Creative Labs, Inc.
1901 McCarthy Blvd.
Milpitas, CA 95035

Phone (408) 428 6600
Fax (408) 428 6611
www.creative.com

EAX® 4.0

Designer's Guide

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EAX 4.0

Designer's Guide

Creative®'s EAX™ 4.0 property sets can be used with OpenAL or with the Microsoft® DirectX™ game and multimedia-programming environment. EAX allows applications to apply digital effects processing to 3D sounds. These effects include both 'environmental' effects such as reverberation, reflection, occlusion, obstruction, and some studio-style effects like chorus, flanger, echo, EQ, and distortion.

The Designer's Guide suggests how designers of interactive 3D audio applications can use EAX in their projects. The document also features a full description of the acoustic models behind EAX's environmental effects, and descriptions of the studio-style effects additionally available. The guide is intended to give you, the sound designer, a background to select, customize, and deploy positional audio and EAX effects, so you can give your interactive project the exact same dynamic and lifelike soundtrack you have imagined in your head!

Introduction to Environmental Audio

Environmental effects add a sense of surroundings to positional audio. EAX 4.0 compatible audio devices such as SoundBlaster Audigy and Audigy2 provide environmental effects on many of today's PC systems. Correctly used, the effects can really bring a game's soundtrack to life. This chapter introduces acoustic phenomena, and discusses how they occur in real life and why they should be used in virtual worlds.

Benefits of simulated acoustics

Using a 3D audio API such as DirectSound3D or OpenAL, an application can create a three dimensional audio world, with sounds mixed and rendered according to their position relative to a listener. Each positional effect that these APIs offer involves simulating the way that sound travels directly from the source to the listener.

However, DirectSound3D and OpenAL have no internal support for effects which model the way that sound interacts with the surroundings. Such 'environmental' effects include sound reflecting off surfaces creating reflections and reverberation, and sounds becoming muffled by absorption and diffraction when a surface blocks the direct path between the source and the listener.

Environmental Reverberation can give the gamer an audible indication of the characteristics of their surroundings. Reverb effects can provide the user with sonic cues to differentiate between locations and reinforce the feeling of immersion.

Reflected sound also helps when the user is judging the distance between a source and the listener. A sound's 'direct path' intensity varies with distance, but reflected sound from a sound source remains at much the same relative volume wherever the listener is within an environment. Therefore, the brain compares the sound perceived directly from the source to the reverberated sound, determining the direct sound to reverberation ratio (DRR). If the DRR is high (the sound's direct path is much louder than its reverberant component), then the sound source is probably nearby. If the DRR is low, the sound source is probably distant.

Sound muffling enhances the realism of an application by simulating the way architectural features such as walls and pillars might block and absorb sound. The muffling effects can reinforce the visual perception of these features, representing obstacles of different thickness and materials by varying the attenuation and filtering applied to sounds passing through them.

Reverberation effects

Principles of room acoustics

We have differentiated between the direct sound and the sound that reflects off surfaces in the surrounding environment before reaching the listener's ears. Now, we will further dissect the reflected sound to identify the different stages of reverberation, and how those stages are affected by the surroundings. We will assume the listener is located in a simple room that has four walls, a ceiling, and a floor, all with acoustically reflective surfaces.

Early Reflections in a Room

As well as the direct sound, the listener also hears a reflection of the sound source from each of the walls as well as from the floor and the ceiling. Each of these one-bounce reflections is called a primary reflection or a first-order reflection. Two-bounce reflections are called secondary reflections or second-order reflections.

Although the primary reflections reach the listener's ears a split second later than the direct sound, the brain integrates these reflections with the direct sound because their content is similar and they follow closely in time. The integrated sound seems louder than the direct sound alone, and it may take on some tonal coloration. If the reflector is far off, the reflection comes much later and sounds like a separate sound source: an echo.

Sound modifications added by primary reflections are environmental audio cues: they give the brain some indications of the immediate surroundings of the listener or the source. A strong and immediate primary reflection, for example, tells the brain that the walls in the environment are close. The change in tonal coloration may also tell the brain something about the reflective quality of the wall — whether it is highly reflective or somewhat absorptive, muting the reflections. However, the brain cannot tell exact room geometry from primary reflection cues (or from the subsequent reflections and reverberation).

If the listener or the sound source moves within the room, the primary reflection cues change: the perceived sound (which has integrated the direct sound and the primary reflections) changes its quality and relative volume. This continuous change provides the brain with more specific information about the locations of reflecting walls. (This reinforcement through continuous cue changes is much like the way that a moving sound source's direct positional audio cues reinforce the sense of the source's location.)

Now let us consider the simple room we just looked at and follow the reflections further. Primary reflections are reflected from the walls, floor, and ceiling, creating a larger number of secondary reflections. Each secondary reflection is reflected twice between the source and the ears, and is likely to overlap with other reflections at the ears of the listener. These reflections are less specular than the primary reflections; that is, they do not resemble the direct-path sound as closely as the primary reflections do, and they lose much of the sense of specific location that primary reflections have. That is because they typically overlap each other and because each time the sound is reflected and transmitted through the air, it loses some of its clarity and becomes more and more diffuse (or "smeared"). This is particularly true if the walls are not highly reflective, are irregularly textured, or both.

Secondary reflections are also lower in amplitude than primary reflections because they follow later and there is always some sound absorption in reflection and transmission. The brain integrates secondary reflections with the direct sound as it does primary reflections and uses them as further environmental audio cues. These reflections contain less specific information about the environment because they have been reflected twice, and have lost clarity.

Reverberation in a Room

Following the sound in the room even further, we see that secondary reflections can have reflections of their own, and these reflections have reflections, and so on until the final reflections of reflections are so diminished in volume that they are inaudible. A sound source's full set of reflections in a room is incredibly complex. Remember, too, that each reflection loses specularity. A good visual analogy is to imagine dropping a bar of soap into a bathtub. The first ripples are clear, but as they reflect and re-reflect from the sides of the tub, they create increasingly smaller sets of ripples until finally the surface of the water has no discernible waves, just a choppy surface all over.

The merging of distinct sound reflections into an overall sonic wash is exactly what happens to sound reflections after the first- and second-order reflections. As each reflection loses specularity, the overall effect is a sound tail that provides information about the general quality of the room, not its specific components. This sound tail is most commonly called reverberation.

The brain does not try to pick out distinct reflections within reverberation. It instead perceives the quality of the reverberation as a whole and integrates it with the first- and second-order reflections to provide yet more of an environmental cue. The length and the loudness of the reverberation tell the brain quite a bit about room size and the reflective quality of the walls. The more reflective the walls are and the larger the room, the longer the reverberation lasts. The more reflective the walls are and the smaller the room, the louder the reverberation.

Filtering effects

Previously, we looked at the way that the walls of an acoustic environment affect the sound by reflection. Obstacles can also affect sound by transmission and diffraction – the way that sounds are modified when they travels through and around solid objects. If a sound source and the listener are separated by one or more obstacles, then the sound that reaches the listener will be changed to a varying degree.

Sound Obstruction

Let us imagine the scenario where an acoustically opaque column is placed in the middle of a room, between the sound source and the listener. The direct-path sound wave can only reach the listener by transmission through the obstacle or by diffraction around the obstacle. In both cases, it will be partially or completely muffled. That muffling effect is called obstruction.

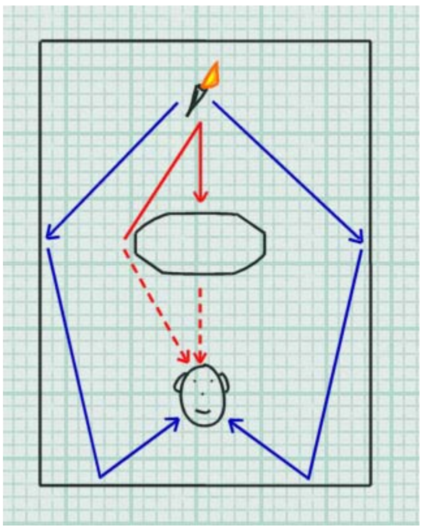

Figure 1 – Obstruction: When an object in a room separates sound from listener, obstruction occurs. In this example, the direct-path sound is muffled through a partially transmissive obstacle or medium.

Sound waves can bend around obstacles that are smaller than their wavelength - this phenomenon is called sound diffraction. The result of sound diffraction is that the listener hears a direct sound source with the high frequencies filtered out. This is because the lower frequency sound, with greater wavelength, can bend around larger obstacles than higher frequency sound. The amount of muffling (attenuation of higher frequencies) due to diffraction depends on the amount of deviation from a straight propagation path: the larger the angle that the direct path (shortest path) must make to go around the obstacle, the stronger the muffling effect.

The tonal effect of transmission through the obstacle or diffraction around an obstacle is similar because materials are typically less transmissive at high frequencies that at low frequencies: in both cases, the direct-path sound is low-pass filtered. However, there is a difference between the two phenomena in the *apparent position* of the sound source. When there is substantial transmission throughout the obstacle, the sound still seems to come essentially from the same direction as if there were no obstacle. In the case of diffraction, the sound seems to come for the edge of the obstacle where the shortest sound path is diffracted.

Because reflected sound waves go around the obstruction for the most part, the obstruction typically blocks only a tiny part of the sound source's reflections and reverberation. These audio cues remain essentially constant with or without the obstruction. The muffling effect of obstruction is essentially confined in the direct sound. The lack of (or muffled) direct sound in combination with normal reflections and reverberation informs the brain that the source is located behind an obstacle.

Sound Occlusion

We now split the rooms, so that the sound source is in one room and the listener is in the adjacent room. Source and listener are completely separated by a wall so there is no direct air connection between them. Any sounds that pass from source to listener must pass through the wall, which muffles the sounds. This is called *occlusion.* It differs from obstruction in that obstruction does have open (although indirect) air space between source and listener.

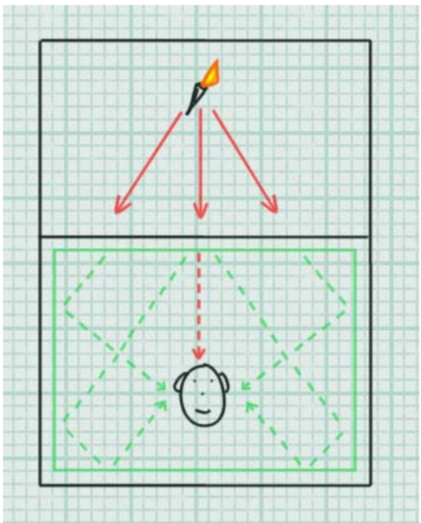

Figure 2 – Occlusion: When a full wall separates the sound source from the listener, it occludes the sound.

The wall that divides the sound source and the listener acts as a big filter. Direct sound waves from the source, along with accompanying reflections and reverberation, hit the wall and are passed through to the other side, where they radiate into the listener's room. As the sounds pass through the wall, they are all altered—typically, their high-frequency components are filtered out, leaving a much muffled result. Note that, as illustrated in Figure 2, the direct sound wave ("first wave front") passing through the wall contributes to the reflected sound in the destination environment.

When the brain hears a muffled sound source along with muffled reflections and reverberation, it can recognize that the source is located behind a wall or other acoustically transmissive material. This is unlike an obstructed source, where the direct sound is muffled but the reflections and reverberation are not. The quality of the muffling tells the brain something about the construction of the wall — whether it is dense, thin, solid, soft, and so on.

Sound Exclusion

Taking the two rooms in which we explained occlusion, we can now make an opening in the wall that separated listener from source, for example a doorway. Source and listener are still separated by the wall, but there is an opening allowing the sound to enter the room and, in certain positions, the direct path between the source and listener is clear.

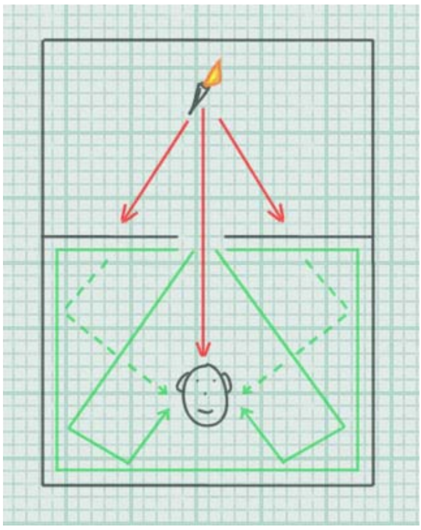

Figure 3 - When an opening breaks the wall separating the sound from the listener, the direct path is clear but the sound remains excluded from the listener's room.

When there is a direct path, a new scenario is defined – exclusion. In this situation, the direct sound is not muffled, but the source can only radiate a small amount of energy into the listener's room through the opening. The amount of reflected sound perceived by the listener from this source's environment depends on the size of the opening and on the distance from the source to the opening. As illustrated in Figure 3, the direct sound wave passing through the opening contributes to the reverb in the destination environment.

The location of the listener can be such that an obstacle blocks the direct path from the source to the listener (that could be the wall separating the two rooms or some other obstacle located in the listener's room). In that case, there is a combination of exclusion (reducing the reflected sound in the listener's room) and obstruction (muffling the direct-path sound).

Approaches to environmental audio

There are two different approaches to simulating environmental sound: *physical modeling* and *perceptual modeling.*

Physical Modeling

Physical modeling is sometimes called "ray tracing" after a similar 3D graphics modeling technique from which its principles were derived. It creates a detailed physical model of a computer-simulated acoustical environment, and it establishes the location of each wall and obstacle as well as the reflective qualities of each surface. It also establishes the positions of each sound source and the listener.

After describing the full physical characteristics of the simulated acoustic world, physical modeling then calculates the path of sound waves from each sound source through the detailed physical model to the ears of the listener. To do so, it must calculate reflections, reflections of reflections, and so on. It must also calculate any obstruction and occlusion effects as well. The combination of the direct sound source and all these effects creates audio with environmental audio cues.

Physical modeling techniques are particularly good at giving the listener a sense of their immediate surroundings. A well-implemented physical model should be able to calculate very realistic early reflections, and it is these initial reflections that can give the human brain clear indications about the proximity and reflective quality of nearby walls. Furthermore, the way that the early reflections change as the listener and sound sources move around an environment reinforces these cues.

On the other hand, physical modeling is not so adept at handling late reverb, because the processing required to trace later reflections can quickly become resource-hungry. The more subtle aspects of reverberation can reveal important information about the listener's surroundings. Variations in the tone and granularity of late reverb can tell us much about a room's layout, contents, and surfaces. An environmental audio implementation that cannot reproduce these phenomena will not be able to tell the listener the whole story of their surroundings.

Physical modeling is also not very good at addressing aesthetic concerns. The world of hearing is a world of general, almost subconscious impressions. To create a convincing illusion of a 3D audio environment, early reflections calculated realistically in a physically modeled environment are often not enough. For a sound designer to realize their vision for environmental sound, they must have at their fingertips full control over all facets of the effects. With physical modeling, if the acoustics generated by certain geometry do not please the designer, they have to get involved with redesigning the room itself, until they get the sound they want.

Perceptual Environmental Audio Modeling

Perceptual modeling takes a very different approach than physical modeling, and can create very convincing environmental effects while minimizing the necessary computing resources.

While physical modeling works by keeping track of every reflecting surface and all the sound paths from each sound source to the listener, perceptual modeling concentrates instead on environmental audio effects at the listener's ears. Perceptual modeling focuses on the use of Digital Signal Processing (DSP) to effectively reproduce environmental sound phenomena, while giving the developer control over the most salient perceptual attributes of a 3D audio environment.

Perceptual modeling uses statistical models of real-world measurements to reproduce different types of acoustic environments. Each statistical model defines the DSP effects to apply to a sound source to make the source seem to exist within the specified acoustic environment.

Examples of perceptually relevant features are the arrival time of the first significant reflection from a source (more significant than the time of subsequent reflections) or the overall decay rate of the reverberation (more significant than the amplitude of any of the individual reflections that constitute the reverberation). So the statistical model for a room might define the time delay between the initial dry sound and the first reflection, how long reverberation takes to die out (its decay time), the ratio of late reverberation to early reflections, and more.

A statistical model for occlusion will define how much attenuation and filtering should be applied to a sound's direct path, and how much should be applied to the sound's reflected paths. These selected audio cues offer direct control over the listener's sensation of the acoustic environment.

As statistical models use real-world measurements, they take into account sound-wave diffusion, random dispersion of high frequencies, and other environmental factors, which would demand heavy calculation with a physical model. This means that the statistical approach can produce more lifelike late reverberation, at less computational cost. The perceived naturalness of statistical models such as these stems directly from the fact that our brain actually learns to understand our acoustical surroundings from the repeated observation of typical acoustical phenomena in rooms.

Because it is impossible to create a statistical model for every potential type of room, perceptual modeling uses extrapolation from statistical room models based on established theories of room acoustics and psychoacoustics (the science of sound perception). The resulting facilities for "effect tweaking" allow the designer to create an infinite variety of room types for environmental audio.

The EAX approach to environmental audio

In essence, EAX represents a perceptual approach to environmental audio. EAX is based round a statistical reverb model, consisting of sets of properties, which together define how sound should be processed to produce the desired environmental effect.

However, the design of the EAX interface allows for the dynamic adjustment of these properties in real time. This means an application's designer can integrate some aspects of physical modeling into their audio implementation, if they wish to. EAX itself has no facilities for automatically calculating reverb parameters based on room geometry. However, the application developer may decide to write their own algorithm for setting, for example, the early reflections level, based on the distance between the listener and the nearest wall. It is simple for them to apply their calculations by adjusting an EAX property each time the listener's position changes within an environment.

EAX environmental reverb model

In the case of Environmental Reverb, the EAX statistical model splits the effect into two sections, early reflections, and late reverb. Anyone who has used EAX (or studio reverberation processors) before will recognize the reverberation response model shown in Figure 4. The line at the left represents the original or direct sound. After the reflections delay, a group of reflections is heard, followed by the reverberation.

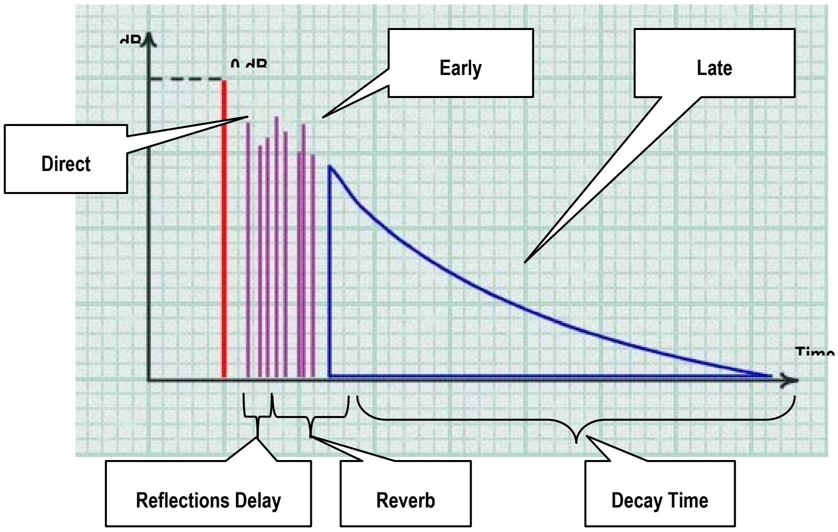

Figure 4 - EAX Reverb Response Model

The EAX 4.0 interface provides the designer with a great deal of control over the shape and tone of the environmental reverb. For each component, early reflections, and late reverberation, tonal adjustments are possible allowing simulation of a wide variety of different surface materials. Time delay and decay parameters allow the design of effects for rooms and spaces of all different shapes and sizes. The designer can introduce cyclic echoes, helping to simulate fluttering sounds characteristic of empty rooms or open environments with significant parallel surfaces.

You can find out more about designing environmental reverb effects with EAX 4.0 in the next section, Designing Environmental Reverb effects.

EAX environmental filtering model

To simulate each of the three sound filtering scenarios described in Filtering effects, EAX 4.0 includes the ability to attenuate and filter a sound source's Direct and Room paths. The appropriate modifications for each situation are gathered together under EAX 4.0 source properties Occlusion, Obstruction, and Exclusion. These are high-level controls, allowing the developer to apply the correct filtering effects on a per-source basis using simple commands, without having to calculate and manipulate the attenuation and filtering properties themselves.

You can find our more about designing environmental filtering effects with EAX 4.0 in Designing Environmental Filtering effects.

Designing Environmental Reverb effects

The previous chapter explained the basic theory behind the different environmental effects available in EAX 4.0. Designing an environmental effect to model the acoustics of a location in a virtual world is perhaps the most involved process that the audio designer will undertake when working with EAX - and probably the most satisfying. This chapter delves deeper into the phenomenon of environmental reverberation, and discusses how to go about designing a reverb for any situation or location.

Using reverb effects

With today's advanced 3D graphics technology, the lifelike visual appearance of a user's surroundings can set very high user expectations in respect to the sophistication of the audio scene. Suitably designed reverb effects will help an application's interactive soundtrack to match the standards of realism set by the graphical rendering of the world. EAX 4.0's fully controllable environmental reverb effect gives an audio designer the sonic palette to create a lifelike simulation of almost any acoustic environment. This flexibility allows the designer to introduce flair and imagination to effects, and thereby inject some distinctive personality into the soundtrack.

Definition: Environmental Reverb presets

The EAX reverberation response is defined by the following parameters:

- The energy in the Reflections and Reverb sections at mid frequencies
- The Reflections Delay and the Reverb Delay
- The Reflections Pan and Reverb Pan vectors, for spatial/directional distribution
- The "Room filter," which affects the Reflections and the Reverb identically and allows correcting their energy at low and high frequencies
- The Decay Time at low, mid and high frequencies
- The reference low and high frequencies
- The Diffusion of the reverberation
- The Echo and Modulation parameters, which affect the late reverberation.

In the EAX 4.0 Introduction, we explained how EAX's environmental reverb properties allow you to tweak every aspect of environmental sound. In total, the reverb effect exposes 24 different parameters. This includes low-level properties, and high level controls, which adjust a subset of low-level properties. Some of the high-level properties can be rendered ineffective by various switches.

An Environment is characterized by the values of all reverberation response parameters, when the distance between source and listener is equal to the source's 3D property Minimum Distance. This defines an "Environment preset."

Approaches to designing Reverb Effects

There are a number of different approaches that you can take to designing EAX 4.0 reverb effects for an application's audio implementation.

Select an environmental preset from the SDK

The EAX 4.0 SDK includes a set of more than eighty environmental reverb presets, designed by Creative's audio specialists. These ready-made acoustic models are presented in a number of different formats. They are stored as EAGLE environment models, for use with Creative's EAGLE 3.0a environmental audio design tool. They are also available in a C/C++ format, as #define macros in the SDK file EAX_UTIL.H. That file can be included in a project, or the individual macros can be cut-and-pasted directly into a programmer's own code.

The models are broadly categorized by scenario. For example, there are sets of acoustic simulations for different rooms in a castle, or in a cave. There is one group of effects, which are relevant to driving simulations, another one containing effects which might occur in a virtual city. There is also a group that contains settings for the twenty-six presets included in the original EAX 1.0 release. Check out the EAX 4.0 Preset Description document for more details on how the effects are arranged.

Therefore, choosing an effect for each location in your application could be as simple as selecting from the list a preset that best fits your criteria.

Customize an environmental preset using high level controls

As specified in the EAX 4.0 Introduction, EAX's Environmental Reverb effect features a 'high-level' control, Size, which make broad changes to an acoustic model. The 'Size' parameter adjusts the set of low-level properties, which can help to indicate the size of a room (reflection delay time, reverb decay time etc…), whilst preserving an environment's general characteristics. Therefore, a more flexible approach is to choose the preset that is closest to the sound you want, and then make broad adjustments using Size, to match the dimensions of the room you are simulating.

The high-level approach is a fine way for a programmer to quickly add the benefits of environmental reverberation to a positional audio application. It is also a useful mechanism allowing the designer to simply create variations on the preset reverb effects. However, although using the provided presets alongside the Size parameter will allow you to cover a broad range of different environments, there are always going to be situations where custom designing an effect will give you the added control you need.

Build an effect from scratch

Any enthusiastic audio designer will be excited by the prospect of being able to control the powerful EAX 4.0 reverberation engine for a specific application. In the same way that a development team will use specially designed models, textures and lighting effects to give the three dimensional world a unique look, custom-designed statistical models of room acoustics can give an application a truly distinctive sonic identity.

Static Modeling

The way that reverberation sounds in an environment is broadly determined by the size of it and the absorptive quality of the boundaries. The contents of the room also have some effect on the reverb.

Dynamic factors such as listener and source position have a direct bearing on the way that sound propagates in an environment, and EAX takes into account the distance between the listener and a source when it automatically sets the reverb level for each source. The EAX statistical approach to environmental modeling means that you do not have to account for these dynamic factors when designing static reverbs. You can create lifelike reverb settings by using 'best fit' reverb models with static values for its properties.

However, it is always possible to add some extra dynamics, so that the environmental reverb changes as the listener moves within a room. The next section (see *Dynamic Modeling*) details the circumstances in which some reverb characteristics change as the listener moves around, and shows how to adjust these parameters in real time to achieve an even more lifelike simulation.

Let us go ahead and start with possibly the most important factor – the room size.

Room Size

In general, it can be understood that a room's size will contribute particularly to the temporal aspects of a room's acoustic model. In normal atmospheric conditions, sound travels at 344 meters (1129 feet) per second, which is approximately equivalent to one foot per millisecond, or one meter every three milliseconds. When calculating factors such as Reflections Delay and Reverb Delay, you should bear this in mind.

ENVIRONMENT SIZE

Value range **1.0 to 100.0**
Default value **7.5**
Value units **Meters**

Environment Size is a 'high level' control that can be used to make coarse changes to the current environment. Changing this parameter alters other properties to simulate the acoustics of a room of different size but otherwise similar properties (wall geometry and materials).

By default, this property will automatically adjust a collection of lower level properties, *Reverb*, *Reverb Delay*, *Reflections*, *Reflections Delay* and *Decay Time* (properties described in more detail in the following sections). It can also adjust two additional properties: *Echo Time* and *Modulation Time*. Changing the value for *Environment Size* will scale the values of the related lower level properties. You can decide whether *Environment Size* will affect any of these seven related lower level properties by setting the related property scale flag.

Environment Size also controls a 'hidden' parameter called Reverb Density, which is not directly exposed to the designer. See Small Rooms for more details.

Property scaling flags

Decay Time Scale

If this flag is TRUE, a change in Environment Size causes a proportional change to the Decay Time. If it is false, changes to Environment Size do not affect it.

Reflections Delay Scale

If this flag is TRUE, a change in Environment Size causes a proportional change to the Reflections Delay. (In effect, as the room gets larger the nearest walls get more distant, so reflections take longer to reach the listener) If it is set to false, changes to Environment Size do not affect Reflections Delay.

Reverb Delay Scale

If this flag is TRUE, a change in Environment Size causes a proportional change to the Reverb Delay. If it is false, changes to Environment Size do not affect Reverb Delay.

Reflections Scale

If both this flag and the Reflections Delay Scale flag are TRUE, an increase in Environment Size value causes an attenuation of the Reflections level. If one of the two flags is FALSE, a change in Environment Size has no effect on Reflections.

Reverb Scale

If this flag is TRUE, an increase in Environment Size value causes an attenuation of the Reverb level. If it is FALSE, a change in Environment Size has no effect on Reverb.

Echo Time Scale

If this flag is TRUE, a change in Environment Size value causes a proportional change of the property Echo Time. If it is FALSE (its default value), a change in Environment Size has no effect on Echo Time.

Modulation Time Scale

If this flag is TRUE, a change in Environment Size value causes a proportional change of the property Modulation Time. If it is FALSE (its default value), a change in Environment Size has no effect on Modulation Time.

Reflections Delay

Value range	**0.0 to 0.3**
Default value	**0.007**
Value units	**Seconds**

Reflections Delay controls the amount of time it takes for the first reflected wave front to reach the listener, relative to the arrival of the direct-path sound. The smaller the value the less time it takes the early reflections to reach the listener. Overall, a higher value will simulate a larger environment where both the listener and the source are distant from the walls.

Reverb Delay

Value range **0.0 to 0.1**
Default value **0.011**
Value units **Seconds**

Reverb Delay sets the amount of time it takes for the reverberated sound to reach the listener. Higher values imply a larger room; lower values a smaller room. This value defines the time between the start of the Reflections and the start of the Reverb. During this phase, reflections will be heard.

Imagine a square room, with walls twenty feet apart. The listener is located in the center of the room, and makes a noise. Discounting the floor and ceiling, the first early reflections from the walls will return to the listener's ears after a journey of (2 x 10 feet) 20 feet. Therefore, the delay between the initial (direct) sound and the appearance of the first reflected sounds will be around 20 milliseconds. This case represents the longest possible reflection delay for this room.

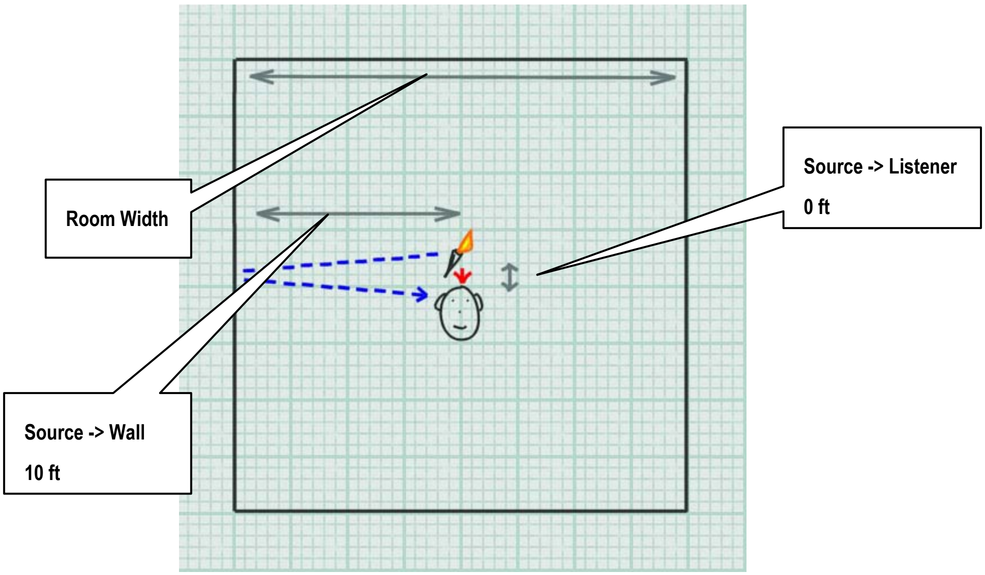

Figure 5 – Worst case: reflections delay approximately 20 milliseconds

Of course, this is only true when sound source and listener are positioned in the center of the room – move them both closer to a common wall and the delay would be much shorter. Again, reckon on one millisecond for each foot in the path traveled by sound moving from the source to the wall and then the listener. If the source is located at a distance from the listener, then in reality things get more complex - the reflections delay will be offset by the time taken for the direct path to reach the listener's ears.

When designing a reverb model, estimation for the time delay between the direct sound and the first reflections reaching the listener's ears must be made based on the most likely set of circumstances. As EAX does not currently support localization of reflections on a 'per-source' basis, we must make a generalized assumption regarding the location of sound sources. A reasonable supposition could be that in an application which focuses around the user's on-screen self, many prominent sounds in the application originate at, or near to, the listener's position.

In addition, it is likely that the position of a moving listener will, over time, be distributed evenly between the center of a room and its perimeter. If you are implementing a dynamic system, you can account for changing listener position by varying the

Reflection Delay. However, if you are designing a static environment for a given room, you will need to assume an average proximity between the listener and the nearest wall. A larger room gives the listener the potential to be further from a reflecting wall, implying longer reflection delay times. Figure 6 shows a typical scenario for the twenty-foot square room, giving a Reflection Delay of around twelve milliseconds when the listener is at a typical distance of around 6 feet from the wall.

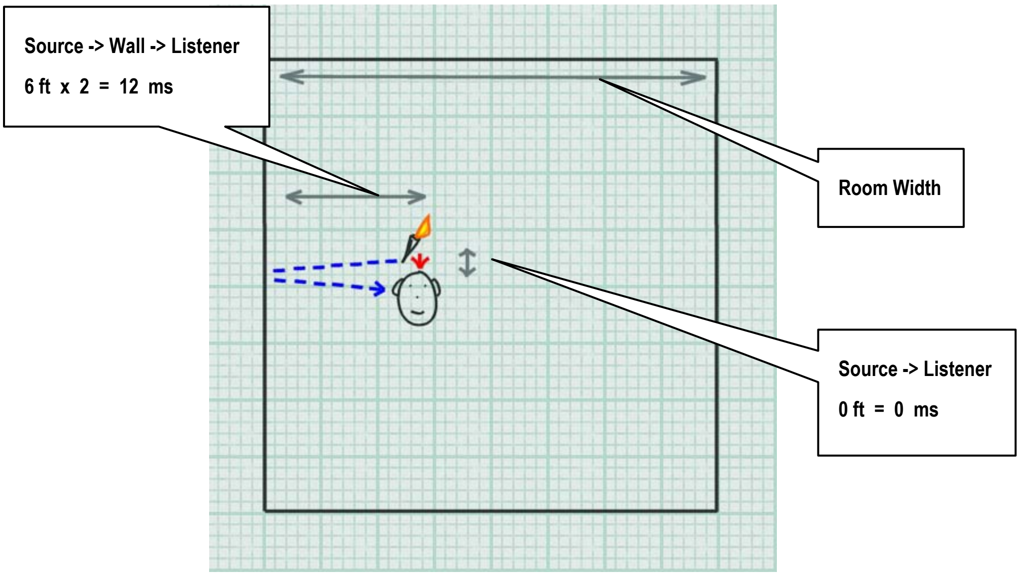

Figure 6 - A typical scenario for calculating Reflection Delay - in this case around 6 milliseconds

The size of the room also has a direct bearing on the time gap between the first reflections and the onset of reverberation. In a small room, reflections will occur more frequently, so any reverberation will build up quicker than in a larger room.

Conclusion

When designing an environmental reverb, making a good estimate of the room's size will help you to approximate a good starting value for Reflections Delay. This in turn will provide a good starting point towards developing values for many of the other EAX environment model properties, including Reverb Delay. These calculations are really *"rules of thumb"* – they can be used to generate suitable values for a general set of situations.

To summarize, the length of delay before a listener hears early reflections (*Reflections Delay*) is directly affected by the distance between the listener and the reflecting surface. In a larger room, a sound is statistically likely to travel further before being reflected back to the listener. Therefore, when designing an environment, assume that *Reflections Delay* will be proportional to room size. If you are working on dynamically adjusting parameters in real time, *Reflections Delay* can be altered according to the distance between the listener and the nearest surface. Reverb becomes prominent quicker in smaller rooms, but the delay between the first early reflections and the onset of reverb (*Reverb Delay)* can be kept independent from the listener's position.

Surface Reflectivity

In order to investigate further properties of environmental reverb, we must take into account another room characteristic – how well the room's walls absorb sound. Some energy is always absorbed when sound strikes a surface. Materials such as metal and concrete are relatively reflective; they absorb less sound energy than materials such as fabric and wood paneling. A room that consists of reflective walls will generally support louder reflections and longer reverberation than a room consisting of absorptive surfaces. Let us look in more detail at how surface material can influence reflections and reverberation.

ROOM

Value range **-10000 to 0**
Default value **-1000**
Value units **Hundredths of a dB**

Room controls the amount of reflected sound in an environment. Changes made to *Room* adjust the level of both the early reflections and the reverberation at the same time. Setting this parameter to –10000 mutes all reflections and reverberant sound in an environment.

Setting the Room value for an environment, or for all environments, is a good way to adjust the overall amount of environmental sound in an application. For instance, a slider could be presented in the application's audio options menu, allowing the user to control the global level of reflected sound. The slider's value could then be applied as a modifier to the *Room* property each time a reverb is applied. However, for independently tweaking and balancing the levels of reflected and reverberant sound within an environment, *Reflections* and *Reverb* should be used.

REFLECTIONS

Value range **-10000 to 1000**
Default value **-2602**
Value units **Hundredths of a dB**

Reflections set the level of the early reflections in an environment. We use early reflections as a cue for determining the size of the environment we are in. The louder (and less delayed) the reflections are the nearer a wall will sound.

REVERB

Value range **-10000 to 2000**
Default value **200**
Value units **Hundredths of a dB**

Reverb controls the level of the reverberant sound in an environment. A high level of reverb is characteristic of rooms with highly reflective walls and/or small dimensions.

DECAY TIME

Value range **0.1 to 20.0**
Default value **1.49**
Value units **Seconds**

Decay Time controls the amount of time the reverberated sound takes to decay 60 dB. Decay time is dependant on both the size of the room and the reflectivity of the walls. Generally speaking, rooms with low absorbance and large size will support a longer reverb decay time.

We have seen how room size affects the time it takes for early reflections to reach the listener. Just like direct path sound, reflected sound is attenuated over distance due to the expansion of the spherical waves and to air absorption (at high frequencies). So the longer the path from source to reflecting wall, and wall to listener, the lower the intensity of perceived early-reflected sounds. This implies that reflections tend to be louder in small rooms. However, path length alone does not determine the intensity of early reflections reaching the listener. Highly absorptive walls will reduce the reflected sound's intensity at each bounce; so early reflections level is a function of path length and surface reflectivity. Reflection intensity is determined by the listener's proximity to a surface, as well as the room's surface absorptivity, so *Reflections* is an ideal candidate for dynamic adjustment.

Reverb, like reflections, tends to be quieter in larger rooms, but for slightly different reasons. The key to overall reverb intensity is the total room volume – in a larger room, the energy emitted by a source must spread over a larger volume. Because reverberation is the combined effect of many instances of reflected sound washing around a room in different directions, its level is more or less the same no matter where the listener is positioned within the room.

Reverb level does vary slightly for each source according to its distance from the listener and to the room's Decay Time. By default, EAX continuously calculates these changes for itself automatically, so the property *Reverb* does not require further dynamic modification. For more details on how EAX works out the attenuation of reflected sound, see the section on Attenuation effects.

Now let us look at the time it takes for reverb to die away – Decay Time. As we have established, sound waves are absorbed when they collide with solid objects, such as a room's walls, and at high frequencies by the air. Therefore, the more often-sound energy collides with absorptive surfaces, the quicker the reverb will decay.

In a bigger room with a larger volume, there will be a longer mean time between each energy-sapping collision of sound with surface. As room dimensions increase, the room's volume increases to a power of three, while the surface area available for absorption just increases to a power of two. The resulting effect is that, if wall absorptivity is kept constant, Decay Time is linearly proportional to the room's size.

Conclusion

A room's absorption coefficient is an indication of how much sound is absorbed by a room's surfaces. Environments with highly reflective walls tend to support louder reflections and longer reverbs. Environments with absorptive walls tend to generate less reflected sound. A large room size implies a greater volume, making the reverb quieter although the decay time is longer.

As the listener moves within an environment, the intensity of early reflections (*Reflections*) varies with the listener's proximity to surfaces. Reverberation intensity (*Reverb*) and reverberation decay (*Decay Time*) remain more or less constant within an environment.

Surface Reflectivity at different frequencies

Up until now, we have assumed that a surface will absorb and reflect sound at a constant level, regardless of the sound's frequency. In fact, materials react differently when struck by sounds of different frequency. Generally, harder and smoother materials such as glass, wood and concrete tend to absorb low frequencies and reflect high frequencies. Softer materials like woven fabrics are reflective of low frequencies, but absorb more high frequency sound.

Room HF

Value range **-10000 to 0**
Default value **-100**
Value units **Hundredths of a dB**

Room HF is used to attenuate the high frequency content of all the reflected sound in an environment. You can use this property to give a room specific spectral characteristic. If you want to model a room that absorbs many high frequencies, simply bring down the value for *Room HF* until you get the timbre you are looking for. *HF Reference* sets the frequency at which the value of this property is measured.

Room LF

Value range **-10000 to 0**
Default value **0**
Value units **Hundredths of a dB**

Room LF is the low frequency counterpart to *Room HF*. Use this to reduce or boost the low frequency content in an environment. *LF Reference* sets the frequency at which the value of this property is measured.

Decay HF Ratio

Value range **0.1 to 2.0**
Default value **0.83**
Value units **A linear multiplier value**

Decay HF Ratio scales the decay time of high frequencies relative to the value of the *Decay Time* property. By changing this value, you are changing the amount of time it takes for the high frequencies to decay compared to the mid frequencies of the reverb. A value of 1.0 means that the high frequency content of the reverb will decay at the same rate as the mid frequencies. Setting this value to 0.5 will cause the high frequencies to decay twice as fast. Setting it to 2.0 means the high frequency content will last twice as long as the mid frequencies. *HF Reference* sets the frequency at which the value of this property is measured.

In most cases, the high frequency content should decay faster than the rest of the reverb (high frequencies are absorbed more easily by most materials and by the air). Increasing *Decay Time* will proportionally increase the high frequency decay time of a reverb. The flag *Decay HF Limit* can be set to cap the high-frequency decay time at a realistic value, determined by the property *Air Absorption HF*.

Decay LF Ratio

Value range **0.1 to 2.0**
Default value **1.0**
Value units **A linear multiplier value**

Decay LF Ratio scales the decay time of low frequencies in the reverberation in the same manner that *Decay HF Ratio* handles high frequencies. *LF Reference* sets the frequency at which the value of this property is measured.

HF Reference

Value range **1000.0 to 20000.0**
Default value **5000.0**
Value units **Hertz**

LF Reference

Value range **20.0 to 1000.0**
Default value **250.0**
Value units **Hertz**

The properties *HF Reference* and *LF Reference* determine respectively the frequencies at which the high-frequency effects and the low-frequency effects created by EAX properties are measured.

Note that, for listener properties, it is necessary to maintain a factor of at least 10 between these two reference frequencies so that low frequency and high frequency properties can be accurately controlled and will produce independent effects. In other words, the *LF Reference* value should be less than 1/10 of the *HF Reference* value.

The properties described above allow you to manage the overall tone of reflected sound, and additionally the tone of the late reverb decay, at the high and low frequencies. What's more, by changing the HF and LF reference frequencies, you can adjust where in the frequency spectrum these controls are effective, similar to a two band parametric equalizer or two swept EQ channels on a mixing console. The HF and LF reference frequencies need to be manipulated with great care. In fact, these parameters will not just modify the frequency spectrum for reflections and reverberation, but they also, as explained later on, greatly influence other EAX parameters such as Air Absorption, Occlusion, and Obstruction. It is therefore advisable to keep the same values of HF and LF reference frequencies across all the Environmental reverb presets in use. This will prevent unwanted changes in Air Absorption, Occlusion, and Obstruction effects when the listener moves from one location to another one.

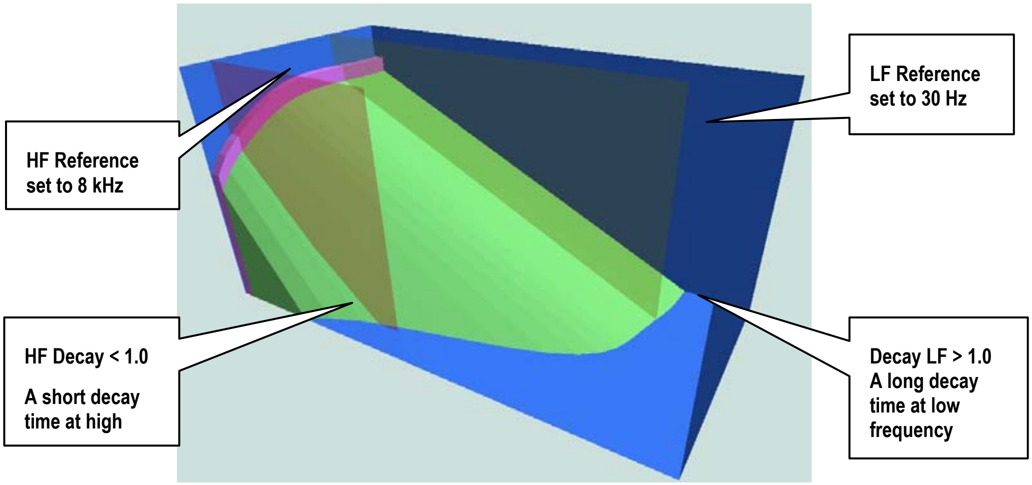

Figure 7 – Three-dimensional reverberation response model from Creative's EAGLE reverb design tool. Frequency is represented in the depth dimension – high (near) to low (far). Note how the reverb decay is longer at lower frequencies in this reverb model.

If you have some idea of how the surfaces of a room react to different frequencies, then you can use this knowledge as a starting point for using the EAX environmental reverb tonal controls. It would be quite involved to work out scientifically accurate values for these settings. So in this case, as with many other aspects of reverb design, it is best to make aesthetic choices, guided by scientific principles:

- Decide the 'feel' you want for the room, guided by the wall materials and the context of the room. Should the room be *'bright'* or *'dark'*? Should the room be *'boomy'* or *'tinny'*?

- Set the parameters to achieve the feel you want. For *'bright'* rooms, boost high frequencies with high values for *Room HF*, and maybe bias the reverb decay towards higher frequencies by setting *Decay HF Ratio* near 1.0 (or even over 1.0). To get rid of *'booming'* at low frequencies, turn down *Room LF* and/or *Decay LF Ratio*.

- Alternatively, if you want a darker, more damped room, get rid of high frequencies with lower settings for *Room HF* and/or putting *Decay HF Ratio* below 1.0.

- Audition some of your application's sounds with your environment applied, and revise the effect as necessary. Of course, it is important to assess how the effects work in relation to the source sounds it will process, and to pay heed to the frequency content of the sound samples when designing effects.

Conclusion

Tonal coloration in reflected sound tells us a lot about the materials that make up our surroundings. Different materials might be particularly absorptive or reflective of certain frequencies. A key task in designing an environmental reverb is to determine a tonal character for the location you want to model. The EAX environmental reverb allows for tonal control over the entire reflected sound (*Room HF, Room LF*). You can also set how quickly the late reverb decays at high and low frequencies (*Decay HF, Decay LF*) in comparison to the decay time at mid frequency (*Decay*). You can vary the frequencies at which these controls are effective (*HF Reference, LF* Reference). Spectral effects are unlikely to be suitable for dynamic adjustment.

Wall configuration

The arrangement of the surfaces in an environment can greatly affect the granularity of the reverberation.

ENVIRONMENT DIFFUSION

Value range	**0.0 to 1.0**
Default value	**1.0**
Value units	**A linear multiplier value**

Environment Diffusion controls how the individual reflections in the reverb are distributed. If the value for this property is high, then there is a richer pattern of reflected waves in the room and therefore a greater number of reflections will reach the listener. This results in a smooth sounding reverberation. This type of environment is said to be "diffuse." Rooms that have small dimensions or very uneven, coarse surfaces or contain many reflective obstacles often exhibit this type of effect.

Rooms or enclosures that have large dimensions and smooth surfaces, or open environments with sparse reflectors, produce fewer reflections patterns than diffuse environments. Such an environment will therefore require a lower value for *Environment Diffusion*. (Smoother surfaces do not "break the reverb up," and therefore, there are fewer echoes.)

ECHO DEPTH

Value range **0.0 to 1.0**
Default value **0.0**
Value units **A linear multiplier value**

ECHO TIME

Value range **0.075 to 0.25**
Default value **0.25**
Value units **Seconds**

Echo Depth introduces a cyclic echo in the reverberation decay, which will be noticeable with transient or percussive sounds. A larger value of Echo Depth will make this effect more prominent. *Echo Time* controls the rate at which the cyclic echo repeats itself along the reverberation decay. For example, the default setting for *Echo Time* is 250 ms. causing the echo to occur 4 times per second. Therefore, if you were to clap your hands in this type of environment, you will hear four repetitions of clap per second.

Together with *Environment Diffusion*, *Echo Depth* will control how long the echo effect will persist along the reverberation decay. In a more diffuse environment, echoes will wash out more quickly after the direct sound. In an environment that is less diffuse, you will be able to hear a larger number of repetitions of the echo, which will wash out later in the reverberation decay. If *Environment Diffusion* is set to 0.0 and *Echo Depth* is set to 1.0, the echo will persist distinctly until the end of the reverberation decay.

In an environment with many rough surfaces, sound will be reflected in countless directions, greatly increasing the chance that a reflection will reach the listener's ears. This will result in a smooth reverberation.

However, in environments with few diffuse surfaces - bare rooms with smooth walls - you are more likely to hear distinct echoes, resulting in a more grainy reverberation sound. Two parallel, smooth surfaces in a room can cause sound energy to bounce backwards and forwards, making a 'fluttering' echo effect. EAX 4.0's Diffusion and Echo parameters allow the sound designer to introduce discrete echoes into the reverberated signal, simulating this phenomenon. This effect can be more easily heard with a percussive source sound such as a pistol shot or a drum hit.

Once again, the environment size will be an influence on the reverb. In this case, the length of the path between the walls will determine the time it takes the wave front to traverse the room. With our rough figure of one millisecond per foot, we can calculate that under the appropriate conditions, a listener in a room containing two smooth, parallel surfaces 20 feet apart, might hear a discrete echo every 20 milliseconds. Furthermore, the echo intensity will be dependant on the wall absorptivity.

Small Rooms

As described above, fluttering echoes are heard in an environment where reflections have a low temporal density. However, there are also situations where environmental sound is also unevenly distributed in the frequency domain. At the root of this phenomenon is the concept of cyclic echoes.

The wall geometry of a room can cause some reflections to take repetitive or cyclic paths. This occurs particularly in small, reverberant rooms. Figure 8 shows a cyclic echo in a square room.

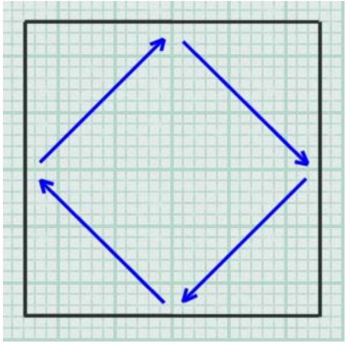

Figure 8 - Cyclic echoes in a room

In this situation, a resonance or *'standing wave'* will be set up in the room, at a frequency corresponding to the length of the cyclic echo's path. The frequencies of the different standing waves in a room are known as the room's modal frequencies.

In larger environments where the distribution of modal frequencies tends to be denser, no particular resonance stands out. But smaller, relatively reverberant rooms such as bathrooms tend to have sparser modal frequencies, giving rise to a more hollow tone.

Although there is no property explicitly defined in EAX 4.0 to modify the density of the modal frequencies in your reverb, there is a special technique, here described, that can be used instead. The section on Room Size shows how the *Environment Size* property controls a range of low-level parameters, with flags available so you can turn off the automatic scaling per parameter. As you move *Environment Size* below 2.0 the modal density will also gradually decrease, regardless of the flag settings. So if you want to simply tweak just the modal density setting, make sure all the scaling flags are set to off, and then experiment with *Environment Size* values between 1.0 and 2.0.

Static DependenciesTable 1, below, shows how room characteristics influence environmental reverb properties. The right hand column simply shows which room features you should consider when tweaking the different EAX 4.0 reverb parameters to design a static acoustic simulation for a room.

EAX Parameter	**Room Characteristic**
Reflection Delay	Room Size (Implying path length)
Reverb Delay	Room Size (Implying path length)
Reflections	Room Size (Implying path length), Room Absorptivity
Reverb	Room Size (Implying surface area), Room Absorptivity
Decay Time	Room Size, Room Absorptivity
HF Reference LF Reference	Absorptivity of surfaces at different frequencies
Room HF Room LF	Absorptivity of surfaces at different frequencies
Decay HF Ratio Decay LF Ratio	Absorptivity of surfaces at different frequencies
Environment Diffusion	Presence of smooth surfaces, sparseness of reflectors
Echo Depth	Presence of reflective parallel surfaces
Echo Time	Room Size (Implying path length)

Table 1 - Showing room characteristic to EAX property dependencies for designing environmental reverbs.

Dynamic Modeling

The physical modeling paradigm has the potential to offer a high quality dynamic acoustic model, particularly in respect of early reflections. However, as we discussed earlier (see *Approaches to environmental audio*), physical modeling implementations have difficulty tracing reflections far enough to realistically re-create the subtleties of late reverb.

To physically model sophisticated phenomena like tonal variation (see *Surface Reflectivity at different frequencies*), echo density (see *Wall configuration*) and modal density (see *Small Rooms*) would require prohibitive amounts of processing power. Possibly the optimum solution, then, is to employ a hybrid system. The idea is to create statistical models simulating life-like late reverberation for each discrete room or zone in an application, and to manipulate the early reflections properties according to the precise relationship between the listener and the walls for added realism.

Localizing Reflections and Reverb

As described in the EAX4.0 Introduction and the EAX Programmer Guide, making a multi-environment audio engine with EAX 4.0 involves dynamically panning the Reflections and the Reverb.

Knowing a little bit about the architecture of a room, as well as the position and orientation of the listener, it is possible for a program to calculate and set the direction from which the early reflections in the listener's environment tend to come. (For instance, by considering the positions of the nearest walls relative to the listener).

In multiple environment implementations, it is vital that reflected sound from environments other than the listener's is correctly localized. If the programmer does not carry out these calculations, then the realism of the multiple environments will severely compromised. *Reflections Pan* and *Reverb Pan* should be set to imply the distance and direction between the listener and another environment. *Reflections* and *Reverb* can be used to further attenuate reflected sound from distant environments.

Reflections Pan

Value range **Vector of length 0 to 1**
Default value **(0, 0, 0)**
Value units **N/A**

The *Reflections Pan* property is a 3D vector that controls the spatial distribution of the cluster of early reflections. The direction of this vector controls the global direction of the reflections, while its magnitude controls how focused the reflections are towards this direction.

In using panning, it is important to note that the direction of the vector is interpreted in the coordinate system of the user, without taking into account the orientation of the virtual listener. For instance, under DirectSound and assuming a four-point loudspeaker playback system, setting *Reflections Pan* to (0., 0., 0.7) means that the reflections are panned to the front speaker pair, whereas as setting of (0., 0., –0.7) pans the reflections towards the rear speakers. These vectors follow the same left-handed co-ordinate system as DirectSound – if you are using OpenAL, you will need to make allowances for the fact that this API uses a right-handed co-ordinate system.

If the magnitude of Reflections Pan is zero (the default setting), the early reflections come evenly from all directions. As the magnitude increases, the reflections become more focused in the direction pointed to by the vector. A magnitude of 1.0 would represent the extreme case, where all reflections come from a single direction, and is therefore unusual.

Reverb Pan

Value range **Vector of length 0 to 1**
Default value **(0, 0, 0)**
Value units **N/A**

Reverb Pan does for the Reverb what *Reflections Pan* does for the Reflections.

Dynamic Dependencies

Table 2 - shows the EAX parameters that are most suitable for real-time adjustment according to listener position, and shows the variables their value would depend from.

EAX Parameter	Listener / Wall relationship
Reflections Delay	Listener to reflecting surface path length
Reflections	Listener to reflecting surface path length
Reflection Pan	Listener to reflecting surface direction, surface size/shape

Table 2 - Showing room characteristic to EAX reflection property dependencies for dynamically updating parameters

Additional Properties

3D Effect Controls

While DirectSound and OpenAL allow you to manage how a sound source's direct path will be attenuated over distance, these APIs have no facility for simulating a more subtle effect of air absorption. In normal conditions, our atmosphere absorbs sound slightly more at high frequencies. This is why distant sounds often sound duller. EAX

4.0 accounts for this phenomenon, and allows you to adjust the level of high frequency attenuation.

AIR ABSORPTION HF

Value range **-100.0 to 0.0**
Default value **-5.0**
Value units **Hundredths of a dB per meter**

Air Absorption HF is used to simulate environments containing propagation mediums that have different levels of sound absorption, particularly at high frequencies. In typical conditions of atmospheric humidity and temperature, air attenuates high frequency sound at approximately 0.05 dB per meter at 5 kHz. Moist or ashy air may slightly muffle sounds traveling through it – use a lower value. Set a higher level for a less absorptive medium such as dry, desert air. The environmental preset's *HF Reference* setting determines the frequency at which the value of this property is measured (set at 5 kHz by default).

If the flag *Decay HF Limit* is set true (its default setting), *Air absorption HF* can also be used to reduce the high-frequency decay time (overriding the setting of Decay HF Ratio).

The flexible EAX 4.0 multi-effect architecture means that the strategy for setting *Air Absorption HF* is a little more involved than it was with the previous EAX API. As mentioned in the EAX 4.0 Introduction, the Primary effect slot ID property, located within the Context object, allows the programmer to inform the EAX 4.0 engine which one of the four effect slots is currently rendering the environment in which the listener is located. If the effect slot indicated by the Primary effect slot ID is rendering environmental reverb, and is flagged as environmental, then the amount of air absorption at high frequencies is determined by the reverb settings in that effect slot.

Attenuation effects

The phenomenon of rolloff on audio is clear – the further you get from a sound source, the quieter it gets. It is also worth considering that the same applies to that sound source's reflected path, although to a lesser degree than its direct path. According to statistical room acoustics theory, the precise rate at which reverb rolls off tends to be roughly proportional to the room's decay time; In rooms with shorter reverb decays, the reflected sound rolls off quicker. Since the decay time is usually shorter at high frequencies, the reverb is subject to a natural-sounding (usually subtle) distance-dependent low-pass filtering effect, in addition to the effect of air absorption (which applies to both the direct and reflected paths).

ROOM ROLLOFF FACTOR

Value range **0.0 to 10.0**
Default value **0.0**
Value units **A linear multiplier value**

The intensity of a sound source's reflections becomes attenuated with distance in a similar way to its direct path, although not as fast. By default the EAX engine calculates the roll-off on the Room path (reflected sound, including reflections and reverb), taking into account such properties as *Decay Time.*

To define your own room roll-off calculation, change *Room Rolloff Factor* from its default of 0.0. Setting a value of 1.0 for *Room Rolloff Factor* will mean that a sound source's reflected sound is attenuated by distance at the same rate as its direct path sound. Setting *Room Rolloff Factor* to 0.5 instead would imply that the reflected sound is attenuated with increasing distance, although not as strongly as the direct path.

Under normal circumstances, you do not need to worry about this effect, as EAX automatically calculates rolloff on reflected sound for each sound source, bearing in mind the distance between the source and the listener, and characteristics of the environment. However, if the situation arises where you think that distant sounds in an environment are accompanied by too much reverb, or too little, you can make adjustments for yourself with Room Rolloff Factor.

Retain the default value of 0.0 and EAX will make the room rolloff calculations. A value of 1.0 means that a source's reflected sound will be attenuated at the same rate as its direct sound, with respect to distance.

To calculate the statistical reverb rolloff and the air absorption effect, the EAX engine assumes that the distance unit in use is the meter. If this is not the case, it is important to set accordingly the distance factor in Direct Sound or the context distance factor in OpenAL to ensure your unit is correctly converted into meters by the EAX engine. For further details about the distance factor subject and its implications please refer to the section on *Distance Units* in the *EAX 4.0 Programmer's Guide.*

Pitch modulation effects

As well as providing amplitude modulation effects (in the form of a tune-able repeating echo), EAX 4.0 allows the designer to introduce pitch modulation in the reverberation decay. While this effect is probably not encountered in many natural environments, it is useful when designing a reverberation effect that signifies emotional state rather than location in an application.

MODULATION DEPTH

Value range	**0.0 to 1.0**
Default value	**0.0**
Value units	**A linear multiplier value**

MODULATION TIME

Value range	**0.04 to 4.0**
Default value	**0.25**
Value units	**Seconds**

Using these two properties, you can create a pitch modulation in the reverberant sound. This will be most noticeable applied to sources that have tonal color or pitch. You can use this to make some trippy effects! *Modulation Time* controls the speed of the vibrato (rate of periodic changes in pitch).

Modulation Depth controls the amount of pitch change. Low values of Diffusion will contribute to reinforcing the perceived effect by reducing the mixing of overlapping reflections in the reverberation decay.

Designers are encouraged to use these parameters, maybe in combination with extreme settings for other reverberation properties, in creating effects that reinforce the experiences of an avatar in a game or other application. Examples include dizziness, intoxication, or high stress.

Designing Environmental Filtering effects

Environmental filtering is a fundamental part of the positional audio experience. Sound sources that are hidden behind a pillar or a wall are perceived very differently from sound sources that have an un-obstructed path to the listener's ears. Without these important audio cues, the virtual 3D world is an artificial and unconvincing place. What's more, compared to creating your own environmental reverb presets, designing filtering effects is a breeze. So with the knowledge and tips you'll pick up in this section, there's no excuse for designing a positional sound implementation without realistic environmental filtering any more!

Using Environmental filtering effects

Environmental filtering effects can be applied on a per-source basis to simulate muffling on sound sources that are hidden from the listener behind walls, partitions, or obstacles. As described previously in the *Filtering effects* section, EAX 4.0 defines three different filtering scenarios: Obstruction, Occlusion, and Exclusion. Each situation can be simulated by filtering and attenuating a specific combination of direct and reflected sound, as shown in the following table:

	Direct Path	**Reflected sound in listener environment**
Obstruction	Filtered and attenuated	-
Occlusion	Filtered and attenuated	Filtered and attenuated
Exclusion	-	Filtered and attenuated

Table 3 - Filtering and attenuation of direct and reflected sound paths for different sound muffling scenarios

EAX 4.0 gives the developer close control over the amount of filtering and attenuation that takes place on every 3D sound source. Whenever transmission through an obstacle is involved, the amount of filtering and attenuation should be determined by the material through which the sound is passing.

Environment Filtering and the primary environment

In a multi-environment implementation where each source can feed more than one effect slot, it is essential to inform the EAX 4.0 engine that one of the four effect slots is currently rendering the environment in which the listener is located. Once again, the context object comes into rescue. The Primary effect slot ID must be correctly set, so that each source that is occluded or excluded has filtering and attenuation applied to the correct effect send.

Determining the reference frequency for spectral source effects

Some of the environmental filtering effect properties, including *Obstruction*, *Occlusion,* and *Exclusion*, make tonal adjustments at a frequency specified by a high frequency reference value. In a single environmental reverb situation, like with EAX 3.0, determining the reference value for these effects was simple - the source effects referred to the value of *HF Reference* of the only reverb effect that was present. However, EAX 4.0's flexible effect slot mechanism introduces a potential problem. There may be between zero and three environmental reverbs at any one time, each with potentially a different *HF Reference* setting (although as previously mentioned in *Surface Reflectivity at different frequencies*, it is recommended to use the same value across all the Environmental reverb presets in use.)

How does EAX determine the *HF Reference* under each circumstance? Again, the key is in the Context Object. If the effect slot indicated by the Primary effect slot ID is flagged as an environmental slot and is rendering the EAX environmental reverb, then all source direct-path effects use that effect's HF Reference value. If these conditions are not met, then source direct-path effects use the value of *HF Reference* specified in EAX 4.0's context object.

Obstruction Parameters

OBSTRUCTION

Value range **-10000 to 0**
Default value **0**
Value units **Hundredths of a dB**

Obstruction occurs when there is an object blocking the direct path from the sound source to the listener. The direct path will be muffled but the reflected sound will not (see Sound Obstruction).

The *Obstruction* parameter determines the attenuation applied to the direct path at the reference high frequency (you set this frequency using the EAX Environmental Reverb / Context property *HF Reference*, see Determining the reference frequency for spectral source effects). You can set the value of the *Obstruction* property for each sound source. If the *Obstruction LF Ratio* property (described below) is set to 0.0, *Obstruction* controls only attenuation at high frequencies. If *Obstruction LF Ratio* is set above 0.0, *Obstruction* also attenuates low frequencies to the extent specified by *Obstruction LF Ratio*.

Obstruction's maximum value, 0, specifies no attenuation and hence no obstruction effect. The minimum value, –10000 (which is -100 dB), indicates that the sound source is so obstructed that the direct path from source to listener is negligible — so only the source's reflected sound is audible. Any value between minimum and maximum indicates partial obstruction.

Note that you can use *Obstruction* and *Occlusion* simultaneously – if, for example, the source is in another room from the listener and there is a large obstacle between the listener and the wall. In this case, the direct path is filtered twice: once by *Occlusion* and once by *Obstruction*.

OBSTRUCTION LF RATIO

Value range **0.0 to 1.0**
Default value **0.0**
Value units **A linear multiplier value**

Obstruction LF Ratio specifies the obstruction attenuation at low frequencies relative to the attenuation at high frequencies defined by the *Obstruction* property. The minimum value of 0.0 (the default value) specifies no attenuation at low frequencies; the maximum value of 1.0 specifies the same low-frequency attenuation as high-frequency attenuation. Note that adjusting *Obstruction LF Ratio* alone has no effect if *Obstruction* is set to 0.

In the *Sound Obstruction* section, we discussed two ways in which obstacles can affect sound – transmission and diffraction.

In the case of *transmission* through an obstacle, an effective way for sound designers to use Obstruction (as well as Occlusion) is to define a material preset. This will comprise a setting for *Obstruction* and for *Obstruction LF Ratio*. Material presets will be assigned to physical objects in the application's world. The audio engine should then apply the object material preset to every sound source that is obstructed by that object.

In the case of *diffraction* around an obstacle, the audio engine should determine the shortest propagation path from source to listener and set the Obstruction value according to the angle of deviation undergone by this shortest path. For an accurate simulation, the application should also set the positional parameters of the source to the corresponding apparent position relative to the listener.

Occlusion Parameters

OCCLUSION

Value range **-10000 to 0**
Default value **0**
Value units **Hundredths of a dB**

Occlusion occurs when the listener is in one room or environment, the sound source is in another room or environment, and the listener hears the sound through a separating wall or through a closed door or window (See Sound Occlusion). Occlusion muffles the direct and reflected sound of a sound source. The value set for the *Occlusion* property determines the attenuation at the reference high frequency.

The effect of occlusion depends a great deal on the sound transmission qualities of the material separating the two rooms. Some materials are thick and absorbent and transmit very little sound; others are stiff and thin and transmit clearly; others have transmission qualities in between. Frequency response varies too. Some materials attenuate high frequencies more than others do.

Occlusion's maximum value, 0, specifies no attenuation and hence no occlusion effect. The minimum value, –10000 (which is -100 dB), indicates that the sound source is so occluded that it is barely audible. Any value between minimum and maximum indicates partial occlusion.

OCCLUSION LF RATIO

Value range **0.0 to 1.0**
Default value **0.25**
Value units **A linear multiplier value**

The *Occlusion LF Ratio* specifies the occlusion attenuation at low frequencies relative to the attenuation at high frequencies. The minimum value of 0.0 specifies no attenuation at low frequencies; the maximum value of 1.0 specifies the same low-frequency attenuation as high-frequency attenuation. The default setting of 0.25 specifies that low frequencies be attenuated much less than high frequencies. Note that adjusting *Occlusion LF Ratio* alone has no effect if *Occlusion* is set to 0.

Occlusion Room Ratio

Value range **0.0 to 10.0**
Default value **1.5**
Value units **A linear multiplier value**

Occlusion Direct Ratio

Value range **0.0 to 10.0**
Default value **1.0**
Value units **A linear multiplier value**

The *Occlusion Room Ratio* and *Occlusion Direct Ratio* properties control the amount of attenuation and filtering applied to the reflected sound and the direct path, respectively, for a given setting of the *Occlusion* property. The attenuation obtained at the reference high frequency is determined by multiplying the *Occlusion* property value by the *Occlusion Room Ratio* or the *Occlusion Direct Ratio*.

When both values are set to 0.0, the *Occlusion* property has no effect. If *Occlusion Room Ratio* is set to 0.0 and *Occlusion Direct Ratio* to 1.0, Occlusion is equivalent to Obstruction. If *Occlusion Room Ratio* is set to 1.0 and *Occlusion Direct Ratio* to 0.0, Occlusion is equivalent to *Exclusion*.

When the Ratio value is set between 0.0 and 1.0, the effect is equivalent, for low and high frequencies, to scaling the setting of the *Occlusion* property by that value. If the value is larger than 1.0, that is only true at high frequencies. At low frequencies, the attenuation is such that the filter slope (difference between low frequencies and high frequencies) remains the same as for a setting of 1.0.

The default settings — 1.0 for *Occlusion Direct Ratio* and 1.5 for *Occlusion Room Ratio* — specify that, compared to the direct sound path, the reflected sound undergoes an additional frequency-independent attenuation equal to half the setting of *Occlusion*. This creates a natural sensation of occlusion because, in the physical world, it is the occluding wall that acts as the actual sound source in the listener's room. Since the wall radiates sound in only half the space that a sound source in the middle of the room can, it generates significantly less reflected sound than the original source would, if it were located in the room (if *Occlusion Room Ratio* is set to 1.0, the perceived effect might be more like that of a sound source located in the same room as the listener, but wrapped in a box or container).

Exclusion Parameters

Exclusion

Value range **-10000 to 0**
Default value **0**
Value units **Hundreds of a dB**

Exclusion LF Ratio

Value range **0.0 to 1.0**
Default value **1.0**
Value units **A linear multiplier value**

The Exclusion and Exclusion LF Ratio properties are defined exactly as the corresponding *Obstruction* and *Obstruction LF Ratio* properties described above, except that they apply an attenuation and filtering effect to the reflected sound instead of the direct path. This is used to simulate sound in another room coming through an opening in the partition (see Sound Exclusion). The value set for the Exclusion property determines the attenuation applied to the reflected sound at the reference high frequency, while the direct path sound is left unaffected

Exclusion's maximum value, 0, specifies no attenuation of the reflected sound. The minimum value, -10000 (which is -100 dB), indicates that the reflected sound is barely audible. Any value between minimum and maximum indicates partial exclusion.

The *Exclusion LF Ratio* property specifies the attenuation at low frequencies relative to the attenuation at high frequencies. The maximum value of 1.0 (the default value) specifies a frequency-independent attenuation. The minimum value of 0.0 specifies no attenuation at low frequencies. Note that adjusting *Exclusion LF Ratio* alone has no effect if *Exclusion* is set to 0.

The *Exclusion* property can be used when the sound source is not in the same room as the listener but its direct path can reach the listener through an opening (window, door). In such a situation, the sound source can only contribute a limited amount of energy to the reverberation of the listener's room (especially if it is distant from the aperture). Therefore, a low setting of the *Exclusion* property would be used. As the source comes closer to entering the room, the *Exclusion* value can be increased to apply more reverberation to this source. Note that you can use *Obstruction* simultaneously with *Exclusion* if the source is not directly visible through the aperture. In this way, some muffling can be applied to the direct path to reproduce the effect of diffraction around the edge of the aperture.

Applying muffling effects in real time

To correctly simulate obstruction, occlusion, and exclusion, the audio engine is likely to check several times every second, to see if any of the situations apply to any of the sources currently playing.

Parameters for simulating obstruction, occlusion, or exclusion on a sound source may well vary as the positions of the source and listener move in relation to their surroundings. In the case of obstruction, the amount of attenuation is likely to reduce as the sound source clears the obstruction. In the case of exclusion, the amount of reverberation heard is likely to increase as the source approaches the aperture. Although the task of calculating these adjustments should be carried out in the application, it is likely that the sound designer will have a hand in defining the policies used.

Additional source specific EAX enhancements

APIs like DirectSound3D and OpenAL provide the application developer with all the basic facilities for positioning sounds in 3D. As well as adding environmental effects like reverberation and filtering, EAX is also capable of extending the positional basics such as roll-off and directivity with some subtle but useful extra enhancements. This section details how you can use EAX to make the 3D soundtrack more convincing, even before you reach for the reverb and filtering effects. It also deals with properties that allow you to balance the overall amount of direct and reflected sound on each sound source.

3D Source Controls

The following properties provide automatic methods for making per-source direct and reflected volume adjustments according to the positional parameters of the sound source or the listener. They extend the rolloff effect, Doppler effects, and directivity effects provided by the positional audio APIs. Note that, as one would expect, these properties have no effect if DirectSound's 3D effects are disabled on the sound source.

ROLLOFF FACTOR

Value range	**0.0 to 10.0**
Default value	**0.0**
Value units	**A linear multiplier value**

Both DirectSound and OpenAL provide a Rolloff Factor property. However, OpenAL's Rolloff Factor is a per-source property while DirectSound's is a global property.

The EAX Rolloff Factor is a per-source property (and is therefore equivalent to OpenAL's Rolloff Factor). If the Rolloff Factor already provided by the positional audio API (DirectSound or OpenAL) is non-zero, then the EAX Rolloff Factor will be added to it to determine the rolloff applied to the sound source.

Essentially, this brings DirectSound3D programmers the extra flexibility to adjust rolloff factor independently on each source.

ROOM ROLLOFF FACTOR

Value range	**0.0 to 10.0**
Default value	**0.0**
Value units	**A linear multiplier value**

This is the per-source equivalent of the EAX Reverb property of the same name. This property only affects one specific sound source. If the EAX Reverb property *Room Rolloff Factor* is non-zero, then the value for these two properties will be added together to determine the rolloff applied to the sound source.

This parameter brings you the ability to change how quickly an individual source's reflected sound will roll off as listener to source distance increases.

Doppler Factor

Value range **0.0 to 10.0**
Default Value **1.0**
Value Units **A linear multiplier value**

This low-level per-source property is defined in the same way as the global Doppler Factor property provided in OpenAL and DirectSound. It is important to keep in mind the source Doppler factor is a multiplier of the global Doppler factor property provided in OpenAL and DirectSound.

A zero value will therefore disable Doppler shift effects for the corresponding source. A value of 1.0 provides natural Doppler effects according to the movement of the source relative to the listener. A value larger then 1.0 will exaggerate these effects.

Air Absorption Factor

Value range **0.0 to 10.0**
Default value **0.0**
Value units **A linear multiplier value**

The Air Absorption Factor property is a multiplier value for the air absorption value set by the Reverb or Context property Air Absorption HF. The resultant air absorption value applies only to this sound source.

The Air Absorption Factor default value is 0.0. This means that the overall Air absorption effect for any source is off by default. A value of 1.0 will correspond to air absorption equal to the value set by the listener Air Absorption HF property.

You can use the Air Absorption Factor to simulate a source located in different atmospheric conditions than the rest of the room. You can increase air absorption, for example, for a sound source that comes from the middle of a cloud of smoke. Alternatively, you can decrease air absorption for a sound source coming from a suddenly visible object in moving clouds.

OUTSIDE VOLUME HF

Value range **-10000 to 0**
Default value **0**
Value units **Hundredths of a dB**

The Outside Volume HF property enhances the directivity effect provided by the positional audio API for individual sound sources. Using standard directivity properties, a source can be made to sound at full volume when the listener is directly in front of the source and will be attenuated as the listener circles the source away from the front.

When DirectSound3D or OpenAL attenuates a source's direct-path sound to simulate directivity, they attenuate high and low frequency sound equally. Real world sources tend to be more directional at high frequencies than at low frequencies.

The Outside Volume HF property allows the developer to enhance the directivity effect by attenuating the high frequencies more than the low frequencies. At the minimum (and default) setting of 0, there is no additional high-frequency attenuation, so the directivity effect is unaltered. A setting of –1000 (-10 dB) means that high frequencies are attenuated by an additional 10 dB on the direct-path sound when the listener is located in the back of the source (anywhere outside of the "Outside Cone"). If the Room Auto flag is set true (its default value), EAX reflections, and reverberation will also be attenuated somewhat at high frequencies (because the source will not radiate as much energy into the room at high frequencies).

You can turn off the effect of Outside Volume HF on the direct-path sound using the *Direct HF Auto* flag, or you can turn off its effect on reflected sound using the *Room HF Auto* flag.

Flags

These flags determine whether you want the EAX 4.0 engine to automatically adjust certain parameters of direct path or reverberation and reflections for a source according to the setting of its position and directivity parameters. They are set to `TRUE` by default to provide a more realistic listening experience without any programming work.

DIRECT HF AUTO

If this flag is `TRUE` (its default value), this source's direct-path sound is automatically filtered according to the orientation of the source relative to the listener and the setting of the sound-source property Outside Volume HF.

If Outside Volume HF is set to 0, the source is not more directive at high frequencies and this flag has no effect. Otherwise, the direct path will be brighter in front of the source than on the side or in the rear.

If this flag is `FALSE`, this sound source's direct-path sound is not filtered at all according to orientation. (Note that this is not the same as setting Outside Volume HF to 0, because this flag doesn't affect high-frequency attenuation of each source's reflected sound according to its directivity – that is controlled by Room HF Auto).

ROOM HF AUTO

When this flag is `TRUE` (its default value), the high frequency content of the reflected sound will be automatically reduced according to the Outside Volume HF property and cone angle settings. When Outside Volume HF is set to a value other than 0, it is in effect taking high frequencies out of the over all sound radiated by the source into the room. The more directive a sound is, and the higher the Outside Volume HF value is, the more high frequencies will be taken out.

If this flag is `FALSE`, the sound source's reflected sound is not filtered at all according to the source's directivity. Note that this is not the same as setting Outside Volume to 0, because this flag does not affect high-frequency attenuation of each source's direct-path sound according to its directivity – that is controlled by the Direct HF Auto flag.

Room Auto

If this flag is `TRUE` (its default value), the intensity of this sound source's reflected sound is automatically attenuated according to source-listener distance and source directivity (defined by Cone parameters and Outside Volume). If it is `FALSE`, the reflected sound is not attenuated according to distance and directivity.

Adjusting source levels independently

EAX provides four per-source volume adjustment properties: *Direct*, *Direct HF*, *Room,* and *Room HF*. The effects of these adjustments are independent of the positional parameters of the sound source or the listener. They can be regarded as extensions of the basic per-source volume control parameter provided by the positional audio APIs (*Volume* in DirectSound3D, *Gain* in OpenAL).

Direct

Value range	**-10000 to 1000**
Default value	**0**
Value units	**Hundredths of a dB**

This is the level of the direct sound path. *Direct* allows you to make manual adjustments to an individual sound source's direct-path volume after all other forms of attenuation are applied (attenuation from distance, orientation etc.). You can use this to make additional corrections to sound source levels if they are not what you want them to be after all other attenuation effects are applied.

Direct HF

Value range	**-10000 to 0**
Default value	**0**
Value units	**Hundredths of a dB**

The *Direct HF* property works the same as the *Direct* property except that it provides an additional attenuation at high frequencies only (measured at *HF Reference*).

Room

Value range **-10000 to 1000**
Default value **0**
Value units **Hundredths of a dB**

Room HF

Value range **-10000 to 0**
Default value **0**
Value units **Hundredths of a dB**

Source *Room* and *Room HF* work just like *Direct* and *Direct HF,* but act on all environmental effect paths, not the direct path. These properties are present so that you can make additional changes to the level of the reflected sound and to its high-frequency attenuation on a source-by-source basis.

Studio-style EAX effects

So far, the Designers Guide has concentrated on environmental enhancements to positional sound. However, EAX 4.0 also offers hardware acceleration of some other effect types, which have a variety of potential uses.

Using Studio-style effects

The studio-style effects which are available in the current version of the EAX 4.0 SDK all output in stereo to the front speakers (panning is not controllable). The effect algorithms are designed by E-MU systems for professional audio use, so they are of extremely high quality. You can load an effect type into each non-locked EAX 4.0 effect slot. That means, on the current hardware, studio-style effects can be freely loaded only into slots 2 and 3, whereas in slots 0 and 1 there is always loaded reverb and chorus effect respectively.

One important note – many of the studio effects provided in EAX 4.0 tend to be most effective when used as 'insert' effects. This means that the 'wet' path, or treated sound, completely replaces the 'dry' path, or untreated sound, in the channel. This is in contrast to the standard way effects work in EAX 4.0, whereby the dry and wet paths are combined when the 3D audio scene is mixed. In order to emulate this behavior with EAX 4.0, ensure that the *Direct* property is turned down on any source you wish to process through such effects, so that only processed sound is sent to the mix. (This applies typically to the following effects: AGC Compressor, Auto-Wah, Distortion, Equalizer, Frequency Shifter, Pitch Shifter, Ring Modulator, and Vocal Morpher).

Normally, effect slots hosting studio-style effects (except perhaps for Echo) should be flagged as "non-environmental" (by setting the Environmental flag false in the FXSlot object). That provision will ensure that the wet/dry mix (set via the Direct/DirectHF, and Send/SendHF properties for each source) is preserved irrespective of the source and listener positions or orientations.

The EAX 4.0 effect types

AGC Compressor

The Automatic Gain Control effect performs the same task as a studio compressor – evening out the audio dynamic range of an input sound. This results in audio exhibiting smaller variation in intensity between the loudest and quietest portions. The AGC Compressor will boost quieter portions of the audio, while louder portions will stay the same or may even be reduced

AGC Compressor On/ Off

Value range	**0 to1**
Default value	**1**
Value units	**0 (off), 1 (on)**

The EAX 4.0 AGC Compressor effect can only be switched on and off – it cannot be adjusted.

The effect may be useful when the application needs to treat the level of a sound that is generated by an outside source. For example, an application might receive external audio streamed from a microphone input, from another program running on the system, or from another system via network communication. The AGC Compressor can be employed to even any troughs and peaks in the sound before it is played back to the user.

Auto-Wah

The Auto-wah effect emulates the sound of a wah-wah pedal used with an electric guitar, or a mute on a brass instrument. Such effects allow a musician to control the tone of their instrument by varying the point at which high frequencies are cut off. The EAX 4.0 effect is *Auto*-wah because there is no user input for modulating the cut-off point. Instead, the effect is achieved by analyzing the input signal, and applying a band-pass filter according the intensity of the incoming audio.

AUTO-WAH ATTACK TIME

Value range **0.0001 to 1.0**
Default value **0.06**
Value units **Seconds**

This property controls the time the filtering effect takes to sweep from minimum to maximum center frequency when it is triggered by input signal.

AUTO-WAH RELEASE TIME

Value range **0.0001 to 1.0**
Default value **0.06**
Value units **Seconds**

This property controls the time the filtering effect takes to sweep from maximum back to base center frequency, when the input signal ends.

AUTO-WAH RESONANCE

Value range **600 to 6000**
Default value **6000**
Value units **mB (milli-bel = 1/100th decibel)**

This property controls the resonant peak, sometimes known as *emphasis* or *Q,* of the auto-wah band-pass filter. Resonance occurs when the effect boosts the frequency content of the sound around the point at which the filter is working. A high value promotes a highly resonant, *sharp* sounding effect.

AUTO-WAH PEAK LEVEL

Value range **-9000 to 9000**
Default value **2100**
Value units **mB (milli-bel = 1/100th decibel)**

This property controls the input signal level at which the band-pass filter will be fully opened.

Chorus

The chorus effect essentially replays the input audio accompanied by another slightly delayed version of the signal, creating a ‘doubling’ effect. This was originally intended to emulate the effect of several musicians playing the same notes simultaneously, to create a thicker, more satisfying sound.

EAX 4.0 provides a stereo chorus, with delayed signals on both left and right channels.

To add some variation to the effect, the delay time of the delayed versions of the input signal is modulated by an adjustable oscillating waveform. This causes subtle shifts in the pitch of the delayed signals, emphasizing the thickening effect.

CHORUS WAVEFORM

Value range **0 to 1**
Default value **1**
Value units **0 (sin), 1 (triangle)**

This property sets the waveform shape of the LFO that controls the delay time of the delayed signals.

CHORUS PHASE

Value range **-180 to 180**
Default value **90**
Value units **degrees**

This property controls the phase difference between the left and right LFO’s. At zero degrees the two LFOs are synchronized. Use this parameter to create the illusion of an expanded stereo field of the output signal.

CHORUS RATE

Value range **0.0 to 10.0**
Default value **1.1**
Value units **Hz**

This property sets the modulation rate of the LFO that controls the delay time of the delayed signals.

CHORUS DEPTH

Value range	**0.0 to 1.0**
Default value	**0.1**
Value units	**absolute amount**

This property controls the amount by which the delay time is modulated by the LFO.

CHORUS FEEDBACK

Value range	**-1.0 to 1.0**
Default value	**0.25**
Value units	**Absolute**

This property controls the amount of processed signal that is fed back to the input of the chorus effect. Negative values will reverse the phase of the feedback signal. At full magnitude, the identical sample will repeat endlessly. At lower magnitudes, the sample will repeat and fade out over time. Use this parameter to create a "cascading" chorus effect.

CHORUS DELAY

Value range	**0.0002 to 0.016**
Default value	**0.016**
Value units	**Seconds**

This property controls the average amount of time the sample is delayed before it is played back, and with feedback, the amount of time between iterations of the sample. Larger values lower the pitch. Smaller values make the chorus sound like a flanger, but with different frequency characteristics.

Distortion

The distortion effect simulates turning up (overdriving) the gain stage on a guitar amplifier or adding a distortion pedal to an instrument's output. It is achieved by clipping the signal (adding more square wave-like components) and adding rich harmonics.

The distortion effect could be very useful for adding extra dynamics to engine sounds in a driving simulator, or modifying samples such as vocal communications.

EAX 4.0's distortion effect also includes EQ on the output signal, to help 'rein in' excessive frequency content in distorted audio. A low-pass filter is applied to input signal before the distortion effect, to limit excessive distorted signals at high frequencies.

Distortion Edge

Value range **0.0 to 1.0**
Default value **0.2**
Value units **Absolute**

This property controls the shape of the distortion. The higher the value for *Edge*, the 'dirtier' and 'fuzzier' the effect.

Distortion Low Pass Cutoff

Value range **80.0 to 24000.0**
Default value **8000**
Value units **Hz**

Input signal can have a low pass filter applied, to limit the amount of high frequency signal feeding into the distortion effect.

Distortion Gain

Value range **-6000 to 0**
Default value **-2600**
Value units **mB**

This property allows you to attenuate the distorted sound.

Distortion EQ Center

Value range **80.0 to 24000.0**
Default value **3600**
Value units **Hz**

This property controls the frequency at which the post-distortion attenuation *(Distortion Gain)* is active.

Distortion EQ Bandwidth

Value range **80.0 to 24000.0**
Default value **3600**
Value units **Hz**

This property controls the bandwidth of the post-distortion attenuation.

Echo

The echo effect generates discrete, delayed instances of the input signal. The amount of delay and feedback is controllable. The delay is 'two tap' – you can control the interaction between two separate instances of echoes.

Echo Delay

Value range **0.002 to 0.207**
Default value **0.1**
Value units **Seconds**

This property controls the delay between the original sound and the first 'tap', or echo instance. Subsequently, the value for *Echo Delay* is used to determine the time delay between each 'second tap' and the next 'first tap'.

Echo LR Delay

Value range **0 to 0.404**
Default value **0.1**
Value units **Seconds**

This property controls the delay between the first 'tap' and the second 'tap'. Subsequently, the value for *Echo LR Delay* is used to determine the time delay between each 'first tap' and the next 'second tap'.

Echo Damping

Value range **0.0 to 0.99**
Default value **0.5**
Value units

This property controls the amount of high frequency damping applied to each echo. As the sound is subsequently fed back for further echoes, damping results in an echo which progressively gets softer in tone as well as intensity.

Echo Feedback

Value range **0.0 to 1.0**
Default value **0.5**
Value units

This property controls the amount of feedback the output signal fed back into the input. Use this parameter to create "cascading" echoes. At full magnitude, the identical sample will repeat endlessly. Below full magnitude, the sample will repeat and fade.

Echo Spread

Value range **-1.0 to 1.0**
Default value **-1.0**
Value units

This property controls how hard panned the individual echoes are. With a value of 1.0, the first 'tap' will be panned hard left, and the second tap hard right. A value of –1.0 gives the opposite result. Settings nearer to 0.0 result in less emphasized panning.

EQ

The EAX 4.0 EQ is very flexible, providing tonal control over four different adjustable frequency ranges. The lowest frequency range is called "low." The middle ranges are called "mid1" and "mid2." The high range is called "high."

Equalizer Low Gain

Value range **-1800 to 1800**
Default value **0**
Value units **mB**

This property controls amount of cut or boost on the low frequency range.

Equalizer Low Cutoff

Value range **50.0 to 800.0**
Default value **200**
Value units **Hz**

This property controls the low frequency below which signal will be cut off.

Equalizer Mid 1 Gain

Value range **-1800 to 1800**
Default value **0**
Value units **mB**

This property allows you to cut / boost signal on the "mid1" range.

Equalizer Mid 1 Center

Value range **200.0 to 3000.0**
Default value **500**
Value units **Hz**

This property sets the center frequency for the "mid1" range.

Equalizer Mid 1 Width

Value range **0.01 to 1.0**
Default value **1.0**
Value units **Octave**

This property controls the width of the "mid1" range.

Equalizer Mid 2 Gain

Value range **-1800 to 1800**
Default value **0**
Value units **mB**

This property controls the cut / boost at the "mid2" range.

Equalizer Mid 2 Center

Value range **1000.0 to 8000.0**
Default value **3000**
Value units **Hz**

This property controls the center frequency of the "mid2" range.

Equalizer Mid 2 Width

Value range **0.01 to 1.0**
Default value **1.0**
Value units **Octave**

This property controls the width of the "mid2" range.

Equalizer High Gain

Value range **-1800 to 1800**
Default value **0**
Value units **mB**

This property allows you to cut / boost the signal at high frequencies.

Equalizer High Cutoff

Value range **4000.0 to 16000.0**
Default value **6000**
Value units **Hz**

This property controls the high frequency above which signal will be cut off.

Flanger

The flanger effect creates a "tearing" or "whooshing" sound (like a jet flying overhead). It works by sampling a portion of the input signal, delaying it by a period modulated between 0 and 4ms by a low-frequency oscillator, and then mixing it with the source signal.

Flanger Waveform

Value range **0 to 1**
Default value **1**
Value units **0 (sin), 1 (triangle)**

Selects the shape of the LFO waveform that controls the amount of the delay of the sampled signal. Zero is a sinusoid and one is a triangle.

Flanger Phase

Value range **-180 to 180**
Default value **0**
Value units **Degrees**

This changes the phase difference between the left and right LFO's. At zero degrees, the two LFOs are synchronized.

Flanger Rate

Value range **0.0 to 10.0**
Default value **0.27**
Value units **Hz**

The number of times per second the LFO controlling the amount of delay repeats. Higher values increase the pitch modulation.

Flanger Depth

Value range **0.0 to 1.0**
Default value **1.0**
Value units **Absolute**

The ratio by which the delay time is modulated by the LFO. Use this parameter to increase the pitch modulation.

Flanger Feedback

Value range **-1.0 to 1.0**
Default value **-0.5**
Value units **Absolute**

This is the amount of the output signal level fed back into the effect's input. A negative value will reverse the phase of the feedback signal. Use this parameter to create an "intense metallic" effect. At full magnitude, the identical sample will repeat endlessly. At less than full magnitude, the sample will repeat and fade out over time.

Flanger Delay

Value range **0.0002 to 0.004**
Default value **0.002**
Value units **Seconds**

The average amount of time the sample is delayed before it is played back; with feedback, the amount of time between iterations of the sample.

Frequency Shifter

The frequency shifter is a single-sideband modulator, which translates all the component frequencies of the input signal by an equal amount. Unlike the pitch shifter, which attempts to maintain harmonic relationships in the signal, the frequency shifter disrupts harmonic relationships and radically alters the sonic qualities of the signal. Applications of the frequency shifter include the creation of bizarre distortion, phaser, stereo widening, and rotating speaker effects.

Frequency Shifter Frequency

Value range **0.0 to 24000.0**
Default value **0**
Value units **Hz**

This is the carrier frequency. For carrier frequencies below the audible range, the single-sideband modulator may produce phaser effects, spatial effects or a slight pitch-shift. As the carrier frequency increases, the timbre of the sound is affected; a piano or guitar note becomes like a bell's chime, and a human voice sounds extraterrestrial!

Frequency Shifter Right Direction
Frequency Shifter Left Direction

Value range **0 to 2**
Default value **0, 0**
Value units **0 (Down), 1 (Up), 2 (Off)**

These select which internal signals are added together to produce the output. Different combinations of values will produce slightly different tonal and spatial effects.

Pitch Shifter

The pitch shifter effect allows adjustments to the pitch of the input signal, without any change to the tempo or length of the source sound.

Pitch Shifter Coarse Tune

Value range **-12 to 12**
Default value **12**
Value units **Semitone**

This sets the number of semitones by which the pitch is shifted. There are 12 semitones per octave. Negative values create a downwards shift in pitch, positive values pitch the sound upwards.

Pitch Shifter Fine Tune

Value range **-50 to 50**
Default value **0**
Value units **Cent**

This sets the number of cents between Semitones a pitch is shifted. A Cent is 1/100th of a Semitone. Negative values create a downwards shift in pitch, positive values pitch the sound upwards.

Ring Modulator

The ring modulator multiplies an input signal by a carrier signal in the time domain, resulting in tremolo or inharmonic effects.

Ring Modulator Frequency

Value range **0.0 to 8000.0**
Default value **440**
Value units **Hz**

This is the frequency of the carrier signal. If the carrier signal is slowly varying (less than 20 Hz), the result is a tremolo (slow amplitude variation) effect. If the carrier signal is in the audio range, audible upper and lower sidebands begin to appear, causing an inharmonic effect. The carrier signal itself is not heard in the output.

Ring Modulator High-pass Cutoff

Value range **0.0 to 24000.0**
Default value **800**
Value units **Hz**

This controls the cutoff frequency at which the input signal is high-pass filtered before being ring modulated. If the cutoff frequency is 0, the entire signal will be ring modulated. If the cutoff frequency is high, very little of the signal (only those parts above the cutoff) will be ring modulated.

Ring Modulator Waveform

Value range **0 to 2**
Default value **0**
Value units **0 (Sin), 1 (Sawtooth), 2 (Square)**

This controls which waveform is used as the carrier signal. Traditional ring modulator and tremolo effects generally use a sinusoidal carrier. Sawtooth and square waveforms are may cause unpleasant aliasing.

Vocal Morpher

The vocal morpher consists of a pair of 4-band formant filters, used to impose vocal tract effects upon the input signal. If the input signal is a broadband sound such as pink noise or a car engine, the vocal morpher can provide a wide variety of filtering effects. A low-frequency oscillator can be used to morph the filtering effect between two different phonemes. The vocal morpher is not necessarily intended for use on voice signals; it is primarily intended for pitched noise effects, vocal-like wind effects, etc.

Vocal Morpher Phoneme A
Vocal Morpher Phoneme B

Value range **0 to 29**
Default value **0, 10**
Value units **0 "A", 1 "E", etc.**
"I", "O", "U", "AA", "AE", "AH", "AO", "EH", "ER", "IH", "IY", "UH", "UW", "B", "D","F", "G", "J", "K", "L", "M", "N", "P", "R", "S", "T", "V",
29 "Z"

These are the available phonemes (formant filter settings). If both parameters are set to the same phoneme, that determines the filtering effect that will be heard. If these two parameters are set to different phonemes, the filtering effect will morph between the two settings at a rate specified by EAXVOCALMORPHER_RATE.

Vocal Morpher Phoneme A coarse tuning
Vocal Morpher Phoneme B coarse tuning

Value range **-24 to 24**
Default value **0, 0**
Value units **Semitone**

These are used to adjust the pitch of phoneme filters A and B in 1-semitone increments.

Vocalmorpher Waveform

Value range **0 to 2**
Default value **0**
Value units **0 (sin), 1 (triangle), 2 (sawtooth)**

This controls the shape of the low-frequency oscillator used to morph between the two phoneme filters.

By selecting a sawtooth wave and a slow EAXVOCALMORPHER_RATE, one can create a filtering effect that slowly increases or decreases in pitch (depending on which of the two phoneme filters A or B is perceived as being higher-pitched).

Vocalmorpher Rate

Value range **0.0 to 10.0**
Default value **1.41**
Value units **Hz**

This controls the frequency of the low-frequency oscillator used to morph between the two phoneme filters.