Effects Extension Guide

Revision History

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Introduction

OpenAL is the most popular and effective programming interface for the development of interactive 3D audio. Today, many hardware manufacturers, platform holders and middleware providers are creating audio renderering technologies that conform to the OpenAL specs. This allows applications developers to write their audio systems using OpenAL, safe in the knowledge that their work will be re-usable across most major platforms.

Although OpenAL provides a number of sophisticated 3D aural effects such as distance based roll-off, directivity and Doppler Shift, it lacks some very important environmental effects: reverberation, reflections, and sound occlusion or obstruction by intervening objects. Without these environmental effects, a listener can tell the direction of each sound source, but has a more difficult time pin-pointing how far away the sources are. Also, the listener has no idea of the environment where the sources are located. This is where the Effects Extensions come in by adding environmental audio and filtering to OpenAL.

Environmental Audio

Consider a sword clanked in a small padded cell. It should sound very different to the same sword clanked in a large cathedral, and it is reverberation that tells the story. Or consider a scream coming from the next room. The occluded (muffled) quality of the scream tells you there's a wall in between you and the screamer. Without these environmental effects, the listener cannot pinpoint how far away the sources are and has no idea of the environment where the sources are located. The Sound sources are naked and lack warmth — the aural equivalent of a visual world without shadows, haze, and independent light sources.

Environmental Audio gives the gamer an audible indication of the characteristics of their surroundings. Reverb effects can provide the listener with sonic cues to differentiate between locations and reinforce the feeling of immersion, and environmental filtering allows architectural features of the environment to be modelled aurally.

Reverberation

Reverb and reflections combine to add a visceral realism to the 3D aural environment, an often subliminal context that can give an emotional depth to the 3D world of the player. This works even when the visual component of the 3D world is out of sight. Think, for example, of a single candle next to a pond of water in dark surroundings. When a drop of water hits the pond and you hear long and luscious reverberation on the plink of the drop, your mind senses the vast cavern surrounding the pond even though you can't see it.

In order to recreate these effects, we need to consider how reverb and reflections are created and how they can alter our perception of the environment:

Primary Reflections

Consider a simple room with floor walls and a ceiling. When a listener hears a noise in this environment, 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 colouration 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.)

Secondary Reflections

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.

Reverb

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 more and more 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.

By controlling reverb parameters such as room size, reverb level, reflections level, air absorption, reverberation decay time etc, a sound designer can accurately simulate many different types of environment.

Multi-Environment Modeling

If you are only using one reverb, there is no way to audibly indicate the presence of acoustically different spaces until the listener enters them. When the listener moves from one distinct acoustic space to another, you have to switch the reverb to reflect the environment the listener has entered.

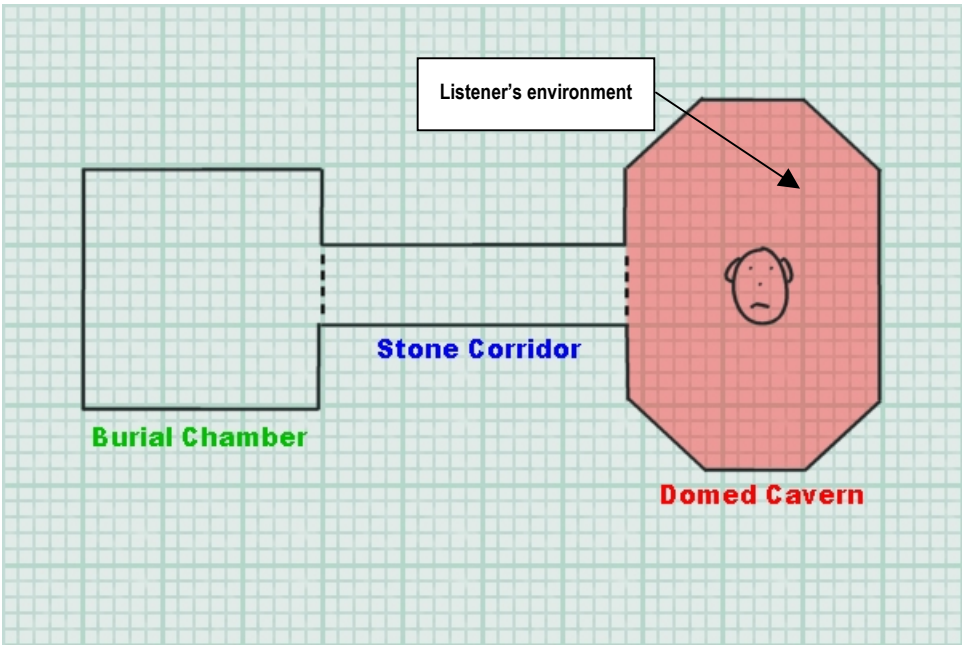


Figure 1 – With only one reverb, only the listener's environment can be modeled

A multi-environment implementation using more than one reverb can provide a more realistic experience by enabling the user to hear acoustic information from rooms other than the one occupied by the listener.

With only one reverb, if the player walks around the Domed Cavern in Figure 1, the sounds of the player's footsteps and breathing will be affected by the appropriate environmental reverb, hinting at the lofty proportions of the Cavern. So how can multi-environments enhance an application's sound-scape? It becomes clearer when other rooms are also populated with sound sources

Add to our scenario the sound of flaming torches burning in the Stone Corridor. Without multi-environments, the best we can do is to apply some Environment filtering effects to muffle the sound if

the opening between the Corridor and the Cavern is blocked or if the flaming torch is instead audible through the opening.

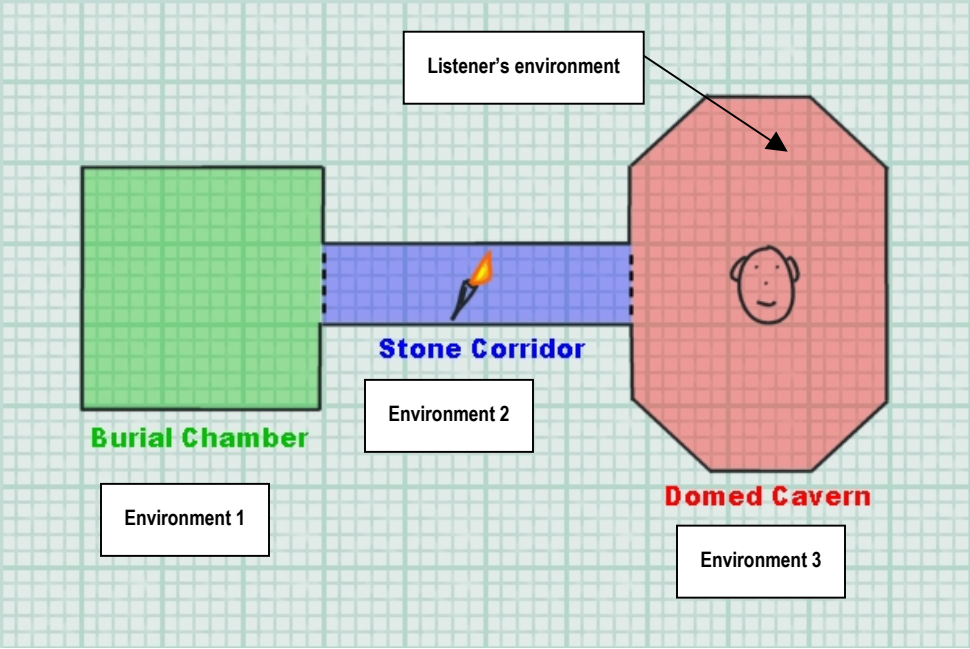


Figure 2 – Multiple reverbs allow all three environments to be rendered simultaneously

But in real life, as well as hearing the direct path of sound from the flaming torch to the listener's ear, the listener will also perceive some reflected sound with the acoustic signature of the source's own environment – the Stone Corridor. Of course, sounds generated inside the listener's environment, the Domed Cavern, will be reflected by the walls, generating that room's own acoustic effect too. The flaming torch probably will also contribute to the reverberation in the Domed Cavern. And what's more, intervening walls and obstacles will modify each of these components of sound!

With the multi-environment model, environmental reverbs can be rendered for both locations, and filtering effects applied to correctly muffle direct and reflected sound.

In addition to having multiple reverbs, you need to be able to set the directivity of each reverb as after all, if the outputs from each environmental reverb effect were simultaneously presented as being evenly spread around the listener's head, the mix of reflected sounds would make little sense; the extra information produced by the simulation of different acoustic environments would be extremely hard to discern. This is achieved by panning environments:

Panning Environments

When the listener hears reflected sound emanating from environments other than his or her own, the location of each external environment needs to be indicated.

For this reason, the Effects Extension's EAX Reverb effect includes parameters that allow you to set the directional panning of initial reflections and late reverberation decay. These panning parameters control both the perceived direction and the "divergence" of the reflections and reverberation. The "divergence" control allows for variation from diffuse surrounding reflections to reflections focused in a chosen direction.

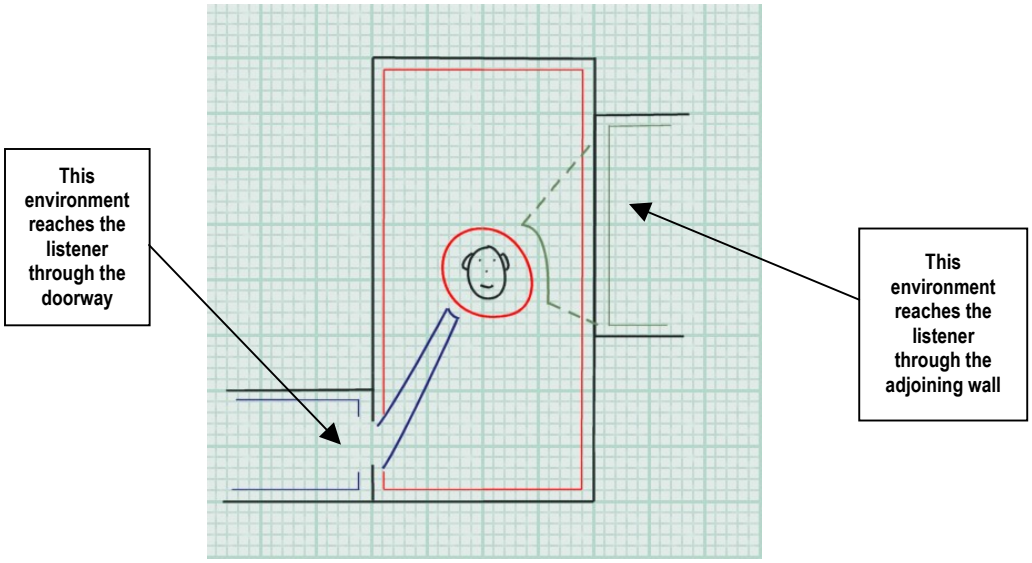


Figure 3– Navigating environments

By setting these parameters in real time, and altering the level of wet sound, you can imply a direction and distance relating the corresponding environment to the listener.

The application should identify the location of other environments currently being rendered. If an opening exists in the geometry, making a clear path between another environment and the listener's, the panning parameters for the other environments should be set, making it seem as though the reflected sound is reaching the listener's ears from that opening. If no direct path exists, the nearby environment should simply be panned and occluded to give the impression that the origin is the adjoining wall.

This technique can also be used in an application that only assigns one reverb to simulate environmental sound, to imply the existence and location of particularly acoustically reflective features in the world.

Environmental Filtering

Environmental filtering enhances the realism of an environment by simulating the way architectural features such as walls and pillars might block and absorb sound. Sound sources which are hidden behind a pillar or a wall are perceived very differently from sound sources which have an un-obstructed path to the listener's ears. 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.

There are three different types of scenario that would cause a sound source to be perceived differently by a listener: Obstruction, Occlusion and Exclusion.

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.

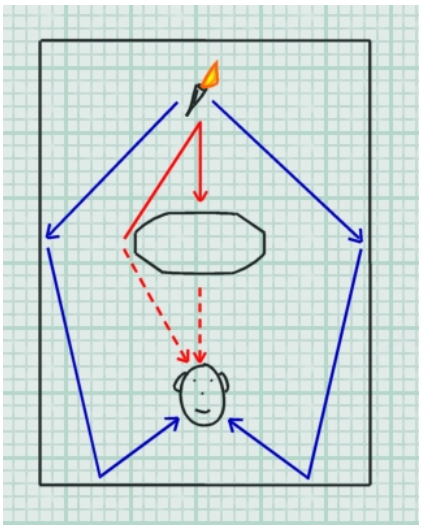


Figure 4– Obstruction: When an object in a room separates sound from listener, obstruction occurs. In this example, the direct-path sound is muffled through a partially transmissive obstacle or medium.

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.

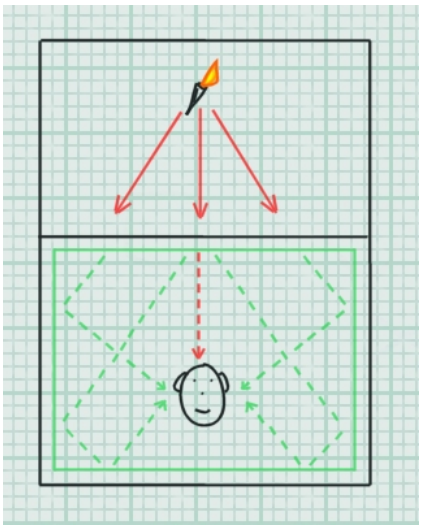


Figure 5 – Occlusion: When a full wall separates the sound source from the listener, it occludes the sound.

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're all altered—typically their high-frequency components are filtered out, leaving a very muffled result. 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.

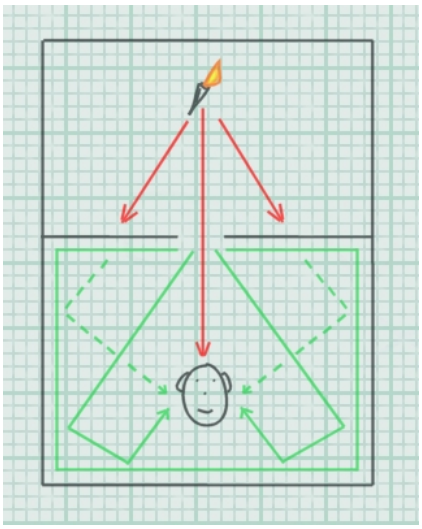


Figure 6 - When an opening breaks the wall separating the sound from the listener, the direct path is clear but the sound remains excluded from the listener's room.

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. 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).

OpenAL's architecture for 3D audio

OpenAL establishes a simple and robust framework for the processing and rendering of interactive 3D audio. The diagram below shows the fundamentals of OpenAL's processing architecture.

Figure 7 - Basic OpenAL architecture

As you can see, the signal flow is rudimentary. Buffer objects contain sample data. Source objects represent playback voices, either 3D (mono point sources for 3D positioning) or 2D, and allow the programmer to control basic properties such as attenuation, position and orientation. A Buffer is attached to a source for playback. The renderer will take care of applying 3D

virtualization techniques according to the Source's position, and mixing together all the playing Sources.

This architecture is perfectly adequate for the implementation of a real-time 3D sound scene. But no provision is made in this model for advanced processing. Examples of processing techniques required for interactive audio include:-

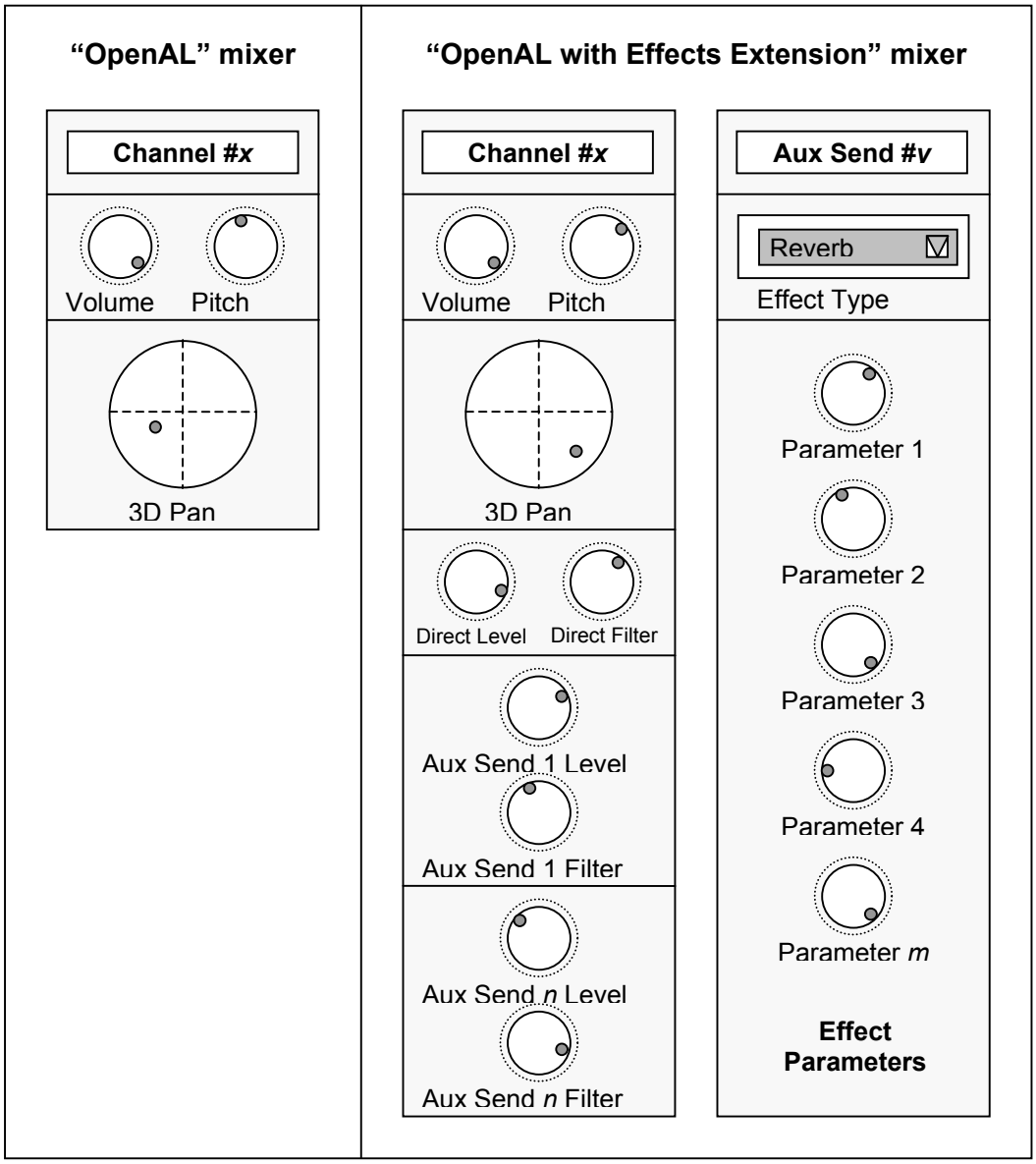
- Per-source environmental filtering (occlusion, obstruction)
- Auxiliary effect sends (e.g. environmental reverb)

Effects Extension

The Effects Extension is designed to provide a generic, cross-platform framework for adding advanced DSP effects to OpenAL. The diagram below illustrates the capabilities of the processing framework established under the OpenAL Effects Extension.

Figure 8 - OpenAL with Effects Extension architecture

This framework offers the OpenAL programmer two new ways to process audio. Firstly, Sources can be processed through filters. Secondly, Auxiliary Effect Sends are introduced, so that effects processing can be applied to groups of Sources. The output of the Auxiliary Effects is fed into the final mix. You'll notice that it is possible to filter Sources as they are fed into each Auxiliary Effect.



A good analogy here is with a traditional studio mixing desk. OpenAL on its own would map to a very simple mixing desk. Each channel strip will have minimal controls – volume, 3D or stereo pan, plus also pitch (frequency). There are no auxiliary buses. In contrast, the equivalent mixer for OpenAL with Effects Extension has an arbitrary number of auxiliary effects buses, into which you can patch a variety of effects processors with adjustable parameters. On the channel strip, a filter control is added. There are also adjustable sends to each auxiliary bus, each with a filter control as well.

Figure 9 - Mixing desk analogy - OpenAL vs OpenAL with Effects Extension

Effects Extension objects

The Effects Extensions introduce a number of new objects to OpenAL. These new objects will be described in depth later on, but in summary they are:

Auxiliary Effect Slot

An Auxiliary Effect Slot object represents an effect that can be fed with a mix of audio from selected Sources. The effect type and settings for the processor are determined by the attached Effect Object. In our mixing console analogy, this is the rack-mount slot where an effects processor can be patched into the mixer's auxiliary send.

Effect Objects
Effects objects consist of the parameters required to define an Auxiliary Effect, i.e. effect type (reverb, chorus, etc…), plus values for each of the parameters that control the effect.

Filter Objects
A filter object contains the information needed to set up a filter. I.e. the filter type (low-pass, high-pass, etc…), and values for any settings such as filter amount, cut-off etc… Filter objects can be used to filter the direct path (dry signal) of a Source, or used to filter the send path (wet signal) to any of the Auxiliary Effect Slots.

Source, Listener and Context Object Extensions
The Effects Extension also adds a number of new properties to the existing OpenAL objects.

Vendor-specific effects

Different OpenAL devices may support different effect types, including effects additional to the standard ones described here.

Auxiliary Effect Slots

Auxiliary Effect Slots are essentially containers for DSP Effects. They are positioned at the end of the signal processing graph which means that the output of an Auxiliary Effect Slot is sent directly to the final output mix. Multiple OpenAL Sources can send to the same Effect Slot, however there are limitations on the number of different Effect Slots that one Source can feed simultaneously (see ALC_MAX_AUXILIARY_SENDS).

Auxiliary Effect Slots need to be generated by the application using alGenAuxiliaryEffectSlots, and should be destroyed using alDeleteAuxiliaryEffectSlots when no longer required. They are identified by an ID in the same way that OpenAL Sources and Buffers are identified.

The Effects Extension does not impose any limits on the number of Auxiliary Effects Slots that can be created, so the application should expect that different OpenAL devices will support varying numbers of Auxiliary Effects Slots (and differing numbers of Auxiliary Sends from each Source). Tutorial 2 shows how to generate Effects Slots and check that they are successfully created.

Effects are loaded into Auxiliary Effect Slots by attaching Effect Objects to them. The Effect Object stores the type of effect and the values for all the parameters of that effect. When an Effect Object is attached to an Auxiliary Effect Slot, a check is made to see if that effect type is already loaded in the slot. If not, the appropriate effect will be loaded. Once the effect is loaded into the slot, the parameter data for the effect (stored in the Effect Object) will be applied to the actual device effect. An application should check that the Effect attaching operation is successful, because some OpenAL devices may have resource limitations that mean that not all Effect types can be loaded into all Auxiliary Effect Slots. Tutorial 3 shows how to attach an Effect to an Auxiliary Effect Slot.

Changing a parameter value in the Effect Object after it has been attached to the Auxiliary Effect Slot will not affect the effect in the effect slot. To update the parameters of the effect in the effect slot, an application must update the parameters of an Effect object and then re-attach it to the Auxiliary Effect Slot.

To unload an Effect from an Auxiliary Effect Slot the application should attach the empty Effect object define: AL_EFFECT_NULL.

Finally, an application can control the output level of an Auxiliary Effect Slot using AL_EFFECTSLOT_GAIN and enable or disable automatic send adjustment using AL_EFFECTSLOT_AUXILIARY_SEND_AUTO.

For a complete list of Auxiliary Effect Slot related functions and properties please refer to the Auxiliary Effect Slot Object section.

Effects

Effect objects are containers for audio Effects. The Effect object stores the type of effect and a list of values for the parameters for that effect. The Effect object can therefore be used to store 'presets' for different effects, e.g. an Effect object could contain a reverb effect with a set of parameters that can be used to model a Bathroom environment, or a reverb effect with a set of parameters that can be used to model a Cave environment, or an echo effect with a set of parameters that can be used to model a Canyon.

Effect objects need to be generated by the application using alGenEffects, and should be destroyed using alDeleteEffects when no longer required. They are identified by an ID in the same way that OpenAL Sources and Buffers are identified.

The Effects Extension does not impose any limits on the number of Effects that can be created, however different OpenAL devices will support different effect types.

After generating an Effect object the application must tell the Effect what type of Effect it should store (see AL_EFFECT_TYPE). If this is successful, the application can set the values for each of the parameters supported by the effect. This is done using the alEffect[i,iv,f,fv] function calls. Tutorial 2 shows how to generate Effects and check that they are successfully created. It also shows how to set the type of effect stored in the Effect object and how to update the parameters of that effect.

In order to hear the results of an Effect it must be loaded into an Auxiliary Effect Slot (see Tutorial 3). Once loaded into an effect slot any Sources that have been configured to send to that effect slot will automatically start to feed the effect (see Tutorial 4).

Changing a parameter value in the Effect Object after it has been attached to the Auxiliary Effect Slot will not affect the effect in the effect slot. To update the parameters of the effect in the effect slot, an application must update the parameters of an Effect object and then re-attach it to the Auxiliary Effect Slot.

For a complete list of Effect related functions and properties please refer to the Effect Object section.

Filters

Filter objects are containers for audio Filters. The Filter object stores the type of filter and a list of values for the parameters for that filter. The Filter object can therefore be used to store 'presets' for different filters, e.g. a Filter object could contain a low-pass filter with a set of parameters that can be used to model sound passing through a concrete wall, or a band-pass filter with a set of parameters that can be used to model transmission of audio over a telephone.

Filter objects need to be generated by the application using alGenFilters, and should be destroyed using alDeleteFilters when no longer required. They are identified by an ID in the same way that OpenAL Sources and Buffers are identified.

The Effects Extension does not impose any limits on the number of Filters that can be created, however different OpenAL devices will support different filter types.

After generating a Filter object the application must tell the Filter what type of filter it should store (see AL_FILTER_TYPE). If this is successful, the application can set the values for each of the parameters supported by the filter. This is done using the alFilter[i,iv,f,fv] function calls. Tutorial 2 shows how to generate Filters and check that they are successfully created. It also shows how to set the type of filter stored in the Filter object and how to update the parameters of that filter.

Filter objects can be used in two different ways; to filter an OpenAL Source, or to filter the send from an OpenAL Source to an Auxiliary Effect Slot. When a filter object is attached to a Source as a Direct Filter (using alSourcei with the property AL_DIRECT_FILTER), filtering is applied to the direct (dry) signal of the Source only. When a filter object is attached to a Source as an Auxiliary Send Filter (using alSource3i with the property AL_AUXILIARY_SEND_FILTER), filtering is applied to the signal being sent to the Auxiliary Effect Slot. Tutorial 5 shows the two different ways of using Filter objects.

Changing a parameter value in the Filter Object after it has been attached to a Source will not affect the Source. To update the filter(s) used on a Source, an application must update the parameters of a Filter object and then re-attach it to the Source.

For a complete list of Filter related functions and properties please refer to the Filter Object section.

Source Extensions

To integrate Effect Extension functionality into OpenAL a number of new Source properties have been added. These properties allow Sources to use Filters, Effects, and Auxiliary Effect Slot objects. In addition a number of properties have been added to enhance the 3D spatialization model of OpenAL.

Enabling a Source Auxiliary Send

In order for a Source to feed an Effect that has been loaded into an Auxiliary Effect Slot the application must configure one of the Source's auxiliary sends. This process involves setting 3 variables – the destination Auxiliary Effect Slot ID, the Auxiliary Send number, and an optional Filter ID.

The ID of the Auxiliary Effect Slot is simply the value returned from a successful call to alGenAuxiliaryEffectSlots.

The Auxiliary Send number identifies which of the Source's Auxiliary Sends is being used to send to the specified Auxiliary Effect Slot. The number of Auxiliary Sends available on each Source is OpenAL device dependent (see ALC_MAX_AUXILIARY_SENDS).

If an application wishes to filter the send from the Source to the Auxiliary Effect Slot it can provide a valid Filter ID. If no filtering is required this value should be set to AL_FILTER_NULL.

The alSource3i function call is used to pass the values to OpenAL using the property AL_AUXILIARY_SEND_FILTER. Tutorial 4 shows how to configure the Auxiliary Sends on a Source.

Disabling a Source Auxiliary Send

To disable a particular Auxiliary Send from a Source, the application should configure that send number to send to the null Auxiliary Effect Slot. Tutorial 4 shows how to disable an Auxiliary Send.

Enabling a Source Filter

To apply filtering on the direct-path (dry signal) of a Source, a Filter object can be attached to the Source using alSourcei with the property AL_DIRECT_FILTER and passing in a Filter ID. Tutorial 5 shows how to attach a filter to a Source.

Disabling a Source Filter

To remove a Filter from a Source, the application should attach the null Filter object to the Source using alSourcei with the property AL_DIRECT_FILTER. Tutorial 5 shows how to remove a filter from a Source.

Enhanced 3D Spatialization Modeling Properties

The amount of Air Absorption applied to each OpenAL Source can be adjusted using the AL_AIR_ABSORPTION_FACTOR property.

If an application is using Cone parameters on Sources, then an additional property available through the Effects Extensions called AL_CONE_OUTER_GAINHF allows the application to control a low-pass filter that is applied when the source is facing away from the listener.

The amount of attenuation applied to the Source's auxiliary send level can be adjusted using the AL_ROOM_ROLLOFF_FACTOR property. This property is disabled if the Auxiliary Effect Slot property AL_EFFECTSLOT_AUXILIARY_SEND_AUTO is set to AL_FALSE.

AL_DIRECT_FILTER_GAINHF_AUTO is used to enable or disable the attenuation of high-frequencies in the Source's direct (dry) path based on the setting of AL_CONE_OUTER_GAINHF.

AL_AUXILIARY_SEND_FILTER_GAIN_AUTO is used to enable or disable attenuation of reflected sound based on the source-listener distance and the source's orientation.

AL_AUXILIARY_SEND_FILTER_GAINHF_AUTO is used to enable or disable high-frequency attenuation of reflected sound based on source orientation and the AL_CONE_OUTER_GAINHF setting.

Listener Extensions

To integrate Effect Extension technology into OpenAL one new property was added to the OpenAL Listener object. This property simply allows the application to provide unit information to the Effects Extension so that distance related properties such as Air Absorption are applied correctly.

The distance unit being used by the application should be set using a call to alListenerf with the property AL_METERS_PER_UNIT. If the application is using centimeters for distance units, then this property should be set to 0.01 so that the amount of air absorption applied is not 100 times too great!

Context Extensions

The Effect Extension adds a few new properties to the OpenAL Context object. The most important property is the number of Auxiliary Sends that are available on each OpenAL Source. The other properties are for querying for the version of the Effects Extension supported by OpenAL.

The ALC_MAX_AUXILIARY_SENDS property is used to hint to the OpenAL Context (at Context creation time) the maximum number of Auxiliary Sends desired on each Source. It is not guaranteed that the desired number of sends will be available, so an application should query this property after creating the context using alcGetIntergerv. Tutorial 1 shows how to initialize OpenAL and the Effect Extension including requesting and querying the number of Auxiliary Sends per Source.

Auxiliary sends on a source are identified by their 0 based indices. As an example if a Source has 2 Auxiliary Sends then they are referred to as Send 0 and Send 1. More information on how to access and control Auxiliary Sends on a Source is given in the Source Extensions section.

The ALC_EFX_MAJOR_VERSION and ALC_EFX_MINOR_VERSION properties are used to query for the version of the Effect Extension supported. An application should use the OpenAL function alcGetIntegerv to retrieve the values for these properties.

Programming the Effects Extension

This section introduces you to the techniques required to access the features of the Effects Extension. The following tutorials illustrate how to use the functionality of the Effect Extension.

Tutorial 1: Initializing OpenAL and the Effects Extension

Shows how to initialize OpenAL and query for the Effect Extension. The source code also illustrates how to use the Context Creation hint ALC_MAX_AUXILIARY_SENDS to request the number of Auxiliary Sends available on each Source (and how to check the actual number available). Finally the code shows how to retrieve the pointers to the Effects Extension functions.

```
ALCdevice *pDevice = NULL;
ALCcontext *pContext = NULL;
ALint attribs[4] = { 0 };
ALCint iSends = 0;

/* Open default OpenAL device */
pDevice = alcOpenDevice(NULL);
if (!pDevice)
      return;

/* Query for Effect Extension */
if (alcIsExtensionPresent(pDevice, "ALC_EXT_EFX") == AL_FALSE)
      return;

printf("EFX Extension found!\n");

/* Use Context creation hint to request 4 Auxiliary */
/* Sends per Source */
attribs[0] = ALC_MAX_AUXILIARY_SENDS;
attribs[1] = 4;

pContext = alcCreateContext(pDevice, attribs);
if (!pContext)
      return;

/* Activate the context */
alcMakeContextCurrent(pContext);

/* Retrieve the actual number of Aux Sends */
/* available on each Source */
alcGetIntegerv(pDevice, ALC_MAX_AUXILIARY_SENDS, 1, &iSends);
```

```
printf("Device supports %d Aux Sends per Source\n", iSends);


/* Get the Effect Extension function pointers */
alGenEffects=(LPALGENEFFECTS)
      alGetProcAddress("alGenEffects");
alDeleteEffects=(LPALDELETEEFFECTS)
      alGetProcAddress("alDeleteEffects");
alIsEffect=(LPALISEFFECT)
      alGetProcAddress("alIsEffect");


/* ... */


/* Check function pointers are valid */
if (!(alGenEffects && alDeleteEffects && alIsEffect))
      return;


/* EFX available and ready to be used ! */
```

Tutorial 2: Creating Auxiliary Effect Slots, Effects, and Filters

Shows how to create Auxiliary Effect Slots, Effects and Filters and check for errors. It also shows how to set Effect types and parameters, and Filter types and parameters.

```
ALuint uiEffectSlot[4] = { 0 };
ALuint uiEffect[2] = { 0 };
ALuint uiFilter[1] = { 0 };
ALuint uiLoop;

/* Try to create 4 Auxiliary Effect Slots */
alGetError();
for (uiLoop = 0; uiLoop < 4; uiLoop++)
{
      alGenAuxiliaryEffectSlots(1, &uiEffectSlot[uiLoop]);
      if (alGetError() != AL_NO_ERROR)
            break;
}

printf("Generated %d Aux Effect Slots\n", uiLoop);

/* Try to create 2 Effects */
for (uiLoop = 0; uiLoop < 2; uiLoop++)
{
      alGenEffects(1, &uiEffect[uiLoop]);
      if (alGetError() != AL_NO_ERROR)
            break;
}

printf("Generated %d Effects\n", uiLoop);

/* Set first Effect Type to Reverb and change Decay Time */
alGetError();
if (alIsEffect(uiEffect[0]))
{
      alEffecti(uiEffect[0], AL_EFFECT_TYPE, AL_EFFECT_REVERB);
      if (alGetError() != AL_NO_ERROR)
            printf("Reverb Effect not supported\n");
      else
```

```
            alEffectf(uiEffect[0], AL_REVERB_DECAY_TIME, 5.0f);
}

/* Set second Effect Type to Flanger and change Phase */
alGetError();
if (alIsEffect(uiEffect[1]))
{
      alEffecti(uiEffect[1],AL_EFFECT_TYPE,AL_EFFECT_FLANGER);
      if (alGetError() != AL_NO_ERROR)
            printf("Flanger effect not support\n");
      else
            alEffecti(uiEffect[1], AL_FLANGER_PHASE, 180);
}

/* Try to create a Filter */
alGetError();
alGenFilters(1, &uiFilter[0]);
if (alGetError() == AL_NO_ERROR)
      printf("Generated a Filter\n");

if (alIsFilter(uiFilter[0]))
{
      /* Set Filter type to Low-Pass and set parameters */
      alFilteri(uiFilter[0],AL_FILTER_TYPE,AL_FILTER_LOWPASS);
      if (alGetError() != AL_NO_ERROR)
            printf("Low Pass Filter not supported\n");
      else
      {
            alFilterf(uiFilter[0], AL_LOWPASS_GAIN, 0.5f);
            alFilterf(uiFilter[0], AL_LOWPASS_GAINHF, 0.5f);
      }
}
```

Tutorial 3: Attaching an Effect to an Auxiliary Effect Slot

Shows how to load an Effect into an Auxiliary Effect Slot.and check for errors.

```
/* Attach Effect to Auxiliary Effect Slot */


/* uiEffectSlot[0] is the ID of an Aux Effect Slot */
/* uiEffect[0] is the ID of an Effect */


alAuxiliaryEffectSloti(uiEffectSlot[0],
      AL_EFFECTSLOT_EFFECT, uiEffect[0]);


if (alGetError() == AL_NO_ERROR)
      printf("Successfully loaded effect into effect slot\n");
```

Tutorial 4: Configuring Source Auxiliary Sends

Shows how to configure the Auxiliary Sends on a Source to feed different Auxiliary Effect Slots.

```
/* Configure Source Auxiliary Effect Slot Sends */


/* uiEffectSlot[0] and uiEffectSlot[1] are Auxiliary */
/* Effect Slot IDs */
/* uiEffect[0] is an Effect ID */
/* uiFilter[0] is a Filter ID */
/* uiSource is a Source ID */


/* Set Source Send 0 to feed uiEffectSlot[0] without */
/* filtering */
alSource3i(uiSource,AL_AUXILIARY_SEND_FILTER,   uiEffectSlot[0],
0, NULL);
if (alGetError() != AL_NO_ERROR)
      printf("Failed to configure Source Send 0\n");


/* Set Source Send 1 to feed uiEffectSlot[1] with */
/* filter uiFilter[0] */
alSource3i(uiSource,AL_AUXILIARY_SEND_FILTER,   uiEffectSlot[1],
1, uiFilter[0]);
if (alGetError() != AL_NO_ERROR)
      printf("Failed to configure Source Send 1\n");


/* Disable Send 0 */
alSource3i(uiSource,AL_AUXILIARY_SEND_FILTER,
      AL_EFFECTSLOT_NULL, 0, NULL);
if (alGetError() != AL_NO_ERROR)
      printf("Failed to disable Source Send 0\n");


/* Disable Send 1 */
alSource3i(uiSource,AL_AUXILIARY_SEND_FILTER,
      AL_EFFECTSLOT_NULL, 1, NULL);
if (alGetError() != AL_NO_ERROR)
      printf("Failed to disable Source Send 1\n");
```

Tutorial 5: Attaching Filters to Sources

Shows how Filters can be used on Sources to filter the direct signal (dry path) and also the send signal (wet path).

```
/* Filter 'uiSource', a generated Source */


alSourcei(uiSource, AL_DIRECT_FILTER, uiFilter[0]);
if (alGetError() == AL_NO_ERROR)
{
      printf("Successfully applied a direct path filter\n");
      /* Remove filter from 'uiSource' */
      alSourcei(uiSource, AL_DIRECT_FILTER, AL_FILTER_NULL);
      if (alGetError() == AL_NO_ERROR)
            printf("Successfully removed direct filter\n");
}



/* Filter the Source send 0 from 'uiSource' to */
/* Auxiliary Effect Slot uiEffectSlot[0] */
/* using Filter uiFilter[0] */


alSource3i(uiSource,AL_AUXILIARY_SEND_FILTER,
      uiEffectSlot[0], 0, uiFilter[0]);
if (alGetError() == AL_NO_ERROR)
{
      printf("Successfully applied aux send filter\n");


      /* Remove Filter from Source Auxiliary Send */
      alSource3i(uiSource,AL_AUXILIARY_SEND_FILTER,
      uiEffectSlot[0], 0, AL_FILTER_NULL);
      if (alGetError() == AL_NO_ERROR)
            printf("Successfully removed filter\n");
}
```

Tutorial 6: Source Properties

Shows how to set Effects Extension specific Source properties.

```
/* Set Source Cone Outer Gain HF value */
alSourcef(uiSource, AL_CONE_OUTER_GAINHF, 0.5f);
if (alGetError() == AL_NO_ERROR)
      printf("Successfully set cone outside gain filter\n");/*
```

Tutorial 7: Listener Properties

Shows how to set Effects Extension specific Listener properties.

```
/* Set distance units to be in feet*/
alListenerf(AL_METERS_PER_UNIT, 0.3f);
if (alGetError() == AL_NO_ERROR)
      printf("Successfully set distance units\n");
```

Environmental Audio Programming Techniques

This section looks at the Effects Extension from a higher-level perspective – from the point of view of integrating environmental effects into the rest of the application.

In an application scenario, the program logic and interaction with the user should provide the audio engine with sufficient information to adjust the configuration of effects in real-time. For an application with multiple environment support, the mechanisms needed to maintain a lifelike acoustic simulation are not trivial.

This section covers all these aspects of application design, giving you the knowledge to make sure your Effects Extension application is robust and efficient, and most importantly sounds great! The information which follows is not necessarily a recipe for the perfect audio implementation. Rather it is a set of suggestions, inspired by Creative's experiences working with a large number of commercial 3D audio projects.

Creating a single environment world with the Effects Extension

Basic hardware and the "Generic Software" device can render one environmental effect (usually reverb).

Using only one auxiliary effect slot, updating the effect according to the environment is simple. Environmental zones can be created in an associated tool such as a level editor, and effect settings defined by the sound designer. Obstruction tests can be performed using ray-casting to determine if any objects are in between the Source and the Listener. Occlusion can be used whenever the Source and Listener are in different environments with no line-of-sight between them. Exclusion can be used when the Source and Listener are in different environments but there is a line-of-sight between them.

When changing environment, it may be beneficial to interpolate parameters in order to avoid audible popping sounds in the effect.

Creating a multi-environment world with the Effects Extension

The ability to render several audio effects at once on supported hardware allows the developer to simulate a world containing multiple environments, with the auxiliary effects slots rendering different effects for environments likely to be heard by the listener. Designing a mechanism to correctly update all the effects properties for an accurate multi-environment simulation is not trivial; a number of tasks are involved. An environmental audio management layer must:

- Ensure that the listener's environment is always being correctly rendered.

- Ensure that the nearby environments most likely to be audible are being correctly rendered in other available auxiliary effects slots.

- Ensure that each source is feeding the correct auxiliary effects slots.

- Ensure that each effect is correctly localised, to correspond with the location of its environment in the game world.

These are in addition to the environmental audio management tasks associated with a single-environment simulation. These include checking the status of occlusion, exclusion and obstruction on each sound source in order to update the filter effects.

The exact manner in which the user interaction, application logic and world geometry influence the audio engine is very much specific to each individual project. However, there is a general set of principles that will apply in most instances.

Environmental Zones

Correctly applying environmental audio involves establishing the interaction between the listener and sound sources in the virtual world, and the various environments that surround the listener and sources. The most common approach to managing environmental data is to separate the world's geometry into environmental zones. Each zone represents a separate acoustic enclosure in the world. The environmental audio management layer should be able to reference the co-ordinates for the listener and each sound source. Given any set of co-ordinates, it should be possible to uniquely identify the environmental zone in which the co-ordinates exist. Typically, a set of parameters for an effect will be associated with each unique zone identifier. This means that, for example, given co-ordinates showing the listener's position, it is simple to retrieve the set of effect parameters for the listener's zone.

Data structures such as binary search partition trees offer efficient solutions for storing and retrieving environmental zone data. The work of creating effect parameter sets for each zone would commonly be undertaken by an audio designer, using an off-line tool.

The ability to separate world geometry into discrete zones, and reference a unique zone identifier from a given set of co-ordinates also helps when determining whether a sound should have occlusion effects applied. If it is detected that a source is in a different zone to the listener, then it is likely that some occlusion will be applied. The amount of occlusion applied when sound travels between two zones could be calculated dynamically at run-time. Or again, this data could be prepared by an audio designer or pre-calculated algorithmically 'off-line' and stored in a look-up table.

Apertures between environmental zones

To work with more advanced environmental audio techniques such as exclusion and multiple environments, identifying 'apertures' or 'portals' between environmental zones is essential. An aperture can be defined as the area where a clear-air opening between two environmental zones exists, for example a doorway or window.

In a multiple environment system where three environments are being rendered, it is important that the two non-listener environments are correctly localised. If an aperture exists between the listener's zone and a secondary zone, then the secondary zone's reverb should be perceived to be emitting from the aperture. For information on localising environmental reverb effects, see Reverb and Reflection Panning Algorithm.

Aperture data can also be used to add an extra degree of realism to occluded sounds. If a sound has been found to exist in an environment zone adjacent to the listener's, then the direct line between the sound source and the listener can be calculated. If this line intersects an aperture, then the sound should have exclusion applied instead of occlusion, because although the source and listener are in different environments the direct sound is not blocked.

Source to listener direct path

Efficiently determining whether a sound source should have an obstruction effect applied can be a tricky process. However, correctly used obstruction effects add a great deal to 3D audio in terms of

realism. In a similar manner to exclusion, this can be achieved by calculating the direct line between the source and the listener, this time checking whether it intersects any world geometry.

A potential optimisation is to assume that only a sound that is in the same environmental zone as the listener can possibly be obstructed. This pre-supposes that an occlusion effect applied to a sound from other environmental zones will mask any subsequent obstruction effect, not strictly a true assumption in the real world.

Pre-generating and storing axis-aligned bounding boxes which identify any potential obstruction (e.g. pillar, statue) within an environmental zone helps to increase efficiency greatly, but at the expense of some accuracy.

In real life, the diffraction of sound waves around an object means that the nearer the direct line between obstructed source and listener comes to the edge of the obstacle, the less filtering and attenuation takes place. So, to make obstruction effects sound even more natural, it is helpful to dynamically calculate the amount of obstruction to be applied according to the angle of incidence between the source, the edge of the obstruction, and the listener

Figure 3 shows how the amount of obstruction varies with the angle around the obstacle.

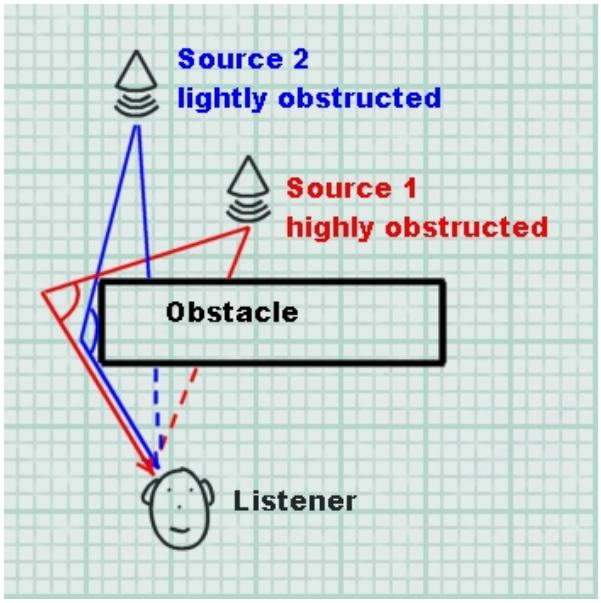


Figure 10- Variable amounts of obstruction

Low-detail models and shared systems

A significant performance optimisation for environmental data management is to utilise a low detail model of world geometry for audio calculations. As is often the case, the best efficiency gain comes when sharing geometry calculations with other application systems. For example, AI, collision-detection and physics systems often involve low-detail geometrical models and similar mechanisms and calculations to those described above.

Multi-environment run-time management algorithm

Simulation of multiple environments is a key aim in the design of the Effects Extension SDK. Creative's engineers have proposed a general algorithm for environmental audio data management in a multiple-environment implementation. The following represents an example of how the effects-related audio program flow might work for a game. In the following pseudo-code '*n*' represents the maximum

number of simultaneous environments that can be rendered (for example, that would be 4 on Sound Blaster X-Fi and 2 on Sound Blaster Audigy).

Each audio frame:

Step 1: Update Environments

If** listener position has changed **then:

Find the (*n*-1) closest environments to the listener (listener environment is always rendered):

- Option 1: Based on radius or priorities assigned to each environment zone
- Option 2: Based on distance from Listener to centre of environment
- Option 3: Based on environment's closest aperture

Update FX Slots

For each Auxiliary Effect slot not rendering one of the chosen 'n' environments, set the Auxiliary slot gain to 0 (silence) and load one of the chosen environment Reverb settings into it.

For each Auxiliary Effect slot set appropriate Pan and Reverb gain levels (based on listener orientation and distance from listener to environment).

Else if** listener orientation has changed **then:

For each auxiliary effect slot set appropriate pan and reverb gain levels (based on listener orientation and distance from listener to environment).

Step 2: Update sources

For every source:

If** source environment has changed **or** listener position changed **then:

If** source environment == listener environment **then :

Activate source send 0 to the auxiliary effect slot rendering the listener environment without filtering.

Deactivate source send 1.

Check for obstruction and add filtering to direct path if necessary.

Else:

Activate source send 0 to the auxiliary effect slot rendering the listener environment and add filtering (occlusion).

If** the source environment is being rendered **then:

Activate source send 1 to the auxiliary effect slot rendering the source environment.

Else:

Deactivate source send 1.

Set all auxiliary effect slots gain to 1.

Reverb and Reflection Panning Algorithm

The EAX Reverb effect exposes complete control over the early reflections and the late reverberation, including parameters that control the spatial distribution of these separate parts.

The spatial distribution of the early reflections is controlled by using a single vector that indicates the direction of the reflections, while its magnitude controls how focused the reflections are toward this direction. A vector of magnitude 0 (the default) creates reflections that come evenly from all directions, whereas a vector of magnitude 1.0 is highly focused to a particular point. The spatial distribution of the late reverb is controlled in the same way, using a separate vector.

Both vectors are interpreted in the co-ordinate system of the user, without taking into account the orientation of the 3D listener. For example, setting the reverb pan to (0.0, 0.0, 0.7) means that the reverb is panned to the front speakers, whereas a vector of (0.0, 0.0, -0.7) pans the reverb to the rear speakers.

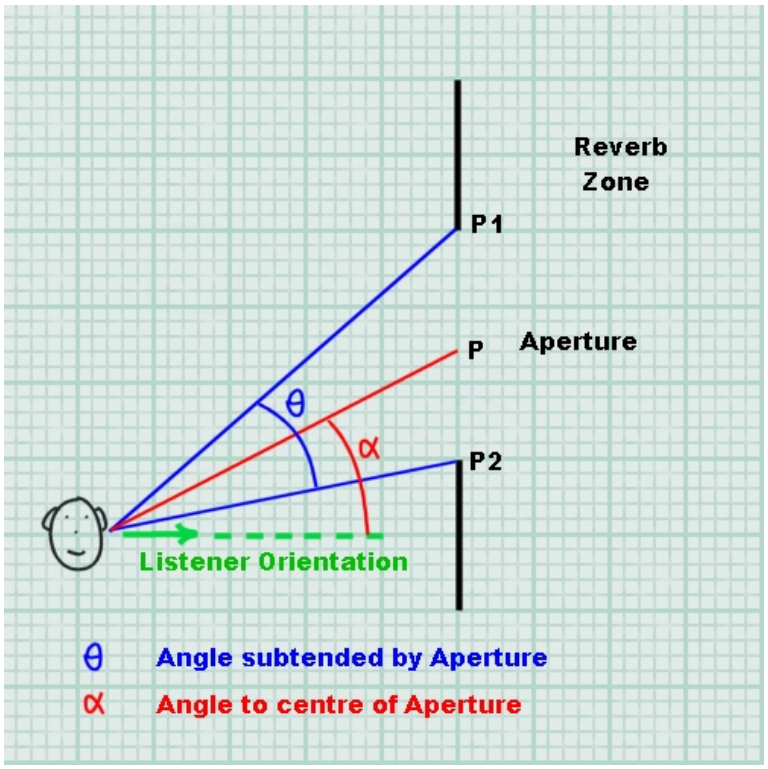


Figure 11 - Example scenario for effects panning

Assume the Listener position is stored in the vector vListenerPos, the orientation in vListenerOri, the centre of the aperture (P) in vAperture, and the aperture corners (P1 and P2) in vApertureLeftSide and vApertureRightSide.

All angles and trigonometric functions are using radians.

Orientation

In order to compute the orientation for the Reflections and Reverb pan vectors it is necessary to know the orientation of the listener, and the desired location of the environment. In Figure 3, the reverb on the other side of the aperture should be panned to the centre of the aperture, point P on the diagram.

To translate P to the "user-relative" position required by the Reflections and Reverb Pan vectors, the vector LP should be inversely rotated by the angle between the listener orientation and the straight-ahead vector (0,0,1).

To calculate this angle, take the dot product of the Listener Orientation (should already be normalized) and (0,0,1). Since several terms cancel out, the angle is simply: -

Angle = arccos (vListenerOri.z).

If the Listener is facing to the left of straight ahead then the angle should be negated: -

```
if (vListenerOri.x < 0)

    Angle = -Angle
```

The vector LP is defined as: -

LP.x = vAperture.x – vListenerPos.x

LP.y = vAperture.y – vListenerPos.y

LP.z = vAperture.z – vListenerPos.z

So, the Vector can be calculated by rotating LP by –Angle: -

Pan.x = (LP.x * cos(-Angle)) + (LP.z * sin(-Angle))

Pan.y = 0

Pan.z = (LP.x * -sin(-Angle)) + (LP.z * cos(-Angle)

The resulting Pan vector is a "user-relative" vector pointing to the location where the reverb should be panned. This vector should be normalized, and then multiplied by the magnitude calculated in the next step.

Magnitude

The magnitude of the Reflection and Reverb pan vectors indicate how focused the reflected sound should be. To compute the magnitude, the angle subtended by the aperture (θ in the diagram) needs to be calculated.

The vectors LP1 and LP2 are defined as: -

LP1.x = vApertureLeftSide.x – vListenerPos.x

LP1.y = vApertureLeftSide.y – vListenerPos.y

LP1.z = vApertureLeftSide.z – vListenerPos.z

LP2.x = vApertureRightSide.x – vListenerPos.x

LP2.y = vApertureRightSide.y – vListenerPos.y

LP2.z = vApertureRightSide.z – vListenerPos.z

After LP1 and LP2 have been normalized, the angle is computed.

θ = arccos((LP1.x * LP2.x) + (LP1.y * LP2.y) + (LP1.z * LP2.z));

The magnitude of the Vector can be calculated using the following formula: -

Magnitude = (2.0f * sin (θ / 2.0)) / θ

Assuming Pan is a normalized vector, the Reflections and Reverb vectors should be set to: -

Pan.x = Pan.x * Magnitude

Pan.y = Pan.y * Magnitude

Pan.z = Pan.z * Magnitude

Angle (θ) Degrees	Angle (θ) Radians	Magnitude
0	0	1.0
45	π / 4	0.97
90	π / 2	0.90
180	π	0.637
270	3 π / 2	0.3
360	2 π	0

Table 1 - magnitude values for different panning focus angles

In addition to controlling the position and focus of the early reflections and late reverb, the overall volume can be attenuated to indicate the distance between the listener and the acoustically reverberant environment.

Performance and Optimization

Over recent years, advanced 3D sound technologies like hardware DSP effects and 3D sound buffers have gained a reputation for hogging the processor and causing performance problems. However, a close understanding of the potential bottlenecks in 3D sound programming will enable the developer to avoid inefficient use of hardware resources. Effects coding techniques are inextricably linked with hardware 3D sound management, so this section will deal with both of these topics together. With a few key optimisations, implementing hardware OpenAL 3D sound with effects extension support in your project won't burn unnecessary CPU time or cause excessive slowdown to the frame rate.

Hardware vs. Software audio

A standard software 3D audio renderer such as the 'generic software' device generally employs simple algorithms to simulate 3D spatialization using stereo panning and attenuation. On a PC system with a modern CPU, multiple audio streams can be panned, balanced and mixed together with a fraction of the available computing resources. However, processing effects in software is a resource hungry task and only a limited amount of effects can be used simultaneously. Software mixers typically update at a relatively slow rate, maybe 40 times per second, thus introducing up to 25 milliseconds of latency. Changes to audio, for instance setting a new listener position or orientation, or playing and stopping sound buffers, will only be effective to a resolution of 25ms.

Hardware 3D sound renderers, such as the 'generic hardware' device or native devices, on the other hand, spatialize sounds over several speakers with sophisticated multi-speaker panning algorithms, or intensive HRTF processing on headphone and stereo playback systems. A typical hardware accelerator will respond immediately to changes in audio, so that all 3D sources will be instantly affected by any change in the listener's position or orientation. New sound sources will typically begin playing straight away. On Creative hardware, when using effects extensions, the hardware audio accelerator is additionally performing complex audio processing such as filtering and digital reverberation. Rendering up to 128 equivalent 3D voices with effects in software would be prohibitively CPU intensive, even on today's fastest processors.

So when comparing audio performance between software and hardware 3D audio renderers, one must bear in mind the quality of the results. That said, much of the work in hardware 3D sound is carried out on the soundcard's own dedicated audio processor. The low level processing which does take up CPU cycles is comparatively simple.

Solutions for optimisation

Updating positions, effects parameters such as panning, send levels or adding filters to an application can be CPU consuming tasks, most of the cost being at the application level. It is possible to propose techniques to lower CPU cost. The solutions involve cutting down the update frequency, and avoiding redundant updates.

Audio frame rate

Establishing an independent audio frame rate is a crucial performance technique that can potentially halve the CPU time spent on 3D audio updates. The human eye detects visual changes at a very high resolution, hence a high graphical frame rate is desirable – graphical updates should ideally occur at around 60 frames per second to give good smooth visuals.

Many developers update their 3D sound scene each time the screen is refreshed. However, human ears do not require such frequent updates to 3D sound positioning. In fact, a rate of 30 positional updates per second is more than adequate to preserve convincing 3D audio. Note that although sound

positioning doesn't require updates over 30Hz, some properties such as sound playback frequency, popular for simulating vehicle engine speed, could need more frequent updates.

On the other hand, some effect settings such as environmental reverb, which can be more expensive to modify, actually require an even lower update rate, as low as 10 to 15Hz. One exception to this is the Reflections and Reverb pan vectors, which if being used to localize secondary environments, should be updated as often as the Listener position and orientation.

Application property	**Suggested update rate**
Graphical scene	Typically 60 Hz+
Sound playback frequency for pitch adjustments on sound sources	60 Hz
3D listener Position / Orientation	30 Hz
3D sound source position / directivity	30 Hz
Reverb / reflections panning vectors	30 Hz
Reverberations parameter adjustments	15 Hz

Table 2 - Optimum update rates for different 3D audio properties

So, developing techniques to limit the frequency of positional audio updates to each source and the listener to pre-determined rates will save plenty of CPU cycles. Table 2 shows optimum independent rates for updating resolution-critical settings.

Programmers Reference

Auxiliary Effect Slot Object

Management Functions

To handle the creation, deletion, and verification of Auxiliary Effect Slot objects the following functions are supplied:

Version	Function Name	Purpose
1.0	alGenAuxiliaryEffectSlots	Create *n* Auxiliary Effect Slot objects
1.0	alDeleteAuxiliaryEffectSlots	Delete *n* Auxiliary Effect Slots objects
1.0	alIsAuxiliaryEffectSlot	Validate an Auxiliary Effect Slot identifier

Property Functions

Auxiliary Effect Slot properties are set using the following functions:

Version	Function Name	Purpose
1.0	alAuxiliaryEffectSloti	Set integer property
1.0	alAuxiliaryEffectSlotiv	Set integer array property
1.0	alAuxiliaryEffectSlotf	Set floating point property
1.0	alAuxiliaryEffectSlotfv	Set floating point array property

Query Property Functions

An application can query properties of the Auxiliary Effect Slot object using the following functions:

Version	Function Name	Purpose
1.0	alGetAuxiliaryEffectSloti	Retrieve an integer property
1.0	alGetAuxiliaryEffectSlotiv	Retrieve an integer array property
1.0	alGetAuxiliaryEffectSlotf	Retrieve a floating point property
1.0	alGetAuxiliaryEffectSlotfv	Retrieve a floating point array property

Properties

To manipulate properties of the Auxiliary Effect Slot the following values are defined:

Version	Property	Range	Default
1.0	AL_EFFECTSLOT_EFFECT	Effect object id	AL_EFFECT_NULL
1.0	AL_EFFECTSLOT_GAIN	0.0 to 1.0	1.0
1.0	AL_EFFECTSLOT_AUXILIARY_SEND_AUTO	AL_TRUE or AL_FALSE	AL_TRUE

alGenAuxiliaryEffectSlots

The **alGenAuxiliaryEffectSlots** function is used to create one or more Auxiliary Effect Slots. The number of slots that can be created will be dependant upon the Open AL device used.

```
ALvoid alGenAuxiliaryEffectSlots(
      ALsizei n,
      ALuint* auxiliaryeffectslots
);
```

Parameters

n
: Number of Auxiliary Effect Slots to be created.

auxiliaryeffectslots
: Pointer addressing sufficient memory to store *n* Effect Slot object identifiers.

Possible Error States

State	Problem(s)
AL_OUT_OF_MEMORY	Not enough resources to complete request
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

An application should check the OpenAL error state after making this call to determine if the Effect Slot was successfully created. If the function call fails then none of the requested Effect Slots are created.
A good strategy for creating any OpenAL object is to use a for-loop and generate one object each loop iteration and then check for an error condition. If an error is set then the loop can be broken and the application can determine if sufficient resources are available.

See also

alDeleteAuxiliaryEffectSlots, alIsAuxiliaryEffectSlot

alDeleteAuxiliaryEffectSlots

The **alDeleteAuxiliaryEffectSlots** function is used to delete and free resources for Auxiliary Effect Slots previously created with alGenAuxiliaryEffectSlots.

```
ALvoid alDeleteAuxiliaryEffectSlots(
      ALsizei n,
      ALuint* auxiliaryeffectslots
);
```

Parameters

n
: Number of Auxiliary Effect Slots to be deleted.

auxiliaryeffectslots
: Pointer to *n* Effect Slot object identifiers.

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

None.

See also

alGenAuxiliaryEffectSlots, alIsAuxiliaryEffectSlot

alIsAuxiliaryEffectSlot

The **alIsAuxiliaryEffectSlot** function is used to determine if an object identifier is a valid Auxiliary Effect Slot object.

```
ALboolean alIsAuxiliaryEffectSlot(
      ALuint auxiliaryeffectslot
);
```

Parameters

auxiliaryeffectslot
: Effect Slot object identifier to validate.

Returns

AL_TRUE if the identifier is a valid Auxiliary Effect Slot, AL_FALSE otherwise.

Version Requirements

Effects extension version 1.0

Remarks

None.

See also

alGenAuxiliaryEffectSlots, alDeleteAuxiliaryEffectSlots

alAuxiliaryEffectSlot[i,iv,f,fv]

The **alAuxiliaryEffectSlot[i,iv,f,fv]** functions are used to set properties on Auxiliary Effect Slot objects.

```
ALvoid alAuxiliaryEffectSloti(
      ALuint auxiliaryeffectslot,
      ALenum param,
      ALint iValue
);
```

```
ALvoid alAuxiliaryEffectSlotiv(
      ALuint auxiliaryeffectslot,
      ALenum param,
      ALint* piValues
);
```

```
ALvoid alAuxiliaryEffectSlotf(
      ALuint auxiliaryeffectslot,
      ALenum param,
      ALfloat flValue
);
```

```
ALvoid alAuxiliaryEffectSlotfv(
      ALuint auxiliaryeffectslot,
      ALenum param,
      ALfloat* pflValues
);
```

Parameters

auxiliaryeffectslot
: Auxiliary Effect Slot object identifier.

param
: Auxiliary Effect Slot property to set.

iValue
: Integer value

piValues
: Pointer to array of integer values

flValue

Floating point value

pflValues
Pointer to array of floating point values

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_ENUM	An unknown property identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

The result of these functions is determined by the property enumeration value. The available properties that can be set include:

Version	Property	Range	Default
1.0	AL_EFFECTSLOT_EFFECT	Effect object id	AL_EFFECT_NULL
1.0	AL_EFFECTSLOT_GAIN	0.0 to 1.0	1.0
1.0	AL_EFFECTSLOT_AUXILIARY_SEND_AUTO	AL_TRUE or AL_FALSE	AL_TRUE

See also

alGetAuxiliaryEffectSlot[i,iv,f,fv]

alGetAuxiliaryEffectSlot[i,iv,f,fv]

The **alGetAuxiliaryEffectSlot[i,iv,f,fv]** functions are used to retrieve properties on Auxiliary Effect Slot objects.

```
ALvoid alGetAuxiliaryEffectSloti(
      ALuint auxiliaryeffectslot,
      ALenum param,
      ALint* piValue
);
```

```
ALvoid alGetAuxiliaryEffectSlotiv(
      ALuint auxiliaryeffectslot,
      ALenum param,
      ALint* piValues
);
```

```
ALvoid alGetAuxiliaryEffectSlotf(
      ALuint auxiliaryeffectslot,
      ALenum param,
      ALfloat* pflValue
);
```

```
ALvoid alGetAuxiliaryEffectSlotfv(
      ALuint auxiliaryeffectslot,
      ALenum param,
      ALfloat* pflValues
);
```

Parameters

auxiliaryeffectslot
: Auxiliary Effect Slot object identifier.

param
: Auxiliary Effect Slot property to retrieve.

piValue
: Address where integer value will be stored.

piValues
: Address where the array of integers will be stored.

pflValue

Address where floating point value will be stored.

pflValues
Address where the array of floating point values will be stored.

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_ENUM	An unknown property identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

The result of these functions is determined by the property enumeration value. The available properties that can be retrieved include:

Version	Property	Range	Default
1.0	AL_EFFECTSLOT_EFFECT	{Effect object id}	AL_EFFECT_NULL
1.0	AL_EFFECTSLOT_GAIN	0.0 to 1.0	1.0
1.0	AL_EFFECTSLOT_AUXILIARY_SEND_AUTO	AL_TRUE or AL_FALSE	AL_TRUE

See also

alAuxiliaryEffectSlot[i,iv,f,fv]

AL_EFFECTSLOT_EFFECT

Version	Property	Range	Default
1.0	AL_EFFECTSLOT_EFFECT	{Effect object id}	AL_EFFECT_NULL

The **AL_EFFECTSLOT_EFFECT** property is used to attach an Effect object to the Auxiliary Effect Slot object. After the attachment, the Auxiliary Effect Slot object will contain the effect type and have the same effect parameters that were stored in the Effect object. Any Sources feeding the Auxiliary Effect Slot will immediate feed the new effect type and new effect parameters.

Example: Create an Auxiliary Effect Slot and attaching an Effect.

```
ALuint uiEffectSlot;    /* Storage for effect slot object ID.*/
ALuint uiEffect;        /* Storage for effect object ID.*/

/* Generate an Aux Effect Slot */
alGenAuxiliaryEffectSlots(1, &uiEffectSlot);

/* Generate an Effect and set its type to reverb */
alGenEffects(1, &uiEffect);
alEffecti(uiEffect, AL_EFFECT_TYPE, AL_EFFECT_REVERB);

/* Attach Effect to Aux Effect Slot */
alAuxiliaryEffectSloti(uiEffectSlot,
      AL_EFFECTSLOT_EFFECT, uiEffect);
```

AL_EFFECTSLOT_GAIN

Version	Property	Range	Default
1.0	AL_EFFECTSLOT_GAIN	0.0 to 1.0	1.0

The **AL_EFFECTSLOT_GAIN** property is used to specify an output level for the Auxiliary Effect Slot. Setting the gain to 0.0 mutes the output.

Example: Setting Auxiliary Slot Gain

```
/* Storage for effect slot object ID.*/
ALuint uiEffectSlot;

/* Generate an Aux Effect Slot */
alGenAuxiliaryEffectSlots(1, &uiEffectSlot);

/* Set Aux Effect Slot Gain to 0.5*/
alAuxiliaryEffectSlotf(uiEffectSlot,
      AL_EFFECTSLOT_GAIN, 0.5f);
```

AL_EFFECTSLOT_AUXILIARY_SEND_AUTO

Version	Property	Range	Default
1.0	AL_EFFECTSLOT_AUXILIARY_SEND_AUTO	AL_TRUE or AL_FALSE	AL_TRUE

The **AL_EFFECTSLOT_AUXILIARY_SEND_AUTO** property is used to enable or disable automatic send adjustments based on the physical positions of the sources and the listener.

This property should be enabled when an application wishes to use a reverb effect to simulate the environment surrounding a listener or a collection of Sources.

Example: Setting the Auxiliary Slot Send Auto Flag

```
/* Storage for effect slot object ID.*/
ALuint uiEffectSlot;

/* Generate an Aux Effect Slot */
alGenAuxiliaryEffectSlots(1, &uiEffectSlot);

/* Set Aux Effect Slot Send Auto flag to true */
alAuxiliaryEffectSloti(uiEffectSlot,
      AL_EFFECTSLOT_AUXILIARY_SEND_AUTO, AL_TRUE);
```

Effect Object

Management Functions

To handle the creation, deletion, and verification of Effect Objects the following functions are supplied:

Version	Function Name	Purpose
1.0	alGenEffects	Create *n* Effect objects
1.0	alDeleteEffects	Delete *n* Effect objects
1.0	alIsEffect	Validate an Effect identifier

Property Functions

Effect Object properties are set using the following functions:

Version	Function Name	Purpose
1.0	alEffecti	Set integer property
1.0	alEffectiv	Set integer array property
1.0	alEffectf	Set floating point property
1.0	alEffectfv	Set floating point array property

Query Property Functions

An application can query properties of the Effect Object using the following functions:

Version	Function Name	Purpose
1.0	alGetEffecti	Retrieve an integer property
1.0	alGetEffectiv	Retrieve an integer array property
1.0	alGetEffectf	Retrieve a floating point property
1.0	alGetEffectfv	Retrieve a floating point array property

Properties

To manipulate properties of the Effect the following values are defined:

Version	Property	Range	Default
1.0	AL_EFFECT_TYPE	EffectType Enum	AL_EFFECT_NULL
1.0	*AL_EFFECT_PARAMETER_NAME*	Variable	Variable

alGenEffects

The **alGenEffects** function is used to create one or more Effect objects. An Effect object stores an effect type and a set of parameter values to control that Effect. In order to use an Effect it must be attached to an Auxiliary Effect Slot object.

```
ALvoid alGenEffects(
      ALsizei n,
      ALuint* effects
);
```

Parameters

n
: Number of Effects to be created.

effects
: Pointer addressing sufficient memory to store *n* Effect object identifiers.

Possible Error States

State	Problem(s)
AL_OUT_OF_MEMORY	Not enough resources to complete request
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

After creation an Effect has no type (AL_EFFECT_NULL), so before it can be used to store a set of parameters, the application must specify what type of effect should be stored in the object, using alEffecti.

See also

alDeleteEffects, alIsEffect, alEffecti

alDeleteEffects

The **alDeleteEffects** function is used to delete and free resources for Effect objects previously created with alGenEffects.

```
ALvoid alDeleteEffects(
      ALsizei n,
      ALuint* effects
);
```

Parameters

n
: Number of Effects to be deleted.

effects
: Pointer to *n* Effect object identifiers.

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

None.

See also

alGenEffects, alIsEffect

alIsEffect

The **alIsEffect** function is used to determine if an object identifier is a valid Effect object.

```
ALboolean alIsEffect(
      ALuint effect
);
```

Parameters

effect
: Effect identifier to validate.

Returns

AL_TRUE if the identifier is a valid Effect, AL_FALSE otherwise.

Version Requirements

Effects extension version 1.0

Remarks

None.

See also

alGenEffects, alDeleteEffects

alEffect[i,iv,f,fv]

The **alEffect[i,iv,f,fv]** functions are used to set properties on Effect objects.

```
ALvoid alEffecti(
      ALuint effect,
      ALenum param,
      ALint iValue
);
```

```
ALvoid alEffectiv(
      ALuint effect,
      ALenum param,
      ALint* piValues
);
```

```
ALvoid alEffectf(
      ALuint effect,
      ALenum param,
      ALfloat flValue
);
```

```
ALvoid alEffectfv(
      ALuint effect,
      ALenum param,
      ALfloat* pflValues
);
```

Parameters

effect
: Effect object identifier.

param
: Effect property to set.

iValue
: Integer value

piValues
: Pointer to array of integer values

flValue
: Floating point value

pflValues

Pointer to array of floating point values

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_ENUM	An unknown property identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

The result of these functions is determined by the property enumeration value. The available properties that can be set include:

Version	Property	Range	Default
1.0	AL_EFFECT_TYPE	EffectType Enum	AL_EFFECT_NULL
1.0	*AL_EFFECT_PARAMETER_NAME*	Variable	Variable

See also

alGetEffect[i,iv,f,fv]

alGetEffect[i,iv,f,fv]

The **alGetEffect[i,iv,f,fv]** functions are used to retrieve properties from Effect objects.

```
ALvoid alGetEffecti(
      ALuint effect,
      ALenum param,
      ALint* piValue
);
```

```
ALvoid alGetEffectiv(
      ALuint effect,
      ALenum param,
      ALint* piValues
);
```

```
ALvoid alGetEffectf(
      ALuint effect,
      ALenum param,
      ALfloat* pflValue
);
```

```
ALvoid alGetEffectfv(
      ALuint effect,
      ALenum param,
      ALfloat* pflValues
);
```

Parameters

effect
: Effect object identifier.

param
: Effect property to retrieve.

piValue
: Address where integer value will be stored.

piValues
: Address where the array of integers will be stored.

pflValue
: Address where floating point value will be stored.

pflValues

Address where the array of floating point values will be stored.

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_ENUM	An unknown property identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

The result of these functions is determined by the property enumeration value. The available properties that can be retrieved include:

Version	Property	Range	Default
1.0	AL_EFFECT_TYPE	EffectType Enum	AL_EFFECT_NULL
1.0	AL_*EFFECT_PARAMETER_NAME*	Variable	Variable

See also

alEffect[i,iv,f,fv]

AL_EFFECT_TYPE

Version	Property	Range	Default
1.0	AL_EFFECT_TYPE	EffectType Enum	AL_EFFECT_NULL

The AL_EFFECT_TYPE is used to set the type of effect represented by the Effect object. When an Effect Object is first created it is of type AL_EFFECT_NULL. A NULL effect is an effect that has no parameters and does nothing. The following effect type values are defined:

Version	Effect Type	Value
1.0	AL_EFFECT_NULL	0x0000
1.0	AL_EFFECT_EAXREVERB	0x8000
1.0	AL_EFFECT_REVERB	0x0001
1.0	AL_EFFECT_CHORUS	0x0002
1.0	AL_EFFECT_DISTORTION	0x0003
1.0	AL_EFFECT_ECHO	0x0004
1.0	AL_EFFECT_FLANGER	0x0005
1.0	AL_EFFECT_FREQUENCY_SHIFTER	0x0006
1.0	AL_EFFECT_VOCAL_MORPHER	0x0007
1.0	AL_EFFECT_PITCH_SHIFTER	0x0008
1.0	AL_EFFECT_RING_MODULATOR	0x0009
1.0	AL_EFFECT_AUTOWAH	0x000A
1.0	AL_EFFECT_COMPRESSOR	0x000B
1.0	AL_EFFECT_EQUALIZER	0x000C

An application can set the effect type using the alEffecti function. Below is a code example showing how an effect object is created using alGenEffects, and its type set using alEffecti with the AL_EFFECT_TYPE property:

```
ALuint uiEffect; /* Storage for effect object ID.*/

/* Generate an effect ID and then set its type to reverb. Setting
the effect type sets all parameters to default values.*/

alGenEffects(1, &uiEffect);

alEffecti(uiEffect, AL_EFFECT_TYPE, AL_EFFECT_REVERB);
```

When the effect type has been set, the effect parameters will set to their default values.

An application can retrieve the effect type from an Effect Object using the alGetEffecti function.

```
Alint iEffectType; /* Storage for effect type retrieved */

alGetEffecti(uiEffect, AL_EFFECT_TYPE, &iEffectType);
```

AL_EFFECT_PARAMETER_NAME

The Effect object can store many different types of audio effects, each with their own set of parameters types and ranges. The parameters of these effects are all set using the generic alEffect[i,iv,f,fv] function calls. This section details the names and parameters of the effects that may be available on an Effects Extension v.1.0 capable device.

AL_EFFECT_NULL

As previously described an effect object with an effect type of AL_EFFECT_NULL has no parameters and does nothing. This effect type is used when an Effect Object is initially created. Attach a NULL effect to an auxiliary effect slot to unload any effect that might be occupying the slot and free the associated resources. Attempting to set a parameter value on a NULL effect will result in an error condition being set.

AL_EFFECT_EAXREVERB

Version	Parameter Name	Units	Range	Default
1.0	AL_EAXREVERB_DENSITY		[0.0, 1.0]	1.0
1.0	AL_EAXREVERB_DIFFUSION		[0.0, 1.0]	1.0
1.0	AL_EAXREVERB_GAIN		[0.0, 1.0]	0.32
1.0	AL_EAXREVERB_GAINHF		[0.0, 1.0]	0.89
1.0	AL_EAXREVERB_GAINLF		[0.0, 1.0]	0.0
1.0	AL_EAXREVERB_DECAY_TIME	Seconds	[0.1, 20]	1.49
1.0	AL_EAXREVERB_DECAY_HFRATIO		[0.1, 2.0]	0.83
1.0	AL_EAXREVERB_DECAY_LFRATIO		[0.1, 2.0]	1.0
1.0	AL_EAXREVERB_REFLECTIONS_GAIN		[0.0, 3.16]	0.05
1.0	AL_EAXREVERB_REFLECTIONS_DELAY	Seconds	[0.0, 0.3]	0.007
1.0	AL_EAXREVERB_REFLECTIONS_PAN	Vector	[[-1.0, -1.0, -1.0], [1.0, 1.0, 1.0]]	[0.0, 0.0, 0.0]
1.0	AL_EAXREVERB_LATE_REVERB_GAIN		[0.0, 10.0]	1.26
1.0	AL_EAXREVERB_LATE_REVERB_DELAY	Seconds	[0.0, 0.1]	0.011
1.0	AL_EAXREVERB_LATE_REVERB_PAN	Vector	[[-1.0, -1.0, -1.0], [1.0, 1.0, 1.0]]	[0.0, 0.0, 0.0]
1.0	AL_EAXREVERB_ECHO_TIME		[0.075, 0.25]	0.25
1.0	AL_EAXREVERB_ECHO_DEPTH		[0.0, 1.0]	0.0
1.0	AL_EAXREVERB_MODULATION_TIME		[0.04, 4.0]	0.25
1.0	AL_EAXREVERB_MODULATION_DEPTH		[0.0, 1.0]	0.0
1.0	AL_EAXREVERB_AIR_ABSORPTION_GAINHF		[0.892,1.0]	0.994
1.0	AL_EAXREVERB_HFREFERENCE	Hz	[1000.0 20000.0]	5000.0
1.0	AL_EAXREVERB_LFREFERENCE	Hz	[20.0, 1000.0]	250.0
1.0	AL_EAXREVERB_ROOM_ROLLOFF_FACTOR		[0.0. 10.0]	0.0
1.0	AL_EAXREVERB_DECAYHF_LIMIT	ON / OFF	[AL_FALSE, AL_TRUE]	AL_TRUE

AL_EFFECT_REVERB

Version	Parameter Name	Units	Range	Default
1.0	AL_REVERB_DENSITY		[0.0, 1.0]	1.0
1.0	AL_REVERB_DIFFUSION		[0.0, 1.0]	1.0

1.0	AL_REVERB_GAIN		[0.0, 1.0]	0.32
1.0	AL_REVERB_GAINHF		[0.0, 1.0]	0.89
1.0	AL_REVERB_DECAY_TIME	Seconds	[0.1, 20]	1.49
1.0	AL_REVERB_DECAY_HFRATIO		[0.1, 2.0]	0.83
1.0	AL_REVERB_REFLECTIONS_GAIN		[0.0, 3.16]	0.05
1.0	AL_REVERB_REFLECTIONS_DELAY	Seconds	[0.0, 0.3]	0.007
1.0	AL_REVERB_LATE_REVERB_GAIN		[0.0, 10.0]	1.26
1.0	AL_REVERB_LATE_REVERB_DELAY	Seconds	[0.0, 0.1]	0.011
1.0	AL_REVERB_AIR_ABSORPTION_GAINHF		[0.892,1.0]	0.994
1.0	AL_REVERB_ROOM_ROLLOFF_FACTOR		[0.0. 10.0]	0.0
1.0	AL_REVERB_DECAY_HFLIMIT	ON / OFF	[AL_FALSE, AL_TRUE]	AL_TRUE

AL_EFFECT_CHORUS

Version	Parameter Name	Units	Range	Default
1.0	AL_CHORUS_WAVEFORM	Sin, Triangle	[AL_CHORUS_WAVEFORM_SINUSOID, AL_CHORUS_WAVEFORM_TRIANGLE]	Triangle
1.0	AL_CHORUS_PHASE	Degrees	[-180, 180]	90
1.0	AL_CHORUS_RATE	Hz	[0.0, 10.0]	1.1
1.0	AL_CHORUS_DEPTH		[0.0, 1.0]	0.1
1.0	AL_CHORUS_FEEDBACK		[-1.0, 1.0]	0.25
1.0	AL_CHORUS_DELAY	Seconds	[0.0, 0.016]	0.016

AL_EFFECT_DISTORTION

Version	Parameter Name	Units	Range	Default
1.0	AL_DISTORTION_EDGE		[0.0, 1.0]	0.2
1.0	AL_DISTORTION_GAIN		[0.01, 1.0]	0.05
1.0	AL_DISTORTION_LOWPASS_CUTOFF	Hz	[80.0, 24000]	8000
1.0	AL_DISTORTION_EQCENTER	Hz	[80.0, 24000]	3600
1.0	AL_DISTORTION_EQBANDWIDTH	Hz	[80.0, 24000]	3600

AL_EFFECT_ECHO

Version	Parameter Name	Units	Range	Default
1.0	AL_ECHO_DELAY	Seconds	[0.0, 0.207]	0.1
1.0	AL_ECHO_LRDELAY	Seconds	[0.0, 0.404]	0.1
1.0	AL_ECHO_DAMPING		[0.0, 0.99]	0.5
1.0	AL_ECHO_FEEDBACK		[0.0, 1.0]	0.5
1.0	AL_ECHO_SPREAD		[-1.0, 1.0]	-1.0

AL_EFFECT_FLANGER

Version	Parameter Name	Units	Range	Default
1.0	AL_FLANGER_WAVEFORM	Sin, Triangle	[AL_FLANGER_WAVEFORM_SINUSOID,	Triangle

			AL_FLANGER_WAVEFORM_TRIANGLE]	
1.0	AL_FLANGER_PHASE		[-180, 180]	0
1.0	AL_FLANGER_RATE		[0.0, 10.0]	0.27
1.0	AL_FLANGER_DEPTH		[0.0, 1.0]	1.0
1.0	AL_FLANGER_FEEDBACK		[-1.0, 1.0]	-0.5
1.0	AL_FLANGER_DELAY	Seconds	[0.0, 0.004]	0.002

AL_EFFECT_FREQUENCY_SHIFTER

Version	Parameter Name	Units	Range	Default
1.0	AL_FREQUENCY_SHIFTER_FREQUENCY	Hz	[0.0, 24000.0]	0.0
1.0	AL_FREQUENCY_SHIFTER_LEFT_DIRECTION	Down, Up, Off	[0, 2]	0
1.0	AL_FREQUENCY_SHIFTER_RIGHT_DIRECTION	Down, Up, Off	[0, 2]	0

AL_EFFECT_VOCAL_MORPHER

Version	Parameter Name	Units	Range	Default
1.0	AL_VOCAL_MORPHER_PHONEMEA	Phoneme Type	[0, 29]	0
1.0	AL_VOCAL_MORPHER_PHONEMEB	Phoneme Type	[0, 29]	10
1.0	AL_VOCAL_MORPHER_PHONEMEA_COARSE_TUNING	Semitones	[-24, 24]	0
1.0	AL_VOCAL_MORPHER_PHONEMEB_COARSE_TUNING	Semitones	[-24, 24]	0
1.0	AL_VOCAL_MORPHER_WAVEFORM	Sin, Triangle, Saw	[0, 2]	0
1.0	AL_VOCAL_MORPHER_RATE	Hz	[0.0, 10.0]	1.41

AL_EFFECT_PITCH_SHIFTER

Version	Parameter Name	Units	Range	Default
1.0	AL_PITCH_SHIFTER_COARSE_TUNE	Semitones	[-12, 12]	12
1.0	AL_PITCH_SHIFTER_FINE_TUNE	Cents	[-50, 50]	0

AL_EFFECT_RING_MODULATOR

Version	Parameter Name	Units	Range	Default
1.0	AL_RING_MODULATOR_FREQUENCY	Hz	[0.0, 8000.0]	440.0
1.0	AL_RING_MODULATOR_HIGHPASS_CUTOFF	Hz	[0.0. 24000.0]	800.0
1.0	AL_RING_MODULATOR_WAVEFORM	Sin, Saw, Square	[0, 2]	0

AL_EFFECT_AUTOWAH

Version	Parameter Name	Units	Range	Default
1.0	AL_AUTOWAH_ATTACK_TIME	Sec	[0.0001, 1.0]	0.06
1.0	AL_AUTOWAH_RELEASE_TIME	Sec	[0.0001, 1.0]	0.06
1.0	AL_AUTOWAH_RESONANCE		[2.0, 1000.0]	1000.0
1.0	AL_AUTOWAH_PEAK_GAIN		[0.00003, 31621.0]	11.22

AL_EFFECT_COMPRESSOR

Version	Parameter Name	Units	Range	Default
1.0	AL_COMPRESSOR_ONOFF	On, Off	[0, 1]	1

AL_EFFECT_EQUALIZER

Version	Parameter Name	Units	Range	Default
1.0	AL_EQUALIZER_LOW_GAIN		[0.126, 7.943]	1.0
1.0	AL_EQUALIZER_LOW_CUTOFF	Hz	[50.0, 800.0]	200.0
1.0	AL_EQUALIZER_MID1_GAIN		[0.126, 7.943]	1.0
1.0	AL_EQUALIZER_MID1_CENTER	Hz	[200.0, 3000.0]	500.0
1.0	AL_EQUALIZER_MID1_WIDTH		[0.01, 1.0]	1.0
1.0	AL_EQUALIZER_MID2_GAIN		[0.126, 7.943]	1.0
1.0	AL_EQUALIZER_MID2_CENTER		[1000.0, 8000.0]	3000.0
1.0	AL_EQUALIZER_MID2_WIDTH		[0.01, 1.0]	1.0
1.0	AL_EQUALIZER_HIGH_GAIN		[0.126, 7.943]	1.0
1.0	AL_EQUALIZER_HIGH_CUTOFF	Hz	[4000.0, 16000.0]	6000.0

Filter Object

Management Functions

To handle the creation, deletion, and verification of Filter Objects the following functions are supplied:

Version	Function Name	Purpose
1.0	alGenFilters	Create *n* Filter objects
1.0	alDeleteFilters	Delete *n* Filter objects
1.0	alIsFilter	Validate a Filter identifier

Property Functions

Filter Object properties are set using the following functions:

Version	Function Name	Purpose
1.0	alFilteri	Set integer property
1.0	alFilteriv	Set integer array property
1.0	alFilterf	Set floating point property
1.0	alFilterfv	Set floating point array property

Query Property Functions

An application can query properties of the Filter Object using the following functions:

Version	Function Name	Purpose
1.0	alGetFilteri	Retrieve an integer property
1.0	alGetFilteriv	Retrieve an integer array property
1.0	alGetFilterf	Retrieve a floating point property
1.0	alGetFilterfv	Retrieve a floating point array property

Properties

To manipulate properties of the Filter the following values are defined:

Version	Property	Range	Default
1.0	AL_FILTER_TYPE	FilterType Enum	AL_FILTER_NULL
1.0	*AL_FILTER_PARAMETER_NAME*	Variable	Variable

alGenFilters

The **alGenFilters** function is used to create one or more Filter objects. A Filter object stores a filter type and a set of parameter values to control that Filter. Filter objects can be attached to Sources as Direct Filters or Auxiliary Send Filters.

```
ALvoid alGenFilters(
      ALsizei n,
      ALuint* filters
);
```

Parameters

n
: Number of Filters to be created.

filters
: Pointer addressing sufficient memory to store *n* Filter object identifiers.

Possible Error States

State	Problem(s)
AL_OUT_OF_MEMORY	Not enough resources to complete request
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

After creation a Filter has no type (AL_FILTER_NULL), so before it can be used to store a set of parameters, the application must specify what type of filter should be stored in the object, using alFilteri.

See also

alDeleteFilters, alIsFilter, alFilteri

alDeleteFilters

The **alDeleteFilters** function is used to delete and free resources for Filter objects previously created with alGenFilters.

```
ALvoid alDeleteFilters(
      ALsizei n,
      ALuint* filters
);
```

Parameters

n
: Number of Filters to be deleted.

filters
: Pointer to *n* Filter object identifiers.

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

None.

See also

alGenFilters, alIsFilter

alIsFilter

The **alIsFilter** function is used to determine if an object identifier is a valid Filter object.

```
ALboolean alIsFilter(
      ALuint filter
);
```

Parameters

filter
: Effect identifier to validate.

Returns

AL_TRUE if the identifier is a valid Filter, AL_FALSE otherwise.

Version Requirements

Effects extension version 1.0

Remarks

None.

See also

alGenFilters, alDeleteFilters

alFilter[i,iv,f,fv]

The **alFilter[i,iv,f,fv]** functions are used to set properties on Filter objects.

```
ALvoid alFilteri(
      ALuint filter,
      ALenum param,
      ALint iValue
);
```

```
ALvoid alFilteriv(
      ALuint filter,
      ALenum param,
      ALint* piValues
);
```

```
ALvoid alFilterf(
      ALuint filter,
      ALenum param,
      ALfloat flValue
);
```

```
ALvoid alFilterfv(
      ALuint filter,
      ALenum param,
      ALfloat* pflValues
);
```

Parameters

filter
: Filter object identifier.

param
: Effect property to set.

iValue
: Integer value

piValues
: Pointer to array of integer values

flValue
: Floating point value

pflValues

Pointer to array of floating point values

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_ENUM	An unknown property identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

The result of these functions is determined by the property enumeration value. The available properties that can be set include:

Version	Property	Range	Default
1.0	AL_FILTER_TYPE	FilterType Enum	AL_FILTER_NULL
1.0	*AL_FILTER_PARAMETER_NAME*	Variable	Variable

See also

alGetFilter[i,iv,f,fv]

alGetFilter[i,iv,f,fv]

The **alGetFilter[i,iv,f,fv]** functions are used to retrieve properties from Filter objects.

```
ALvoid alGetFilteri(
      ALuint filter,
      ALenum param,
      ALint* piValue
);
```

```
ALvoid alGetFilteriv(
      ALuint filter,
      ALenum param,
      ALint* piValues
);
```

```
ALvoid alGetFilterf(
      ALuint filter,
      ALenum param,
      ALfloat* pflValue
);
```

```
ALvoid alGetFilterfv(
      ALuint filter,
      ALenum param,
      ALfloat* pflValues
);
```

Parameters

filter
: Filter object identifier.

param
: Effect property to retrieve.

piValue
: Address where integer value will be stored.

piValues
: Address where the array of integers will be stored.

pflValue
: Address where floating point value will be stored.

pflValues

Address where the array of floating point values will be stored.

Possible Error States

State	Problem(s)
AL_INVALID_NAME	An unknown object identifier was found
AL_INVALID_ENUM	An unknown property identifier was found
AL_INVALID_OPERATION	Operation cannot be completed at this time
AL_INVALID_VALUE	Bad value passed to function

Version Requirements

Effects extension version 1.0

Remarks

The result of these functions is determined by the property enumeration value. The available properties that can be retrieved include:

Version	Property	Range	Default
1.0	AL_FILTER_TYPE	FilterType Enum	AL_FILTER_NULL
1.0	*AL_FILTER_PARAMETER_NAME*	Variable	Variable

See also

alFilter[i,iv,f,fv]

AL_FILTER_TYPE

Version	Property	Range	Default
1.0	AL_FILTER_TYPE	FilterType Enum	AL_FILTER_NULL

The AL_FILTER_TYPE is used to set the type of filter represented by the Filter object. When a Filter object is first created it is of type AL_FILTER_NULL. A NULL filter is a filter that has no parameters and does nothing. The following filter type values are defined:

Version	Effect Type	Value
1.0	AL_FILTER_NULL	0x0000
1.0	AL_FILTER_LOWPASS	0x0001
1.0	AL_FILTER_HIGHPASS	0x0002
1.0	AL_FILTER_BANDPASS	0x0003

An application can set the filter type using the alFilteri function. Below is a code example showing how a Filter object is created using alGenFilters, and its type set using alFilteri with the AL_FILTER_TYPE property:

```
ALuint uiFilter; /* Storage for filter object ID.*/

/* Generate a filter ID and then set its type to low-pass.
Setting the filter type sets all parameters to default values.*/

alGenFilters(1, &uiFilter);

alFilteri(uiFilter, AL_FILTER_TYPE, AL_FILTER_LOWPASS);
```

When the filter type has been set, the filter parameters will be set to their default values.

An application can retrieve the filter type from a Filter object using the alGetFilteri function.

```
Alint iFilterType; /* Storage for filter type retrieved */

alGetFilteri(uiFilter, AL_FILTER_TYPE, &iFilterType);
```

AL_FILTER_PARAMETER_NAME

The Filter object can store different types of Filters, each with their own set of parameters types and ranges. The parameters of these filters are all set using the generic alFilter[i,iv,f,fv] function calls. This section details the names and parameters of the effects that may be available on an Effects Extension v.1.0 capable device

AL_FILTER_NULL

As previously described a Filter object with a type of AL_FILTER_NULL has no parameters. This Filter type is used when a Filter object is initially created. Attempting to set a parameter value on a NULL effect will result in an error condition being set.

AL_FILTER_LOWPASS

A low-pass filter is used to remove high frequency content from a signal.

Parameter Name	Type	Units	Range	Default
AL_LOWPASS_GAIN	FLOAT		[0.0, 1.0]	1.0
AL_LOWPASS_GAINHF	FLOAT		[0.0, 1.0]	1.0

AL_FILTER_HIGHPASS

Not currently implemented. A high-pass filter is used to remove low frequency content from a signal.

Parameter Name	Type	Units	Range	Default
AL_HIGHPASS_GAIN	FLOAT		[0.0, 1.0]	1.0
AL_HIGHPASS_GAINLF	FLOAT		[0.0, 1.0]	1.0

AL_FILTER_BANDPASS

Not currently implemented. A band-pass filter is used to remove high and low frequency content from a signal.

Parameter Name	Type	Units	Range	Default
AL_BANDPASS_GAIN	FLOAT		[0.0, 1.0]	1.0
AL_BANDPASS_GAINLF	FLOAT		[0.0, 1.0]	1.0
AL_BANDPASS_GAINHF	FLOAT		[0.0, 1.0]	1.0

Source Object Extensions

Properties

The following Source properties are defined:

Version	Property	Range	Default
1.0	AL_DIRECT_FILTER	Filter ID	AL_FILTER_NULL
1.0	AL_AUXILIARY_SEND_FILTER	Filter ID	AL_FILTER_NULL
1.0	AL_AIR_ABSORPTION_FACTOR	0 to 10	0
1.0	AL_ROOM_ROLLOFF_FACTOR	0 to 10	0
1.0	AL_CONE_OUTER_GAINHF	0.0 to 1.0	1.0
1.0	AL_DIRECT_FILTER_GAINHF_AUTO	AL_FALSE or AL_TRUE	AL_TRUE
1.0	AL_AUXILIARY_SEND_FILTER_GAIN_AUTO	AL_FALSE or AL_TRUE	AL_TRUE
1.0	AL_AUXILIARY_SEND_FILTER_GAINHF_AUTO	AL_FALSE or AL_TRUE	AL_TRUE

AL_DIRECT_FILTER

Version	Property	Range	Default
1.0	AL_DIRECT_FILTER	Filter ID	AL_FILTER_NULL

This Source property is used to apply filtering on the direct-path (dry signal) of a Source. A previously generated Filter object can be attached to the Source using alSourcei with the property AL_DIRECT_FILTER.

To attach Filter 'uiFilterID' to Source 'uiSourceID':

```
alSourcei(uiSourceID, AL_DIRECT_FILTER, uiFilterID);
```

To remove a Filter from a Source, the application should attach the null Filter object to the Source.

To remove a Filter from Source 'uiSourceID':

```
alSourcei(uiSourceID, AL_DIRECT_FILTER, AL_FILTER_NULL);
```

AL_AUXILIARY_SEND_FILTER

Version	Property	Range	Default
1.0	AL_AUXILIARY_SEND_FILTER	N/A	N/A

This Source property is used to establish connections between Sources and Auxiliary Effect Slots. For a Source to feed an Effect that has been loaded into an Auxiliary Effect Slot the application must configure one of the Source's auxiliary sends. This process involves setting 3 variables – the destination Auxiliary Effect Slot ID, the Auxiliary Send number, and an optional Filter ID.

The ID of the Auxiliary Effect Slot is simply the value returned from a successful call to alGenAuxiliaryEffectSlots.

The Auxiliary Send number identifies which of the Source's Auxiliary Sends is being used to send to the specified Auxiliary Effect Slot. The number of Auxiliary Sends available on each Source is OpenAL device dependent (see ALC_MAX_AUXILIARY_SENDS).

If an application wishes to filter the send from the Source to the Auxiliary Effect Slot it can provide a valid Filter ID. If no filtering is required this value should be set to AL_FILTER_NULL.

The alSource3i function call is used to pass the values to OpenAL using the property AL_AUXILIARY_SEND_FILTER.

To configure Source 'uiSourceID' to send to Auxiliary Effect Slot 'uiEffectSlot[0]' using auxiliary send 0 without filtering:

```
alSource3i(uiSourceID,AL_AUXILIARY_SEND_FILTER,
uiEffectSlot[0], 0, AL_FILTER_NULL);
```

To configure Source 'uiSourceID' to send to Auxiliary Effect Slot 'uiEffectSlot[1]' using auxiliary send 1 with Filter uiFilterID:

```
alSource3i(uiSourceID,AL_AUXILIARY_SEND_FILTER,
uiEffectSlot[1], 1, uiFilterID);
```

To disable a particular Auxiliary Send from a Source, the application should configure that send number to send to the null Auxiliary Effect Slot.

To disable Auxiliary Send number 1 from Source 'uiSourceID':

```
alSource3i(uiSourceID,AL_AUXILIARY_SEND_FILTER,
AL_EFFECTSLOT_NULL, 1, AL_FILTER_NULL);
```

AL_AIR_ABSORPTION_FACTOR

Version	Property	Range	Default
1.0	AL_AIR_ABSORPTION_FACTOR	0 to 10	0

This Source property is a multiplier on the amount of Air Absorption applied to the Source. The AL_AIR_ABSORPTION_FACTOR is multiplied by an internal Air Absorption Gain HF value of 0.994 (-0.05dB) per meter which represents normal atmospheric humidity and temperature.

By default the value is set to 0 which means that Air Absorption effects are disabled. A value of 1.0 will tell the Effects Extension engine to apply high frequency attenuation on the direct path of the Source at a rate of 0.05dB per meter.

Air Absorption controls the distance-dependent attenuation at high frequencies caused by the propagation medium. It can simulate sound transmission through foggy air, dry air, smoky atmosphere, and so on.

The AL_AIR_ABSORPTION_FACTOR property can be used 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.

AL_ROOM_ROLLOFF_FACTOR

Version	Property	Range	Default
1.0	AL_ROOM_ROLLOFF_FACTOR	0 to 10	0

This Source property is defined the same way as the Reverb Room Rolloff property: it is one of two methods available in the Effect Extension to attenuate the reflected sound (early reflections and reverberation) according to source-listener distance. In this case, however, Room Rolloff applies only to this sound source, and therefore affects only the reflected sound generated by this source.

AL_CONE_OUTER_GAINHF

Version	Property	Range	Default
1.0	AL_CONE_OUTER_GAINHF	0 to 10	0

This Source property enhances the directivity for individual Sources. A directed Source points in a specified direction. The Source sounds at full volume when the listener is directly in front of the source; it is attenuated as the listener circles the Source away from the front.

When OpenAL attenuates a Source's direct-path sound to simulate directivity, it attenuates high-frequency and low-frequency sounds equally. Real world sources tend to be more directive at high frequencies than at low frequencies.

The AL_CONE_OUTER_GAINHF property enhances the directivity effect at your option by attenuating high frequencies more than low frequencies in the rear of the source. At the default setting of 1.0, there is no additional high-frequency attenuation, so OpenAL's directivity effect is unaltered. At the minimum setting of 0.0, directivity attenuation for high frequencies is 100 dB more than it is for low frequencies.

This property sets directivity high-frequency attenuation for both the direct-path and the reflected sounds of the sound source. You can turn off its effect on direct-path sound using the AL_DIRECT_FILTER_GAINHF_AUTO property, or you can turn off its effect on reflected sound using the AL_AUXILIARY_SEND_FILTER_GAINHF_AUTO property.

AL_DIRECT_FILTER_GAINHF_AUTO

Version	Property	Range	Default
1.0	AL_DIRECT_FILTER_GAINHF_AUTO	AL_FALSE or AL_TRUE	AL_TRUE

If this Source property is set to AL_TRUE (its default value), this Source's direct-path is automatically filtered according to the orientation of the source relative to the listener and the setting of the Source property AL_CONE_OUTER_GAINHF. If AL_CONE_OUTER_GAINHF is set to 1.0, the source is not more directive at high frequencies and this property 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 property is set to AL_FALSE, this Source's direct-path sound is not filtered at all according to orientation.

AL_AUXILIARY_SEND_FILTER_GAIN_AUTO

Version	Property	Range	Default
1.0	AL_AUXILIARY_SEND_FILTER_GAIN_AUTO	AL_FALSE or AL_TRUE	AL_TRUE

If this Source property is set to AL_TRUE (its default value), the intensity of this Source's reflected sound is automatically attenuated according to source-listener distance and source directivity (as determined by the cone parameters). If it is AL_FALSE, the reflected sound is not attenuated according to distance and directivity.

AL_AUXILIARY_SEND_FILTER_GAINHF_AUTO

Version	Property	Range	Default
1.0	AL_AUXILIARY_SEND_FILTER_GAINHF_AUTO	AL_FALSE or AL_TRUE	AL_TRUE

If this Source property is AL_TRUE (its default value), the intensity of this Source's reflected sound at high frequencies will be automatically attenuated according to the high-frequency source directivity as set by the AL_CONE_OUTER_GAINHF property. If AL_CONE_OUTER_GAINHF is set to 1.0, the Source is not more directive at high frequencies and this property has no effect. Otherwise, making the Source more directive at high frequencies will have the natural effect of reducing the amount of high frequencies in the reflected sound.

If this property is AL_FALSE, the Source's reflected sound is not filtered at all according to the Source's directivity.

Listener Object Extensions

Properties

The following Listener property is defined:

Version	Property	Range	Default
1.0	AL_METERS_PER_UNIT	> 0	1.0

AL_METERS_PER_UNIT

Version	Property	Range	Default
1.0	AL_METERS_PER_UNIT	> 0	1.0

The Listener property allows an application to specify the relationship between the units passed to OpenAL calls such as positions, velocities, reference distance, etc … and the real-world. This information is critical if Air Absorption effects are enabled because the amount of Air Absorption applied is directly related to the real-world distance between the Source and the Listener. If the application is using centimeters for distance units, then this property should be set to 0.01 so that the amount of air absorption applied is not 100 times too great!

Context Object Extensions

Properties

The following Context properties are defined:

Version	Property	Range	Default
1.0	ALC_EFX_MAJOR_VERSION	N/A	N/A
1.0	ALC_EFX_MINOR_VERSION	N/A	N/A
1.0	ALC_MAX_AUXILIARY_SENDS	N/A	2

ALC_EFX_MAJOR_VERSION

Version	Property	Range	Default
1.0	ALC_EFX_MAJOR_VERSION	N/A	N/A

This Context property can be used by the application to retrieve the Major version number of the Effects Extension supported by this OpenAL implementation. As this is a Context property is should be retrieved using alcGetIntegerv.

ALC_EFX_MINOR_VERSION

Version	Property	Range	Default
1.0	ALC_EFX_MINOR_VERSION	N/A	N/A

This Context property can be used by the application to retrieve the Minor version number of the Effects Extension supported by this OpenAL implementation. As this is a Context property is should be retrieved using alcGetIntegerv.

ALC_MAX_AUXILIARY_SENDS

Version	Property	Range	Default
1.0	ALC_MAX_AUXILIARY_SENDS	N/A	2

This Context property can be passed to OpenAL during Context creation (alcCreateContext) to request a maximum number of Auxiliary Sends desired on each Source. It is not guaranteed that the desired number of sends will be available, so an application should query this property after creating the context using alcGetIntergerv. Tutorial 1 shows how to initialize OpenAL and the Effect Extension including requesting and querying the number of Auxiliary Sends per Source.

Appendix 1 – Effect property descriptions

EAX Reverb

The EAX Reverb effect type is exclusive to selected Creative OpenAL implementations. The EAX Reverb parameter set is a superset of the standard OpenAL Effects Extension environmental reverb effect. Additional parameters allow for:

- Closer control over the tone of the reverb
- Reverb directivity, using panning vectors
- Reverb granularity, using echo controls

The EAX Reverb is natively supported on any devices that support the EAX 3.0 or above standard, including:

- SoundBlaster Audigy series soundcards
- SoundBlaster X-Fi series soundcards

The EAX Reverb will be emulated on devices that only support EAX 2.0. Note: The "Generic Software" device falls into this category as the software mixer supports the EAX 2.0 Reverb effect.

Reverb Density

Specify using this ID	AL_EAXREVERB_DENSITY
Value type	FLOAT
Value range	0.0 to 1.0
Default value	1.0

Reverb Modal Density controls the coloration of the late reverb. Lowering the value adds more coloration to the late reverb.

Reverb Diffusion

Specify using this ID	AL_EAXREVERB_DIFFUSION
Value type	FLOAT
Value range	0.0 to 1.0
Default value	1.0
Value units	A linear multiplier value

The Reverb Diffusion property controls the echo density in the reverberation decay. It's set by default to 1.0, which provides the highest density. Reducing diffusion gives the reverberation a more "grainy" character that is especially noticeable with percussive sound sources. If you set a diffusion value of 0.0, the later reverberation sounds like a succession of distinct echoes.

Reverb Gain, Reverb Gain HF and Reverb Gain LF

Specify using this ID	AL_EAXREVERB_GAIN
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.32
Value units	Linear gain

The Reverb Gain property is the master volume control for the reflected sound (both early reflections and reverberation) that the reverb effect adds to all sound sources. It sets the maximum amount of reflections and reverberation added to the final sound mix. The value of the Reverb Gain property ranges from 1.0 (0db) (the maximum amount) to 0.0 (-100db) (no reflected sound at all).

Specify using this ID	AL_EAXREVERB_GAINHF
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.89
Value units	Linear gain

The Reverb Gain HF property further tweaks reflected sound by attenuating it at high frequencies. It controls a low-pass filter that applies globally to the reflected sound of all sound sources feeding the particular instance of the reverb effect. The value of the Reverb Gain HF property ranges from 1.0 (0db) (no filter) to 0.0 (-100db) (virtually no reflected sound). HF Reference sets the frequency at which the value of this property is measured.

Specify using this ID	AL_EAXREVERB_GAINLF
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.0
Value units	Linear gain

The Reverb Gain LF property further tweaks reflected sound by attenuating it at low frequencies. It controls a high-pass filter that applies globally to the reflected sound of all sound sources feeding the particular instance of the reverb effect. The value of the Reverb Gain LF property ranges from 1.0 (0db) (no filter) to 0.0 (-100db) (virtually no reflected sound). LF Reference sets the frequency at which the value of this property is measured.

Decay Time, Decay HF Ratio and Decay LF Ratio

Specify using this ID	AL_EAXREVERB_DECAY_TIME
Value type	FLOAT
Value range	0.1 to 20.0
Default value	1.49
Value units	Seconds

The Decay Time property sets the reverberation decay time. It ranges from 0.1 (typically a small room with very dead surfaces) to 20.0 (typically a large room with very live surfaces).

Specify using this ID	AL_EAXREVERB_DECAY_HFRATIO
Value type	FLOAT
Value range	0.1 to 20.0
Default value	0.83
Value units	A linear multiplier value

The Decay HF Ratio property adjusts the spectral quality of the Decay Time parameter. It is the ratio of high-frequency decay time relative to the time set by Decay Time. The Decay HF Ratio value 1.0 is neutral: the decay time is equal for all frequencies. As Decay HF Ratio increases above 1.0, the high-frequency decay time increases so it's longer than the decay time at mid frequencies. You hear a more brilliant reverberation with a longer decay at high frequencies. As

the Decay HF Ratio value decreases below 1.0, the high-frequency decay time decreases so it's shorter than the decay time of the mid frequencies. You hear a more natural reverberation.

Specify using this ID	AL_EAXREVERB_DECAY_LFRATIO
Value type	FLOAT
Value range	0.1 to 20.0
Default value	1.0
Value units	A linear multiplier value

The Decay LF Ratio property adjusts the spectral quality of the Decay Time parameter. It is the ratio of low-frequency decay time relative to the time set by Decay Time. The Decay LF Ratio value 1.0 is neutral: the decay time is equal for all frequencies. As Decay LF Ratio increases above 1.0, the low-frequency decay time increases so it's longer than the decay time at mid frequencies. You hear a more booming reverberation with a longer decay at low frequencies. As the Decay LF Ratio value decreases below 1.0, the low-frequency decay time decreases so it's shorter than the decay time of the mid frequencies. You hear a more tinny reverberation.

Reflections Gain and Reflections Delay

Specify using this ID	AL_EAXREVERB_REFLECTIONS_GAIN
Value type	FLOAT
Value range	0.0 to 3.16
Default value	0.05
Value units	Linear gain

The Reflections Gain property controls the overall amount of initial reflections relative to the Gain property. (The Gain property sets the overall amount of reflected sound: both initial reflections and later reverberation.) The value of Reflections Gain ranges from a maximum of 3.16 (+10 dB) to a minimum of 0.0 (-100 dB) (no initial reflections at all), and is corrected by the value of the Gain property. The Reflections Gain property does not affect the subsequent reverberation decay.

You can increase the amount of initial reflections to simulate a more narrow space or closer walls, especially effective if you associate the initial reflections increase with a reduction in reflections delays by lowering the value of the Reflection Delay property. To simulate open or semi-open environments, you can maintain the amount of early reflections while reducing the value of the Late Reverb Gain property, which controls later reflections.

Specify using this ID	AL_EAXREVERB_REFLECTIONS_DELAY
Value type	FLOAT
Value range	0.0 to 0.3
Default value	0.007
Value units	Seconds

The Reflections Delay property is the amount of delay between the arrival time of the direct path from the source to the first reflection from the source. It ranges from 0 to 300 milliseconds. You can reduce or increase Reflections Delay to simulate closer or more distant reflective surfaces—and therefore control the perceived size of the room.

Reflections Pan

Specify using this ID	AL_EAXREVERB_REFLECTIONS_PAN
Value type	VECTOR

Value range	Magnitude between 0 and 1
Default value	[0.0, 0.0, 0.0]

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.

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, 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 a left-handed co-ordinate system, unlike OpenAL 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.

Late Reverb Gain and Late Reverb Delay

Specify using this ID	AL_EAXREVERB_LATE_REVERB_GAIN
Value type	FLOAT
Value range	0.0 to 10.0
Default value	1.26
Value units	Linear gain

The Late Reverb Gain property controls the overall amount of later reverberation relative to the Gain property. (The Gain property sets the overall amount of both initial reflections and later reverberation.) The value of Late Reverb Gain ranges from a maximum of 10.0 (+20 dB) to a minimum of 0.0 (-100 dB) (no late reverberation at all).

Note that Late Reverb Gain and Decay Time are independent properties: If you adjust Decay Time without changing Late Reverb Gain, the total intensity (the averaged square of the amplitude) of the late reverberation remains constant.

Specify using this ID	AL_EAXREVERB_LATE_REVERB_DELAY
Value type	FLOAT
Value range	0.0 to 0.1
Default value	0.011
Value units	Seconds

The Late Reverb Delay property defines the begin time of the late reverberation relative to the time of the initial reflection (the first of the early reflections). It ranges from 0 to 100 milliseconds. Reducing or increasing Late Reverb Delay is useful for simulating a smaller or larger room.

Late Reverb Pan

Specify using this ID	AL_EAXREVERB_LATE_REVERB_PAN
Value type	VECTOR
Value range	Magnitude between 0 and 1
Default value	[0.0, 0.0, 0.0]

The Late Reverb Pan property is a 3D vector that controls the spatial distribution of the late reverb. The direction of this vector controls the global direction of the reverb, while its magnitude controls how focused the reverb are towards this direction. The details under Reflections Pan, above, also apply to Late Reverb Pan.

Echo Time, Echo Depth

Specify using this ID	AL_EAXREVERB_ECHO_TIME
Value type	FLOAT
Value range	0.075 to 0.25
Default value	0.25
Value units	Seconds

Specify using this ID	AL_EAXREVERB_ECHO_DEPTH
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.0
Value units	A linear multiplier value

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 Reverb 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 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.

Modulation Time, Modulation Depth

Specify using this ID	AL_EAXREVERB_MODULATION_TIME
Value type	FLOAT
Value range	0.004 to 4.0
Default value	0.25
Value units	Seconds

Specify using this ID	AL_EAXREVERB_MODULATION_DEPTH
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.0
Value units	A linear multiplier value

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.

HF Reference, LF Reference

Specify using this ID	AL_EAXREVERB_HFREFERENCE
Value type	FLOAT
Value range	1000.0 to 20000.0
Default value	5000.0
Value units	Hz

Specify using this ID	AL_EAXREVERB_LFREFERENCE
Value type	FLOAT
Value range	20.0 to 1000.0
Default value	250.0
Value units	Hz

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 Reverb properties are measured, for example Decay HF Ratio and Decay LF Ratio.

Note that 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.

Room Rolloff Factor

Specify using this ID	AL_EAXREVERB_ROOM_ROLLOFF_FACTOR
Value type	FLOAT
Value range	0.0 to 10.0
Default value	0.0
Value units	A linear multiplier value

The Room Rolloff Factor property is one of two methods available to attenuate the reflected sound (containing both reflections and reverberation) according to source-listener distance. It's defined the same way as OpenAL's Rolloff Factor, but operates on reverb sound instead of direct-path sound. Setting the Room Rolloff Factor value to 1.0 specifies that the reflected sound will decay by 6 dB every time the distance doubles. Any value other than 1.0 is equivalent to a scaling factor applied to the quantity specified by ((Source listener distance) - (Reference Distance)). Reference Distance is an OpenAL source parameter that specifies the inner border for distance rolloff effects: if the source comes closer to the listener than the reference distance, the direct-path sound isn't increased as the source comes closer to the listener, and neither is the reflected sound.

The default value of Room Rolloff Factor is 0.0 because, by default, the Effects Extension reverb effect naturally manages the reflected sound level automatically for each sound source to simulate the natural rolloff of reflected sound vs. distance in typical rooms. (Note that this isn't the case if the source property flag AL_AUXILIARY_SEND_FILTER_GAIN_AUTO is set to AL_FALSE) You can use Room Rolloff Factor as an option to automatic control so you can exaggerate or replace the default automatically-controlled rolloff.

Air Absorption Gain HF

Specify using this ID	AL_EAXREVERB_AIR_ABSORPTION_GAINHF
Value type	FLOAT
Value range	0.892 to 1.0
Default value	0.994
Value units	Linear gain per meter

The Air Absorption Gain HF property controls the distance-dependent attenuation at high frequencies caused by the propagation medium. It applies to reflected sound only. You can use Air Absorption Gain HF to simulate sound transmission through foggy air, dry air, smoky atmosphere, and so on. The default value is 0.994 (-0.05 dB) per meter, which roughly corresponds to typical condition of atmospheric humidity, temperature, and so on. Lowering the value simulates a more absorbent medium (more humidity in the air, for example); raising the value simulates a less absorbent medium (dry desert air, for example).

Decay HF Limit

Specify using this ID	AL_EAXREVERB_DECAYHF_LIMIT
Value type	INTEGER
Value range	AL_FALSE, AL_TRUE
Default value	AL_TRUE

When this flag is set, the high-frequency decay time automatically stays below a limit value that's derived from the setting of the property Air Absorption Gain HF. This limit applies regardless of the setting of the property Decay HF Ratio, and the limit doesn't affect the value of Decay HF Ratio. This limit, when on, maintains a natural sounding reverberation decay by allowing you to increase the value of Decay Time without the risk of getting an unnaturally long decay time at high frequencies. If this flag is set to AL_FALSE, high-frequency decay time isn't automatically limited.

Standard Reverb

AL_EFFECT_REVERB is the standard Effects Extension's environmental reverberation effect.

Reverb Density

Specify using this ID	AL_REVERB_DENSITY
Value type	FLOAT
Value range	0.0 to 1.0
Default value	1.0

Reverb Modal Density controls the coloration of the late reverb. Lowering the value adds more coloration to the late reverb.

Reverb Diffusion

Specify using this ID	AL_REVERB_DIFFUSION
Value type	FLOAT
Value range	0.0 to 1.0
Default value	1.0
Value units	A linear multiplier value

The Reverb Diffusion property controls the echo density in the reverberation decay. It's set by default to 1.0, which provides the highest density. Reducing diffusion gives the reverberation a more "grainy" character that is especially noticeable with percussive sound sources. If you set a diffusion value of 0.0, the later reverberation sounds like a succession of distinct echoes.

Reverb Gain and Reverb Gain HF

Specify using this ID	AL_REVERB_GAIN
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.32
Value units	Linear gain

The Reverb Gain property is the master volume control for the reflected sound (both early reflections and reverberation) that the reverb effect adds to all sound sources. It sets the maximum amount of reflections and reverberation added to the final sound mix. The value of the Reverb Gain property ranges from 1.0 (0db) (the maximum amount) to 0.0 (-100db) (no reflected sound at all).

Specify using this ID	AL_REVERB_GAINHF
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.89
Value units	Linear gain

The Reverb Gain HF property further tweaks reflected sound by attenuating it at high frequencies. It controls a low-pass filter that applies globally to the reflected sound of all sound sources feeding the particular instance of the reverb effect. The value of the Reverb Gain HF property ranges from 1.0 (0db) (no filter) to 0.0 (-100db) (virtually no reflected sound).

Decay Time and Decay HF Ratio

Specify using this ID	AL_REVERB_DECAY_TIME
Value type	FLOAT
Value range	0.1 to 20.0
Default value	1.49
Value units	Seconds

The Decay Time property sets the reverberation decay time. It ranges from 0.1 (typically a small room with very dead surfaces) to 20.0 (typically a large room with very live surfaces).

Specify using this ID	AL_REVERB_DECAY_HFRATIO
Value type	FLOAT
Value range	0.1 to 2.0
Default value	0.83
Value units	A linear multiplier value

The Decay HF Ratio property sets the spectral quality of the Decay Time parameter. It is the ratio of high-frequency decay time relative to the time set by Decay Time. The Decay HF Ratio value 1.0 is neutral: the decay time is equal for all frequencies. As Decay HF Ratio increases above 1.0, the high-frequency decay time increases so it's longer than the decay time at low frequencies. You hear a more brilliant reverberation with a longer decay at high frequencies. As

the Decay HF Ratio value decreases below 1.0, the high-frequency decay time decreases so it's shorter than the decay time of the low frequencies. You hear a more natural reverberation.

Reflections Gain and Reflections Delay

Specify using this ID	AL_REVERB_REFLECTIONS_GAIN
Value type	FLOAT
Value range	0.0 to 3.16
Default value	0.05
Value units	Linear gain

The Reflections Gain property controls the overall amount of initial reflections relative to the Gain property. (The Gain property sets the overall amount of reflected sound: both initial reflections and later reverberation.) The value of Reflections Gain ranges from a maximum of 3.16 (+10 dB) to a minimum of 0.0 (-100 dB) (no initial reflections at all), and is corrected by the value of the Gain property. The Reflections Gain property does not affect the subsequent reverberation decay.

You can increase the amount of initial reflections to simulate a more narrow space or closer walls, especially effective if you associate the initial reflections increase with a reduction in reflections delays by lowering the value of the Reflection Delay property. To simulate open or semi-open environments, you can maintain the amount of early reflections while reducing the value of the Late Reverb Gain property, which controls later reflections.

Specify using this ID	AL_REVERB_REFLECTIONS_DELAY
Value type	FLOAT
Value range	0.0 to 0.3
Default value	0.007
Value units	Seconds

The Reflections Delay property is the amount of delay between the arrival time of the direct path from the source to the first reflection from the source. It ranges from 0 to 300 milliseconds. You can reduce or increase Reflections Delay to simulate closer or more distant reflective surfaces—and therefore control the perceived size of the room.

Late Reverb Gain and Late Reverb Delay

Specify using this ID	AL_REVERB_LATE_REVERB_GAIN
Value type	FLOAT
Value range	0.0 to 10.0
Default value	1.26
Value units	Linear gain

The Late Reverb Gain property controls the overall amount of later reverberation relative to the Gain property. (The Gain property sets the overall amount of both initial reflections and later reverberation.) The value of Late Reverb Gain ranges from a maximum of 10.0 (+20 dB) to a minimum of 0.0 (-100 dB) (no late reverberation at all).

Note that Late Reverb Gain and Decay Time are independent properties: If you adjust Decay Time without changing Late Reverb Gain, the total intensity (the averaged square of the amplitude) of the late reverberation remains constant.

Specify using this ID	AL_REVERB_LATE_REVERB_DELAY

Value type	FLOAT
Value range	0.0 to 0.1
Default value	0.011
Value units	Seconds

The Late Reverb Delay property defines the begin time of the late reverberation relative to the time of the initial reflection (the first of the early reflections). It ranges from 0 to 100 milliseconds. Reducing or increasing Late Reverb Delay is useful for simulating a smaller or larger room.

Room Rolloff Factor

Specify using this ID	AL_REVERB_ROOM_ROLLOFF_FACTOR
Value type	FLOAT
Value range	0.0 to 10.0
Default value	0.0
Value units	A linear multiplier value

The Room Rolloff Factor property is one of two methods available to attenuate the reflected sound (containing both reflections and reverberation) according to source-listener distance. It's defined the same way as OpenAL's Rolloff Factor, but operates on reverb sound instead of direct-path sound. Setting the Room Rolloff Factor value to 1.0 specifies that the reflected sound will decay by 6 dB every time the distance doubles. Any value other than 1.0 is equivalent to a scaling factor applied to the quantity specified by ((Source listener distance) - (Reference Distance)). Reference Distance is an OpenAL source parameter that specifies the inner border for distance rolloff effects: if the source comes closer to the listener than the reference distance, the direct-path sound isn't increased as the source comes closer to the listener, and neither is the reflected sound.

The default value of Room Rolloff Factor is 0.0 because, by default, the Effects Extension reverb effect naturally manages the reflected sound level automatically for each sound source to simulate the natural rolloff of reflected sound vs. distance in typical rooms. (Note that this isn't the case if the source property flag AL_AUXILIARY_SEND_FILTER_GAIN_AUTO is set to AL_FALSE) You can use Room Rolloff Factor as an option to automatic control so you can exaggerate or replace the default automatically-controlled rolloff.

Air Absorption Gain HF

Specify using this ID	AL_REVERB_AIR_ABSORPTION_GAINHF
Value type	FLOAT
Value range	0.892 to 1.0
Default value	0.994
Value units	Linear gain per meter

The Air Absorption Gain HF property controls the distance-dependent attenuation at high frequencies caused by the propagation medium. It applies to reflected sound only. You can use Air Absorption Gain HF to simulate sound transmission through foggy air, dry air, smoky atmosphere, and so on. The default value is 0.994 (-0.05 dB) per meter, which roughly corresponds to typical condition of atmospheric humidity, temperature, and so on. Lowering the value simulates a more absorbent medium (more humidity in the air, for example); raising the value simulates a less absorbent medium (dry desert air, for example).

Decay HF Limit

Specify using this ID	AL_REVERB_DECAY_HFLIMIT
Value type	INTEGER
Value range	AL_FALSE, AL_TRUE
Default value	AL_TRUE

When this flag is set, the high-frequency decay time automatically stays below a limit value that's derived from the setting of the property Air Absorption HF. This limit applies regardless of the setting of the property Decay HF Ratio, and the limit doesn't affect the value of Decay HF Ratio. This limit, when on, maintains a natural sounding reverberation decay by allowing you to increase the value of Decay Time without the risk of getting an unnaturally long decay time at high frequencies. If this flag is set to AL_FALSE, high-frequency decay time isn't automatically limited.

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.

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

Specify using this ID	AL_CHORUS_WAVEFORM
Value type	INTEGER
Value range	0 (sin), 1 (triangle)
Default value	1

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

Chorus Phase

Specify using this ID	AL_CHORUS_PHASE
Value type	INTEGER
Value range	-180 to 180
Default value	90

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

Specify using this ID	AL_CHORUS_RATE
Value type	FLOAT
Value range	0.0 to 10.0

Default value	1.1

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

Chorus Depth

Specify using this ID	AL_CHORUS_DEPTH
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.1

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

Chorus Feedback

Specify using this ID	AL_CHORUS_FEEDBACK
Value type	FLOAT
Value range	-1.0 to 1.0
Default value	0.25

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

Specify using this ID	AL_CHORUS_DELAY
Value type	FLOAT
Value range	0.0 to 0.016
Default value	0.016

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.

The OpenAL Effects Extension 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

Specify using this ID	AL_DISTORTION_EDGE
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.2

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

Distortion Gain

Specify using this ID	AL_DISTORTION_GAIN
Value type	FLOAT
Value range	0.01 to 1.0
Default value	0.05

This property allows you to attenuate the distorted sound.

Distortion Low Pass Cutoff

Specify using this ID	AL_DISTORTION_LOWPASS_CUTOFF
Value type	FLOAT
Value range	80.0 to 24000.0
Default value	8000.0

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

Distortion EQ Center

Specify using this ID	AL_DISTORTION_EQCENTER
Value type	FLOAT
Value range	80.0 to 24000.0
Default value	3600.0

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

Distortion EQ Bandwidth

Specify using this ID	AL_DISTORTION_EQBANDWIDTH
Value type	FLOAT
Value range	80.0 to 24000.0
Default value	3600.0

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

Specify using this ID	AL_ECHO_DELAY
Value type	FLOAT
Value range	0.0 to 0.207
Default value	0.1

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

Specify using this ID	AL_ECHO_LRDELAY
Value type	FLOAT
Value range	0.0 to 0.404
Default value	0.1

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

Specify using this ID	AL_ECHO_DAMPING
Value type	FLOAT
Value range	0.0 to 0.99
Default value	0.5

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

Specify using this ID	AL_ECHO_FEEDBACK
Value type	FLOAT
Value range	0.0 to 1.0
Default value	0.5

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

Specify using this ID	AL_ECHO_SPREAD
Value type	FLOAT

Value range	-1.0 to 1.0
Default value	-1.0

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.

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

Specify using this ID	AL_FLANGER_WAVEFORM
Value type	INTEGER
Value range	0 (sin), 1 (triangle)
Default value	1

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

Specify using this ID	AL_FLANGER_PHASE
Value type	INTEGER
Value range	-180 to 180
Default value	0

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

Flanger Rate

Specify using this ID	AL_FLANGER_RATE
Value type	FLOAT
Value range	0.0 to 10.0
Default value	0.27

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

Flanger Depth

Specify using this ID	AL_FLANGER_DEPTH
Value type	FLOAT
Value range	0.0 to 1.0
Default value	1.0

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

Flanger Feedback

Specify using this ID	AL_FLANGER_FEEDBACK
Value type	FLOAT
Value range	-1.0 to 1.0
Default value	-0.5

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

Specify using this ID	AL_FLANGER_DELAY
Value type	FLOAT
Value range	0.0 to 0.004
Default value	0.002

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

Specify using this ID	AL_FREQUENCY_SHIFTER_FREQUENCY
Value type	FLOAT
Value range	0.0 to 24000.0
Default value	0.0

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 Left Direction

Specify using this ID	AL_FREQUENCY_SHIFTER_LEFT_DIRECTION
Value type	INTEGER
Value range	0 (down), 1 (up), 2 (off)
Default value	0

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

Frequency Shifter Right Direction

Specify using this ID	AL_FREQUENCY_SHIFTER_RIGHT_DIRECTION
Value type	INTEGER
Value range	0 (down), 1 (up), 2 (off)
Default value	0

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

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.

These are the available phoneme (formant filter settings) types:-

ID	Value	Enumerated
0	“A”	AL_VOCAL_MORPHER_PHONEME_A
1	“E”	AL_VOCAL_MORPHER_PHONEME_E
2	“I”	AL_VOCAL_MORPHER_PHONEME_I
3	“O”	AL_VOCAL_MORPHER_PHONEME_O
4	“U”	AL_VOCAL_MORPHER_PHONEME_U
5	“AA”	AL_VOCAL_MORPHER_PHONEME_AA
6	“AE”	AL_VOCAL_MORPHER_PHONEME_AE
7	“AH”	AL_VOCAL_MORPHER_PHONEME_AH
8	“AO”	AL_VOCAL_MORPHER_PHONEME_AO
9	“EH”	AL_VOCAL_MORPHER_PHONEME_EH
10	“ER”	AL_VOCAL_MORPHER_PHONEME_ER
11	“IH”	AL_VOCAL_MORPHER_PHONEME_IH
12	“IY”	AL_VOCAL_MORPHER_PHONEME_IY
13	“UH”	AL_VOCAL_MORPHER_PHONEME_UH
14	“UW”	AL_VOCAL_MORPHER_PHONEME_UW
15	“B”	AL_VOCAL_MORPHER_PHONEME_B
16	“D”	AL_VOCAL_MORPHER_PHONEME_D
17	“F”	AL_VOCAL_MORPHER_PHONEME_F
18	“G”	AL_VOCAL_MORPHER_PHONEME_G
19	“J”	AL_VOCAL_MORPHER_PHONEME_J
20	“K”	AL_VOCAL_MORPHER_PHONEME_K
21	“L”	AL_VOCAL_MORPHER_PHONEME_L
22	“M”	AL_VOCAL_MORPHER_PHONEME_M
23	“N”	AL_VOCAL_MORPHER_PHONEME_N
24	“P”	AL_VOCAL_MORPHER_PHONEME_P
25	“R”	AL_VOCAL_MORPHER_PHONEME_R
26	“S”	AL_VOCAL_MORPHER_PHONEME_S
27	“T”	AL_VOCAL_MORPHER_PHONEME_T
28	“V”	AL_VOCAL_MORPHER_PHONEME_V
29	“Z”	AL_VOCAL_MORPHER_PHONEME_Z

Vocal Morpher Phoneme A and Vocal Morpher Phoneme B

Specify using this ID	AL_VOCAL_MORPHER_PHONEMEA
Value type	INTEGER
Value range	0 to 29
Default value	0 (“A”)

Specify using this ID	AL_VOCAL_MORPHER_PHONEMEB
Value type	INTEGER
Value range	0 to 29
Default value	10 (“ER”)

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 AL_VOCAL_MORPHER_RATE.

Vocal Morpher Phoneme A and Vocal Morpher Phoneme B coarse tuning

Specify using this ID	AL_VOCAL_MORPHER_PHONEMEA_COARSE_TUNING
Value type	INTEGER
Value range	-24 to 24
Default value	0
Specify using this ID	AL_VOCAL_MORPHER_PHONEMEB_COARSE_TUNING
Value type	INTEGER
Value range	-24 to 24
Default value	0

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

Vocal Morpher Waveform

Specify using this ID	AL_VOCAL_MORPHER_WAVEFORM
Value type	INTEGER
Value range	0 (sin), 1 (triangle), 2 (saw)
Default value	0

This controls the shape of the low-frequency oscillator used to morph between the two phoneme filters. By selecting a saw tooth wave and a slow AL_VOCAL_MORPHER_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).

Vocal Morpher Rate

Specify using this ID	AL_VOCAL_MORPHER_RATE
Value type	FLOAT
Value range	0.0 to 10.0
Default value	1.41

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

Pitch Shifter

The pitch shifter applies time-invariant pitch shifting to the input signal, over a one octave range and controllable at a semi-tone and cent resolution.

Pitch Shifter Coarse Tune

Specify using this ID	AL_PITCH_SHIFTER_COARSE_TUNE
Value type	INTEGER
Value range	-12 to 12
Default value	12

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

Specify using this ID	AL_PITCH_SHIFTER_FINE_TUNE
Value type	INTEGER
Value range	-50 to 50
Default value	0

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

Specify using this ID	AL_RING_MODULATOR_FREQUENCY
Value type	FLOAT
Value range	0.0 to 8000.0
Default value	440.0

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

Specify using this ID	AL_RING_MODULATOR_HIGHPASS_CUTOFF
Value type	FLOAT
Value range	0.0 to 24000.0
Default value	800.0

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

Specify using this ID	AL_RING_MODULATOR_WAVEFORM
Value type	INTEGER
Value range	0 (sin), 1 (saw), 2 (square)
Default value	0

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.

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. This OpenAL Effects Extension effect is called *Auto*-wah because there is no user input for modulating the cut-off point. Instead the effect is achieved by analysing the input signal, and applying a band-pass filter according the intensity of the incoming audio.

Auto-Wah Attack Time

Specify using this ID	AL_AUTOWAH_ATTACK_TIME
Value type	FLOAT
Value range	0.0001 to 1.0
Default value	0.06

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

Specify using this ID	AL_AUTOWAH_RELEASE_TIME
Value type	FLOAT
Value range	0.0001 to 1.0
Default value	0.06

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

Specify using this ID	AL_AUTOWAH_RESONANCE
Value type	FLOAT
Value range	2.0 to 1000.0
Default value	1000.0

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 Gain

Specify using this ID	AL_AUTOWAH_PEAK_GAIN
Value type	FLOAT
Value range	0.00003 to 31621.0
Default value	11.22

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

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.
The Compressor effect cannot be tweaked in depth – it can just be switched on and off.

Compressor

Specify using this ID	AL_COMPRESSOR_ONOFF
Value type	INTEGER
Value range	0 (off), 1(on)
Default value	1

The OpenAL Effect Extension Compressor can only be switched on and off – it cannot be adjusted.

Equalizer

The OpenAL Effects Extension 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

Specify using this ID	AL_EQUALIZER_LOW_GAIN
Value type	FLOAT
Value range	0.126 to 7.943
Default value	1.0

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

Equalizer Low Cutoff

Specify using this ID	AL_EQUALIZER_LOW_CUTOFF
Value type	FLOAT
Value range	50.0 to 800.0
Default value	200.0

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

Equalizer Mid 1 Gain

Specify using this ID	AL_EQUALIZER_MID1_GAIN
Value type	FLOAT
Value range	0.126 to 7.943
Default value	1.0

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

Equalizer Mid 1 Center

Specify using this ID	AL_EQUALIZER_MID1_CENTER
Value type	FLOAT
Value range	200.0 to 3000.0
Default value	500.0

This property sets the center frequency for the “mid1” range.

Equalizer Mid 1 Width

Specify using this ID	AL_EQUALIZER_MID1_WIDTH
Value type	FLOAT
Value range	0.01 to 1.0
Default value	1.0

This property controls the width of the “mid1” range.

Equalizer Mid 2 Gain

Specify using this ID	AL_EQUALIZER_MID2_GAIN
Value type	FLOAT
Value range	0.126 to 7.943
Default value	1.0

This property allows you to cut / boost signal on the “mid2” range.

Equalizer Mid 2 Center

Specify using this ID	AL_EQUALIZER_MID2_CENTER
Value type	FLOAT
Value range	1000.0 to 8000.0
Default value	3000.0

This property sets the center frequency for the “mid2” range.

Equalizer Mid 2 Width

Specify using this ID	AL_EQUALIZER_MID2_WIDTH
Value type	FLOAT
Value range	0.01 to 1.0
Default value	1.0

This property controls the width of the “mid2” range.

Equalizer High Gain

Specify using this ID	AL_EQUALIZER_HIGH_GAIN
Value type	FLOAT
Value range	0.126 to 7.943
Default value	1.0

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

Equalizer High Cutoff

Specify using this ID	AL_EQUALIZER_HIGH_CUTOFF
Value type	FLOAT
Value range	4000.0 to 16000.0
Default value	6000.0

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

Appendix 2 - Designing Environmental effects for interactive applications

Introduction

This section of the documentation suggests how designers of interactive 3D audio applications can use DSP effects in their projects.

As described in the Environmental Audio Introduction section, reverberation is the principle audio effect employed to enhance an interactive audio application such as a game. Here, you will find a description of the acoustic model that lies behind the effects extension's environmental reverb effects.

The Environmental Audio Introduction section also described scenarios where audio sources are obstructed, occluded or excluded. In the Designing Environmental Filtering effects section you will find an explanation of how using the effects extension, you actually use low-pass filters applied to sound sources and to their reverb sends in order to simulate these scenarios.

The details here are intended to give the sound designer or programmer the background to select, customise and deploy environmental effects like reverb, so you can give your interactive project the exact same dynamic and lifelike soundtrack you've imagined in your head!

Designing and using Environmental Reverb effects

On today's advanced 3D graphics renderers, 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 images.

OpenAL's Effect Extension SDK includes a fully controllable environmental reverb effect, giving 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 Effects Extension reverberation model is derived from Creative Labs' *EAX* standards.

There are two reverb effect types in this SDK. The reverb effect enumerated as "AL_EFFECT_REVERB" is supported on all known OpenAL Effect Extension implementations, including Creative's PC software renderer.

The reverb "AL_EFFECT_EAXREVERB" has a more advanced parameter set. Not all OpenAL implementations support this effect type. It is however guaranteed to be available on native implementations for Creative's Audigy and X-Fi series PCI sound cards (providing the user has the latest driver update installed.) The parameter set for the EAX Reverb is a superset of that of the standard OpenAL reverb.

The table below shows the reverb controls available in the EAX Reverb, plus where they exist the corresponding properties exposed in the general OpenAL Reverb.:

Reverb Effect Parameters

OpenAL EAXReverb Parameter Name	OpenAL standard reverb equivalent
AL_EAXREVERB_DENSITY	AL_REVERB_DENSITY
AL_EAXREVERB_DIFFUSION	AL_REVERB_DIFFUSION
AL_EAXREVERB_GAIN	AL_REVERB_GAIN
AL_EAXREVERB_GAINHF	AL_REVERB_GAINHF
AL_EAXREVERB_GAINLF	N/A
AL_EAXREVERB_DECAY_TIME	AL_REVERB_DECAY_TIME
AL_EAXREVERB_DECAY_HFRATIO	AL_REVERB_DECAY_HFRATIO
AL_EAXREVERB_DECAY_LFRATIO	N/A
AL_EAXREVERB_REFLECTIONS_GAIN	AL_EAXREVERB_REFLECTIONS_GAIN
AL_EAXREVERB_REFLECTIONS_DELAY	AL_EAXREVERB_REFLECTIONS_DELAY
AL_EAXREVERB_REFLECTIONS_PAN	N/A
AL_EAXREVERB_LATE_REVERB_GAIN	AL_EAXREVERB_LATE_REVERB_GAIN
AL_EAXREVERB_LATE_REVERB_DELAY	AL_EAXREVERB_LATE_REVERB_DELAY
AL_EAXREVERB_LATE_REVERB_PAN	N/A
AL_EAXREVERB_ECHO_TIME	N/A
AL_EAXREVERB_ECHO_DEPTH	N/A
AL_EAXREVERB_MODULATION_TIME	N/A
AL_EAXREVERB_MODULATION_DEPTH	N/A
AL_EAXREVERB_AIR_ABSORPTION_GAINHF	AL_EAXREVERB_AIR_ABSORPTION_GAINHF
AL_EAXREVERB_HFREFERENCE	N/A
AL_EAXREVERB_LFREFERENCE	N/A
AL_EAXREVERB_ROOM_ROLLOFF_FACTOR	AL_EAXREVERB_ROOM_ROLLOFF_FACTOR
AL_EAXREVERB_DECAY_HFLIMIT	AL_EAXREVERB_DECAY_HFLIMIT

The reverb 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 (available in EAX Reverb only)
- The “Room filter,” which affects the Reflections and the Reverb identically and allows correcting their energy at high frequencies (both reverb types) and low frequencies (EAX Reverb only)
- The Decay Time at mid and high frequencies (both reverb types) and low frequencies (EAX Reverb only)
- The reference low and high frequencies (EAX Reverb only)
- The Diffusion of the reverberation
- The Echo and Modulation parameters, which affect the late reverberation. (EAX Reverb only)

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.”

A note on distance balancing

Environmental Reverberation is not the only effect available in OpenAL's Effects Extension system. However the reverb effect is unique in one particular respect, which is this; the reverb algorithm can actually influence the effect send amount for each source. This is important because, as you will see later, the perceived level of reverb on a 3D sound source can vary depending on the distance between source and listener. The default settings provide a realistic effect, but you can adjust the balance if necessary, or even turn off the automatic setting and control reverb level for yourself.

Approaches to designing Reverb Effects

There are a couple of different approaches that you can take to crafting reverb effects for an application's audio implementation.

Select an environmental preset from the SDK

The OpenAL SDK includes a set of reverb presets, designed by Creative's audio specialists. These ready-made acoustic models are presented as #define macros in the efx-util.h file. 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 presets are broadly categorised 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.

So, 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.

Build an effect from scratch

Any enthusiastic audio designer will be excited by the prospect of being able to control the powerful Effects Extension reverberation model 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 Modelling

The way that reverberation sounds in an environment is broadly determined by the size of the room 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. The OpenAL Effects Extension renderer takes into account the distance between the listener and a source when it automatically sets the reverb level for each source.

The statistical approach to environmental modelling means that you don't 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 Modelling*) 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.

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.

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 metres (1129 feet) per second, which is approximately equivalent to one foot per millisecond, or one metre every three milliseconds. When calculating factors such as Reflections Delay and Reverb Delay, you should bear this in mind.

Imagine a square room, with walls twenty feet apart. The listener is located in the centre 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.

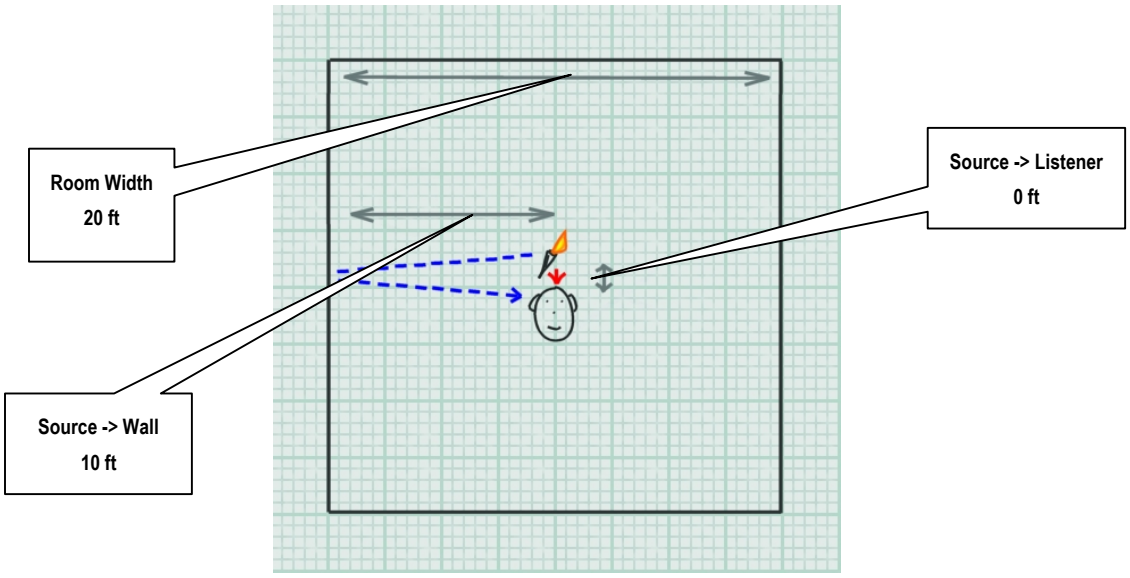


Figure 12 – Worst case: reflections delay approximately 20 milliseconds

Of course, this is only true when sound source and listener are positioned in the centre 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 travelled 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, an 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 the reverb model does not support localisation of reflections on a 'per-source' basis, we must make a generalised 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 an application originate at, or near to, the listener's position.

Also, it is likely that the position of a moving listener will, over time, be distributed fairly evenly between the centre 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. But if you are designing a static environment for a given room, you'll 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 13 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.

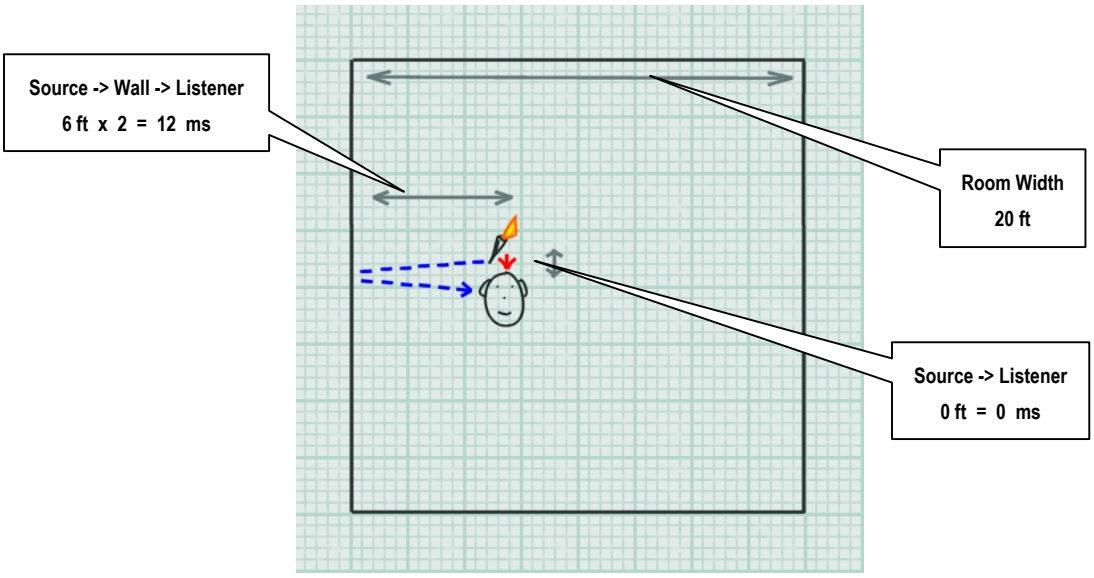


Figure 13 - A typical scenario for calculating Reflection Delay - in this case around 6 milliseconds

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 environmental reverb 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 summarise, 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 considered 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 panelling. A room that consists of reflective walls will generally support louder reflections and longer reverberation than a room consisting of absorptive surfaces. Let's look in more detail at how surface material can influence reflections and reverberation.

Gain

Value range 0.0 to 1.0
Default value 0.32
Value units Linear gain

Gain controls the amount of reflected sound in an environment. Changes made to Gain adjust the level of both the early reflections and the reverberation at the same time. Setting this parameter to 0.0 mutes all reflections and reverberant sound in an environment.

Setting the Gain 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 *Gain* property each time a reverb is applied. However, for independently tweaking and balancing the levels of reflected and reverberant sound within an environment, *Reflections Gain* and *Reverb Gain* should be used.

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. But 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; this means that 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 Gain* 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.

Reflections Gain

Value range 0.0 to 3.16
Default value 0.05
Value units Linear gain

Reflections Gain sets 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 level does vary slightly for each source according to its distance from the listener and to the room's Decay Time. The OpenAL Reverb specifies that the renderer will continuously calculate these changes for itself automatically, so the property Reverb Gain does not require further dynamic modification. For more details on how attenuation is calculated and deployed for reflected sound, see the section on Additional source specific enhancements.

Reverb Gain

Value range 0.0 to 10.0
Default value 1.26
Value units Linear gain

Reverb Gain 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.

Now let's 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 also at high frequencies by the air. So the more often sound energy collides with absorptive surfaces, the quicker the reverb will decay.

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.

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 Gain*) varies with the listener's proximity to surfaces. Reverberation intensity (*Reverb Gain*) and reverberation decay (*Decay Time*) remain more or less constant within an environment.

Surface Reflectivity at different frequencies

Up until now, we have made the assumption that a surface will absorb and reflect sound at a constant level, regardless of the sound's frequency. In actual 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.

Gain HF

Value range 0.0 to 1.0
Default value 0.89
Value units Linear gain

Gain 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, reduce *Gain HF* until you get the timbre you are looking for. *HF Reference* sets the frequency at which the value of this property is measured.

Gain LF

EAX Reverb Only
Value range 0.0 to 1.0
Default value 1.0
Value units Linear gain

Gain LF is the low frequency counterpart to *Gain 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

EAX Reverb Only
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.

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 equaliser or two swept EQ channels on a mixing console.

HF Reference

EAX Reverb Only
Value range 1000.0 to 20000.0
Default value 5000.0
Value units Hertz

LF Reference

EAX Reverb Only
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 Reverb 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.

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 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 *Gain HF*, and maybe also 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 *Gain LF* and/or *Decay LF Ratio*.

- Or if you want a darker, more damped room, get rid of high frequencies with lower settings for *Gain 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's important to assess how the effects work in relation to the source sounds it will process, and to pay heed to the spectral content of the sound samples when designing effects.

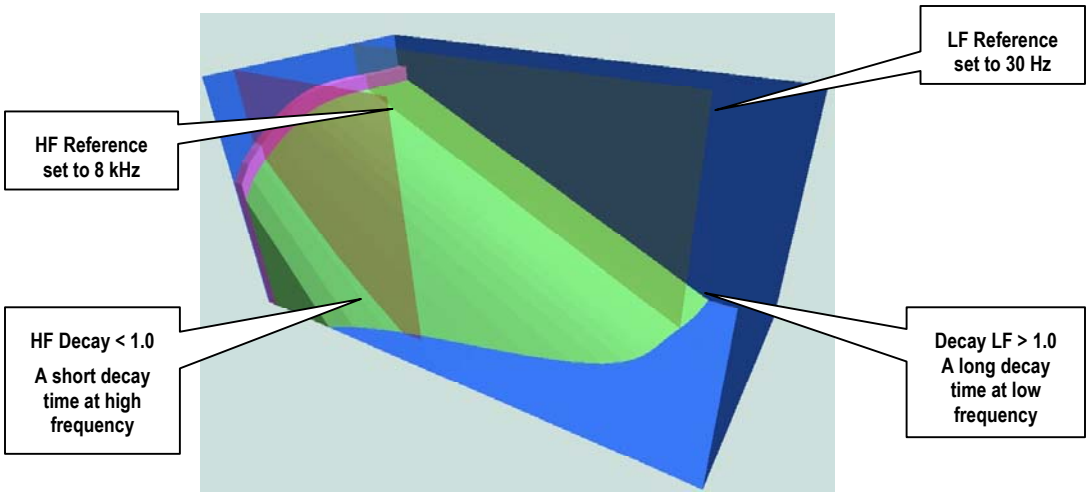


Figure 14 – Three dimensional reverberation response model from Creative's EAGLE reverb design tool. Frequency is represented in the depth dimension – high (near) to low (far). Note how the reverb decay is longer at lower frequencies in this reverb model.

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 Reverb allows for tonal control over the entire reflected sound (*Gain HF, Gain 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). The standard reverb effect is not so adjustable. You only have controls for damping (i.e. attenuating at high frequency), and you cannot change the reference frequency at which the damping is applied. 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. 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. The 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.

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.)

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 for the wave front to traverse the room. With our

Echo Depth

EAX Reverb Only
Value range 0.0 to 1.0
Default value 0.0
Value units A linear multiplier value

Echo Time

EAX Reverb Only
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.

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. And 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. But 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.

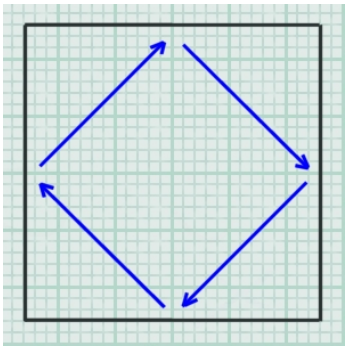


Figure 15 - Cyclic echoes in a room

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

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.

Reverb Density

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

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. Reverb Density controls the coloration of the late reverb. Lowering the value adds more coloration to the late reverb.

Static Dependencies

Table 3, 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 reverb parameters to design a static acoustic simulation for a room.

Reverb Parameter	Room Characteristic
Reflection Delay	Room Size (Implying path length)
Reverb Delay	Room Size (Implying path length)
Reflections Gain	Room Size (Implying path length), Room Absorptivity

Reverb Gain	Room Size (Implying surface area), Room Absorptivity
Decay Time	Room Size, Room Absorptivity
HF Reference LF Reference	Absorptivity of surfaces at different frequencies
Gain HF Gain 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 3 - Showing room characteristic to reverb property dependencies for designing environmental reverbs.

Dynamic Modelling

The physical modelling paradigm has the potential to offer a high quality dynamic acoustic model, particularly in respect of early reflections. However physical modelling 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 also manipulate the early reflections properties according to the precise relationship between the listener and the walls for added realism.

Localising Reflections and Reverb

Making a multi-environment audio engine 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).

As mentioned in the Panning Environments section, it is vital that the reflected sound from environments other than the listener's is correctly localised when implementing multiple environments. 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 Gain* and *Reverb Gain* can be used to further attenuate reflected sound from distant environments.

Reverb Pan

EAX Reverb Only
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.

Reflections Pan

EAX Reverb Only
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, assuming a four-point loudspeaker playback system, setting *Reflections Pan* to (0.0, 0.0, 0.7) means that the reflections are panned to the front speaker pair, whereas as setting of (0.0, 0.0, –0.7) pans the reflections towards the rear speakers. For legacy reasons these vectors follow a left-handed co-ordinate system – *you will need to make allowances for the fact that OpenAL 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.

Dynamic Dependencies

Table 4shows the reverb parameters that are most suitable for real-time adjustment according to listener position, and shows the variables their value would depend on.

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

Table 4 - Showing room characteristic to reverb property dependencies for dynamically updating parameters

Additional Properties

Pitch modulation effects

As well as providing amplitude modulation effects (in the form of a tune-able repeating echo), The EAX Reverb 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.

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.

Modulation Depth

EAX Reverb Only
Value range 0.0 to 1.0
Default value 0.0
Value units A linear multiplier value

Modulation Time

EAX Reverb Only
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.

Distance Controls

The remaining Reverb properties do not adjust the reverberation characteristics. They are concerned with controlling how the reverb level on sources varies according to distance.

The phenomenon of rolloff on audio is simple – 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 acoutics 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).

Under normal circumstances you don't need to worry about this effect, as the reverb effects automatically calculate 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, you can make adjustments for yourself with Room Rolloff Factor.

The default value of 0.0 means that no additional attenuation is added to each source's reflected sound. A value of 1.0 will add additional attenuation at the same rate as the direct sound attenuates with respect to distance.

To calculate the statistical reverb rolloff and the air absorption effect, the reverb engine makes the assumption that the distance unit in use is the meter. If this is not the case, it is important to set accordingly the OpenAL listener extension AL_METERS_PER_UNIT. This will ensure that your unit is correctly converted into metres by the OpenAL reverb engine.

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 reverb engine calculates the roll-off on the Room path (reflected sound, including reflections and reverb), taking into account such properties as *Decay Time.*

Additional attenuation on each source's reflections can be added by increasing this factor. A factor of 1.0 will add attenuation at the same rate as attenuation is added to the source's direct sound as the distance between the source and the listener increases.

The Effect Extension filters enable the simulation of a more subtle rolloff effect, that of air absorption. In normal conditions, our atmosphere absorbs sound slightly more at high frequencies. This is why distant sounds often sound duller. The source object extension property AL_AIR_ABSORPTION_FACTOR controls how much air absorption is applied to each source's dry path. But it is also possible to add extra high frequency attenuation to the reflected sound using the reverb's Air Absorption HF property.

Air Absorption Gain HF

Value range 0.892 to 1.0
Default value 0.994
Value units Linear gain per meter

The reverb property *Air Absorption Gain 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. This property only applies high frequency attenuation to reflected sound. To adjust air absorption filtering on individual 3D sources' dry paths, use the source property AL_AIR_ABSORPTION_FACTOR.

Designing and Using Environmental Filtering effects

As mentioned earlier, Environmental filtering is a fundamental part of the positional audio experience. 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!
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.

Previous environmental audio systems such as Creative's EAX API defined three different filtering scenarios: Obstruction, Occlusion and Exclusion. These were in actual fact simply 'high-level' abstractions, which managed lower level properties controlling attenuation and filtering on sources' direct and processed paths.

The specific combinations of attenuation and filtering on direct and reflected sound is shown in the following table:

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

Table 5 - Filtering and attenuation of direct and reflected sound paths for different sound muffling scenarios

The old 'high-level' controls are not present in the OpenAL Effects Extension. But the filter objects which can be attached to sources give the developer close control over the amount of filtering and attenuation that takes place on every 3D sound. So it is entirely possible to simulate all kinds of muffling scenarios by directly manipulating the following:-

Sound element to control	**Control technique**
Attenuation on Direct path	Attach direct filter, adjust AL_LOWPASS_GAIN
Attenuation on Reverb path	Attach aux send filter, adjust AL_LOWPASS_GAIN
Filtering on Direct path	Attach direct filter, adjust AL_LOWPASS_GAINHF
Filtering on Reverb path	Attach aux send filter, adjust AL_LOWPASS_GAINHF

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.

Obstruction

Consider the *Obstruction* scenario mentioned in the Environmental Filtering section earlier:

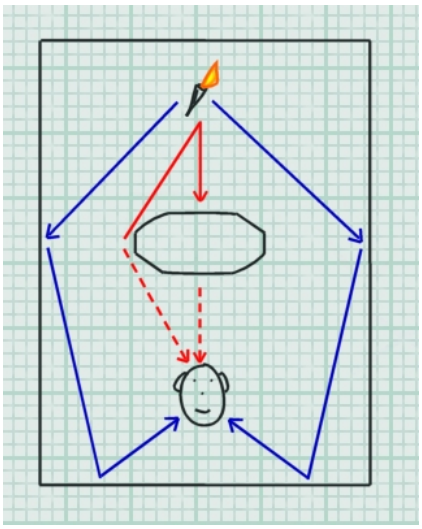


Figure 16 - Obstruction

In this scenario the direct path between the sound source and the listener (shown in red) is broken. The effect of diffraction and absorption mean that any dry sound reaching the listener will be muffled. However, reflected and reverberant sound, shown in blue, reaches the listeners ears largely without interruption.

So when the obstruction scenario is detected for a sound source, the source's *direct* path (the signal path where the sound source directly feeds the 3D mix) should have attenuation and filtering applied. However, the *room* path (the signal path where the sound source feeds the 3D reverb) should be unaffected.

Occlusion

Now consider the *Occlusion scenario* mentioned earlier in the Environmental Filtering section:

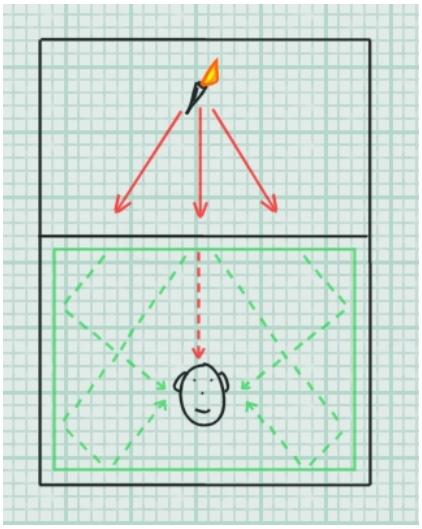


Figure 17 - Occlusion

In this instance, both the *direct* path and the *room* path will be interrupted. Hence attenuation and filtering should be applied to both components. As with obstruction, occlusion on a sound source can be simulated using the source's properties and Filter Objects.

Exclusion

Finally, consider the *Exclusion* scenario mentioned earlier in the Environmental Filtering section:

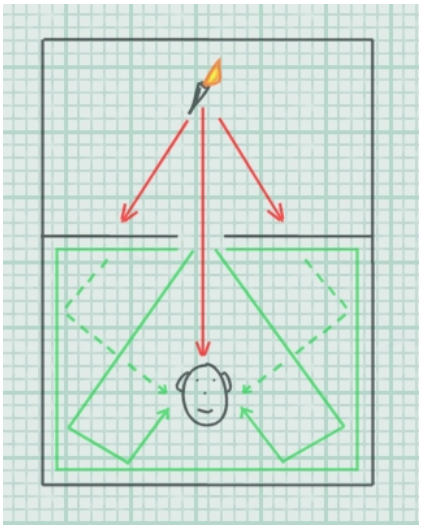


Figure 18 - Exclusion

Since the *direct* sound source to listener path is uninterrupted, but an aperture such as a window or doorway restricts the amount of *room* path heard by the listener, the resultant effect is the opposite to that heard in the obstruction scenario; muffled *room* path but un-attenuated *direct* path.

Applying muffling effects in real time

There are many considerations when determining the amount of attenuation and filtering to apply to a sound that is heard through an architectural object such as a wall of a pillar. First, it is sensible to first consider the material the sound is travelling through, and its thickness. The amount of filtering and attenuation in these situations 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.

In a dynamic audio engine, it might possible to assign different surface types with a sound transmissiveness rating. The engine could use this value when calculating filtering and attenuation figures.

Here's another useful tip for realistic occlusion effects. Compared to the direct sound path, the reflected sound should undergo 50% more frequency-independent attenuation. 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 you want to simulate *diffraction* around an obstacle in the obstruction scenario, the audio engine could determine the shortest propagation path from source to listener and set the values 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.

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*.

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 enhancements

OpenAL provides the application developer with all the basic facilities for positioning sounds in 3D. As well as adding environmental effects like reverberation and filtering, the Effects Extension also extends the positional basics such as roll-off and directivity with some subtle but useful extra enhancements. This section details how you can use the extensions 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 OpenAL.

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 constant air absorption value set by OpenAL, which is 0.994 (-0.05dB per meter). 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 constant value 0.994.

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.

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 Reverb property of the same name. This property only affects one specific sound source. The Reverb property *Room Rolloff Factor* will be added to this property to determine the additional attenuation applied to the sound source's reflected path.
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 is a low-level per-source property that 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.

Cone Outer Gain HF

Value range 0.0 to 1.0
Default value 0
Value units Gain

The Cone Outer Gain HF property enhances the directivity effect provided by OpenAL 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 OpenAL attenuates a source's direct-path sound to simulate directivity, it attenuates high and low frequency sound equally. Real world sources tend to be more directional at high frequencies than at low frequencies.

The Cone Outer Gain HF property allows the developer to enhance the directivity effect by attenuating the high frequencies more than the low frequencies. At the default setting of 1.0, there is no additional high-frequency attenuation, so OpenAL's directivity effect is unaltered. At the minimum setting of 0.0, directivity attenuation for high frequencies is 100 dB more than it is for low frequencies.
This property sets directivity high-frequency attenuation for both the direct-path and the reflected sounds of the sound source. You can turn off its effect on direct-path sound using the AL_DIRECT_FILTER_GAINHF_AUTO property, or you can turn off its effect on reflected sound using the AL_AUXILIARY_SEND_FILTER_GAINHF_AUTO property.

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