable and have other uses besides synchronization.

Processing (or Group) Delays
Converting signals back and forth from the analog to digital domain will slightly delay the signal. These conversion delays are often called processing (or group) delays, and usually range between 0.9 to 5 milliseconds. You will notice that Sabine delays display the processing delay as the smallest possible delay value. You can simply bypass the unit for 0 seconds delay.

Not all manufacturers acknowledge processing delays in their specifications, but you must take them into account when synchronizing your system. Make sure all digital equipment is on and not bypassed when synchronizing. Also, be careful to make an appropriate adjustment in your delay lines if you later add any type of digital equipment to the system.

Center Cluster Speakers
Center cluster speakers offer several advantages over systems that have speakers mounted on the sides. The most obvious advantage is that the distance to the closest and most distant locations in the audience is almost equal, so most listeners hear similar levels of amplified sound. Center clusters also offer two other advantages regarding the visual imaging.

Studies have shown that people can detect even small horizontal changes in the direction of a sound source, but vertical shifts are much less noticeable. This suggests that the sound from center-cluster speakers is more likely to be visually aligned with the performer than loudspeakers placed on each side of the stage.

All those in the audience who are closer to the performer than the center cluster will hear the direct sound from the performer before they hear the sound from the loudspeakers. This makes the sound seem to come from the performer, not the loudspeakers. (See the Precedence Effect below.)

Comb Filter Distortion

Many who took high school science may remember ripple tank experiments where waves are generated from two separate point sources. The waves from each source combine to form visible interference patterns. In some places, the wave crests and troughs are in phase so they combine to make a larger wave. In other places the crests are out of phase, so the crest of one wave source is canceled by the trough of the other. Ripple tank experiments show the interference patterns are strongest when the amplitude of the waves from each source is equal.

A similar interference occurs in sound systems when a signal is delayed and mixed back into the original signal. These interference patterns are called COMB FILTERS because their frequency response plots look like the teeth of a comb (see Figs. 1 & 2). There are a number of common situations that cause comb filters. For example, when the program is played through two loudspeakers, the loudspeaker that is farther away interferes with the closer loudspeaker. Comb filters are also created when a performer is picked up by two microphones, one closer than the other. You even introduce comb filters by mixing digital effects back into the "dry" signal at the mixer's effects loop.

Fig. 1:
COMB FILTERS. Input signal mixed with a 2 msec. delayed signal (both signals have the same amplitude); max. filter gain is +6 dB, and max. depth is - ∞ dB).

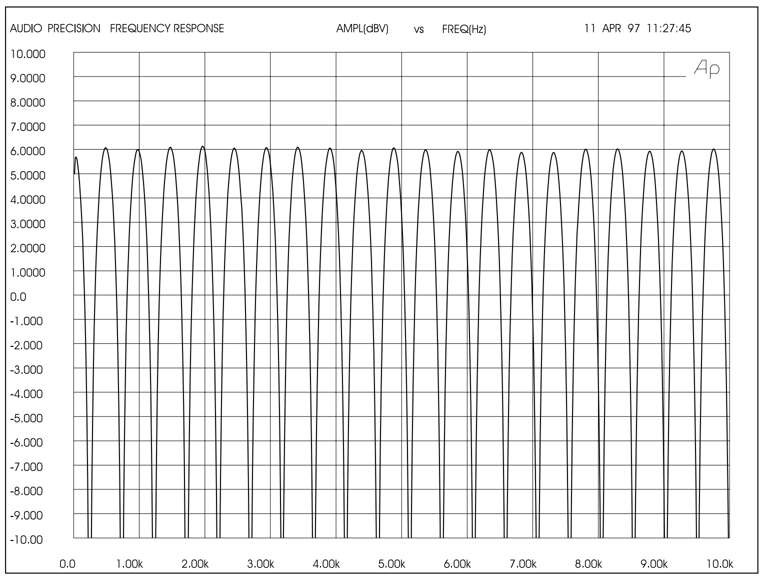
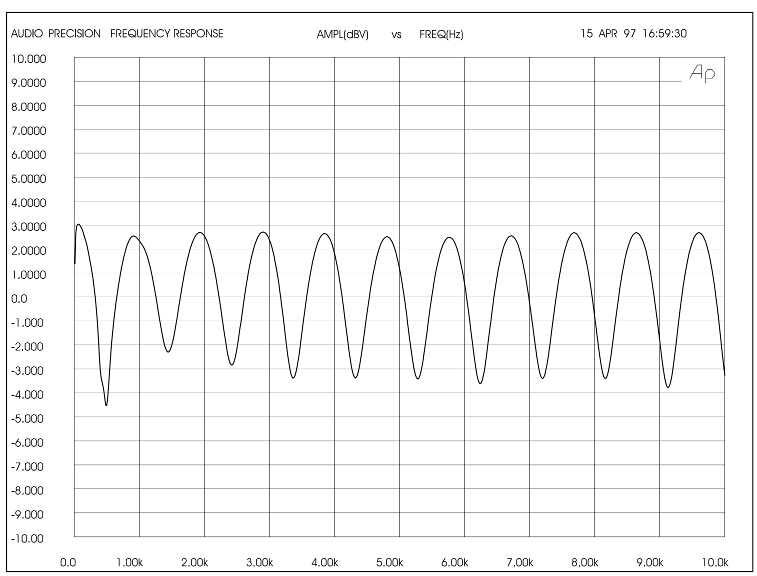


Fig. 2:
COMB FILTERS. Input signal mixed with a 2 msec. delayed signal (delayed signal has 10 dB less amplitude; max. filter gain is +2.5 dB, and max. depth is -3). Reducing the amplitude of the delayed signal reduces the comb filters' effect.



Calculating Comb Filter Frequencies

The reinforcement and cancellation frequencies depend on the delay time (the time difference between the arrival time of the original signal and the delayed signal). The frequency of the first cancellation occurs at 1/(2 x t) Hz, where t = the delay time in seconds. The cancellations are separated by (1/ t) Hz. Fig. 3 shows how the comb filters change with the delay time.

Fig. 3:
Comb filters get closer as delay time increases.

Delay time = 0.002 sec.		Delay time = 0.003 sec.		Delay time = 0.004 sec.	
Cancellation Freq. (Hz)	Reinforcement Freq. (Hz)	Cancellation Freq. (Hz)	Reinforcement Freq. (Hz)	Cancellation Freq. (Hz)	Reinforcement Freq. (Hz)
250	500	167	333	125	250
750	1000	500	667	375	500
1250	1500	833	1000	625	750
1750	2000	1167	1333	875	1000
2250	2500	1500	1667	1125	1250
2750	3000	1833	2000	1375	1500
3250	3500	2167	2333	1625	1750
3750	4000	2500	2667	1875	2000
4250	4500	2833	3000	2125	2250

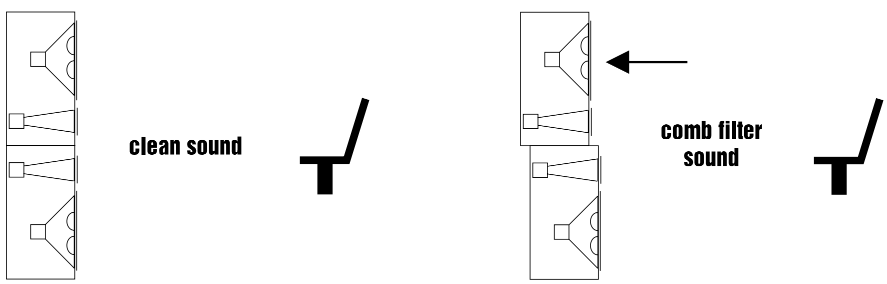
Comb Filter Amplitude

If the original signal and the delayed signal are the same amplitude, the reinforced frequencies increase in amplitude by 6 dB, while the out-of-phase frequencies cancel completely to $-\infty$ dB.

Comb filters cause a lot of problems. The frequencies that are reinforced are prone to excite feedback, while the out-of-phase cancellations make the program sound thin and over equalized.

Try this simple experiment to hear what comb filters do to your sound.

Fig. 4:
Comb filters noticeably affect your sound.



Stack two identical full-range loudspeakers as shown in Fig. 4. Carefully align the HF horns and wire the speakers in mono. Stand in front while listening to your favorite full-spectrum CD. Ask a friend to move the top speaker slowly away from you. The degradation in sound quality you hear is caused by comb filters. The experiment is most dramatic when you use good quality speakers.

Correcting Comb Filters

Comb filters are inevitable to some degree in every live sound system, and they cannot be corrected with equalization. Fortunately, most comb filter problems can be reduced to a minimum by synchronizing the signals and reducing the amplitude of the delayed signal. The examples below show several practical applications.

The Precedence Effect: Aligning the Acoustic Image

Helmut Haas published a study in 1951 describing a series of experiments that demonstrated how people perceive delayed signals and echoes. In his experiments, a listener was positioned between two speakers placed 3 meters away; one was placed 45 degrees to the right and the other was placed 45 degrees to the left. When the same program was played through both speakers simultaneously, the listener perceived the acoustic image (the direction from which the sound seemed to be coming) centered between the speakers.

When Haas delayed the signal going to one of the speakers by somewhere between 5 to 35 milliseconds, the listener perceived a shift in the acoustic image to the speaker heard first. While the delayed speaker did not contribute to the apparent direction of the sound, it did make the program seem louder and “fuller.”

Haas showed that you must increase the loudness of the delayed signal by about 8 to 10 dB (twice the perceived loudness) in order for the acoustic image to move back to the original center position. Increasing the loudness more than this, or increasing the delay somewhat more than 35 milliseconds, makes the delayed signal sound like an echo.

The phenomenon describing how the acoustic image follows the signal we hear first is called the Precedence Effect. The phenomenon that makes two distinct sounds heard less than 35 msec. apart seem like only one sound is call the Haas Effect. However, the terms are often used interchangeably in the sound industry.

THREE APPLICATIONS

APPLICATION I: Under-The-Balcony Speakers

Fig. 5:

Overhead view of under-balcony application.

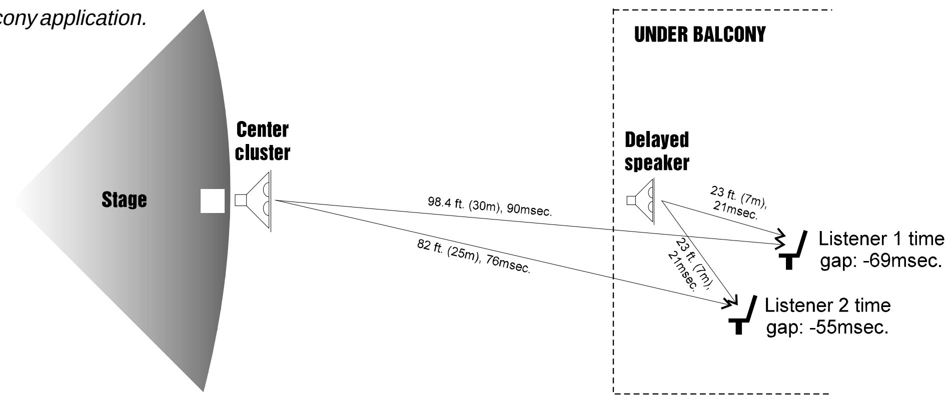


Fig. 5 above shows a typical situation where the performer is amplified by a center cluster hanging above the stage. Almost everybody in the audience will enjoy good sound, except those seated in the shadow of the balcony. So we add an under-balcony speaker to fill in the shadow.

Now we have sufficient volume under the balcony, but the sound from the two speakers arrives at the listener's ears some 76 to 84 milliseconds apart. The two signals, along with their echoes, result in an unintelligible cacophony. We must delay the sound from the under-balcony speaker to synchronize the signals. Do we set the digital delay to 76 or 84 milliseconds? Obviously, the geometry will not allow us to exactly synchronize every location under the balcony; we have to compromise.

First, you must consider the program type. For spoken word programs, you will produce the best intelligibility if the signals from the under-balcony speakers arrive within 10 msec. of the signals from the center cluster. Therefore we should set the delay to 84-86 msec. You can allow a little more reverberation for programs that are mostly music.

Next, we must eliminate comb filter distortion. Find the axis where the levels of the center cluster and under-balcony speaker are equal. Use the digital delay to precisely synchronize the speakers along this axis to eliminate the most severe comb filters. Comb filters off the equal-level axis are much less of a problem since a louder signal is not affected very much by a weaker signal.

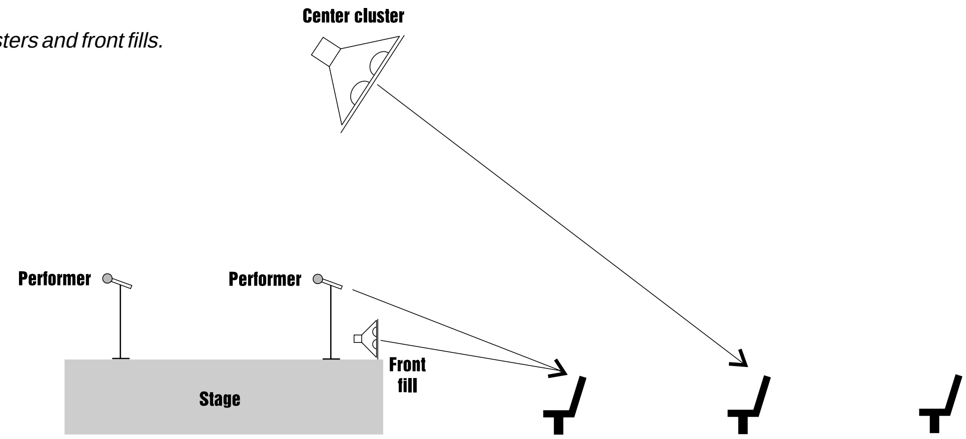
Finally, you can experiment with adding 5 to 10 milliseconds delay to both sets of speakers to take advantage of the Precedence Effect for the audience seated near the performer.

In the final analysis, every setting is a compromise, and your ear has to be the final judge. Check the sound in several different locations throughout the auditorium and correct the most severe irregularities.

Application II: Center Cluster with Front Fills

Fig. 6 below describes a typical application that has a stage with a microphone, a center cluster above the stage, and front fills in front of the stage. There must be thousands of installations throughout the world like this that "get by" without digital delays. But with the Sabine digital delay, you can improve the intelligibility and add a new quality without ringing up any significant costs. Use the digital delay in this situation to align the visual image with the acoustic image. The program is much more enjoyable when the amplified sound seems to be originating with the performer, not the loudspeakers.

Fig. 6:
Synchronizing center clusters and front fills.



Find a central place in the audience where the center cluster is 6 to 8 dB louder than the direct sound from the performer. Delay them so that their sound arrives 5 to 8 milliseconds after the direct sound from the performer. Experiment by bypassing the digital delay in and out to hear how the source of the sound seems to move from the loudspeakers to the performer and back. Now your ears have the same directional information as your eyes, so the performance will sound more natural and exciting. The best seats in the house just got better.

What about the front fills? Their purpose is to add intelligibility and listening comfort to the first few rows nearest the stage by filling in the areas missed by the center clusters. Simply add about 8 msec. to the front fills to take advantage of the Precedence Effect.

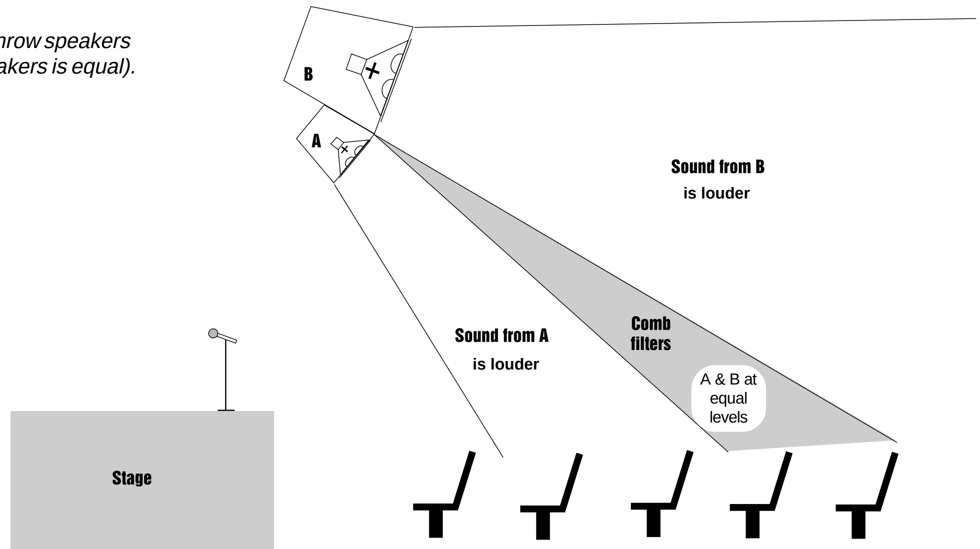
The 8 msec. setting presumes the performer is standing on the front few feet of the stage. But some stages are well over 30 feet deep. What if there is a second performer standing 25 feet behind the first? The direct sound from his or her voice will reach the first few rows about 25 msec. after the first performer's. The audience will perceive the first performer directly and the second performer through the loudspeakers.

We can add the advantage of the Precedence Effect to the second performer by placing an digital delay in the mixer's channel insert point and adding a 25 msec. delay.

Certainly taking advantage of the Precedence Effect is not as obvious to the audience as eliminating feedback, but it is nice to know you did all that is possible to make the performance enjoyable.

Application III: Synchronizing the signals of a far-throw and short-throw loudspeaker.
In order to reach the proper coverage in larger venues, we often stack two full range speakers - a short-throw center cluster for the audience below and a far-throw speaker for the back of the auditorium. It is almost impossible to perfectly align the stacked speakers mechanically, so comb filter distortion becomes a problem in the area where the levels from both speakers are equal. The same thing happens with speakers mounted on the right and left sides.

Fig. 7:
Aligning far- and short-throw speakers (the level from both speakers is equal).



It is impossible to remove comb filters with equalization, but a Sabine digital delay eliminates them in short order without affecting the spectral balance for the rest of the audience. Find the axis where the levels from the two speakers are equal. This is where the comb filters are most severe. Carefully adjust the digital delay so that the signal from both speakers arrives at precisely the same time.

Use the same procedure to align speakers within a cluster when necessary.

CALCULATING DELAY TIME USING DISTANCE

Calculating delay time in terms of distance is a common and accepted method. For a good start, estimate the delay at 1 millisecond per foot between speakers. Use the following equation for more precise estimates:

$$\text{Delay (milliseconds)} = 1000 \left(\frac{\text{D (distance in meters)}}{344} \right)$$

OR

$$\text{Delay (milliseconds)} = 1000 \left(\frac{\text{D (distance in feet)}}{1130} \right)$$

These measurements presume standard temperature (68 degrees F, 20 degrees C) and pressure (29.2 in. Hg., 760 mm Hg.). Sound travels slower in cooler or drier air and at higher pressures. For example, the speed of sound decreases about 0.61 meters per second as the temperature drops from 20 degrees C to 0 degrees C.