480L

Digital Effects System

Surround/HD Cart Programs

The Lexicon 480L Surround/HD (High Definition) Reverb Cartridge provides 480L users with the very latest reverberation algorithms and presets from Lexicon. The programs and presets come in two flavors – a two-input, two-output algorithm that delivers the highest quality reverberation for standard applications, and a two-input, four-output version that is ideal for multi-channel surround production. Both algorithms incorporate Lexicon's continuing work in the human perception of reflected sound. This research allows us to go beyond emulating halls and rooms. We can now create patterns of reflected energy that precisely meet the needs of recorded music – patterns that do not exist in real rooms.

Both algorithms run in "Single" mode. This means they utilize all of the DSP power available in the 480L. To maximize the available DSP power even further, the algorithms are designed for use with an external mixer. They do not include a "Mix" control.

Operating the Surround/HD Reverb Cartridge requires a 480L mainframe with v4.0 (or later) software. The 480L mainframe should be powered down whenever inserting or removing the cartridge.

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Psychoacoustic principles

The Surround/HD Reverb Cartridge uses psychoacoustic principles to separately augment both the sense of distance and the sense of space in a recorded sound. A carefully controlled randomly varying pattern of early reflections generates a sense of blend and distance. The sense of reverberation and envelopment, which depends primarily on much later energy, is created by a reverberation algorithm with a time profile tailored to the requirements of human perception. Richness and envelopment can be maximized with no change to the clarity of the original material.

The key to the design of Surround/HD Reverb Cartridge lies in the properties of the neurology we humans use to understand speech. Although speech is not the same as music, there is considerable evidence that human hearing, human speech patterns, and music have evolved together. For understanding reverberation, the most important aspect of speech perception is the neural process that separates incoming sounds into related streams of sound events.

Separation of sound events (phones) from incoming sounds

Human speech consists of streams of short sounds called "phones". When we recognize a phone as a particular part of speech we call the phone a "phoneme", the basic element of spoken language. The goal of the human speech processing system is to extract phones from the incoming sound field, identify the phones as particular phonemes, and then assemble the phonemes into understandable syllables, words, and sentences.

All of these steps are difficult, and separate areas of the brain perform each function. We are concerned here with the most basic of these functions – the separation of phones from the incoming sounds. The most basic element of this separation process is determining when each phone starts, and when it stops.

Determining when a phone starts is usually pretty easy. Phones typically have a rapid rise-time, where the sound pressure increases by many decibels in a time period of a few milliseconds. In addition, the rise time of a phone is usually uncorrupted by reverberation – room reflections arrive later. Finding the ends of phones is more difficult, particularly in the presence of room reflections.

It is not immediately obvious that we need to find the ends of phones. After all, we could simply wait for the beginning of the next phone, and assume that the previous phone stopped when the next one starts. However there is considerable evidence that human hearing does not work this way. One of the most obvious aspects of human speech perception is that people can listen to several streams at the same time. For example, when two people are talking at the same time one can choose to concentrate on either one or the other. Likewise background sounds, such as room noise or reverberation, are often easily audible even while someone is talking.

To be able to hear background sounds as separate from speech sounds it is essential to determine the end of each speech phone – otherwise all sounds would be heard as part of that particular phone. Thus there must be neural circuitry devoted to finding the starts and the stops of phones. How do we detect the ends of phones?

Finding the ends of phones is not easy because the pressure at the ears can fluctuate wildly during a phone. These fluctuations are sometimes a property of the speech sounds themselves, but mostly they are due to interference with reflected sounds. When there is an abrupt drop in level the hearing mechanism goes into a "waiting" mode. Sounds that arrive during the next 50ms are assigned to the phone that just occurred, and the separate detection of background sounds is inhibited. If the level remains low after the 50ms period, the phone is assumed to be finished. Sounds that arrive after this time are interpreted as a new sound event – perhaps another phone in the same speech stream – or a part of the sonic background.

The 50ms time limit seems to be remarkably constant among different people. It is one of the most stable time constants in human hearing. Why are we concerned with it here? Because we can exploit this time limit to give "close-miked" sounds a sense of distance without harming intelligibility. If we add lateral reflections – that means reflections in the front left and right speakers AND the left and right surround speakers – we can create a "room-like" impression. If these reflections arrive before the 50ms limit expires, they will be perceptually bound to the foreground stream. It will not be possible to hear them as separate sound events, and intelligibility will be unchanged.

Intelligibility

The intelligibility of both speech and music depends critically on the ability to detect both the beginnings and endings of sound events. In speech there is typically a 100ms gap between phones. Reflected energy (either lateral or medial) that arrives in the critical period between 50 and 150ms after the end of a phone seriously degrades our ability to separate two phones. If we want to have maximum intelligibility in our product, we must be very careful not to add much energy in this time region. Most rooms that hold between 50 and about 1000 people reduce intelligibility, because too much reflected energy is concentrated in the 50 to 150ms time range. Very large rooms can produce a sense of reverberation with less degradation of intelligibility, but successful examples of this are quite rare.

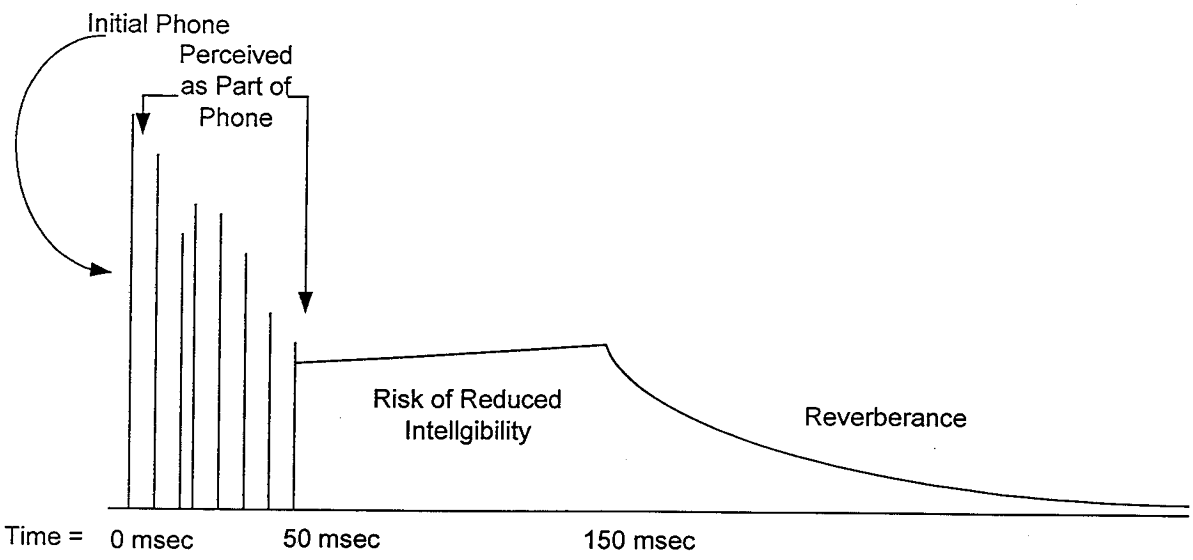
Envelopment

The perception of envelopment depends on the strength of lateral reflections – reflections that come from the sides of the listeners. When the sound source is continuous – like a legato string section – the time delay of these reflections doesn't matter very much. However when the sound source consists of a stream of sound events – like the vocals or most instruments in popular music – the perception of envelopment is much stronger. For such music envelopment is strongly dependent on late arriving reflections, which are heard as part of the background between sound events (notes). The separation process of the background from the foreground events takes time. For most people full sensitivity to the sonic background does not develop until about 160ms after the end of a sound event.

Furthermore, music tends to mask its own reverberation. Unmasking the reverberation requires more time, so that in practice reflected energy more than 300ms after the ends of sound events contributes most strongly to the perception of reverberation and envelopment.

With an understanding of distance, intelligibility, and envelopment in hand, we can design the ideal profile for reverberation. We want strong early reflections in the range of 20 to 50ms, followed by a much lower reverberant level in the 50 to 150ms range. Beyond this time range we need sufficient reverberant level to produce a satisfying impression of envelopment. Fortunately, it is possible to have an excellent envelopment impression by maintaining a constant reverberant level from about 50ms to over 300ms. After 300ms the reverberation can decay with the desired reverberation time.

This reverberant profile produces ideal results with recorded music, particularly if we separately adjust the level of the early reflections and the later reverberation. This is just what the Surround/HD Reverb Cartridge allows you to do.



About the Algorithms

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Page Two					
BAS Bass Multiply	XOV Crossover	RTC RT High Cut 1	SPN Spin	SIZ Size	RPD PreDelay
Page Three					
LBD LF Boost Delay	LBF LF Boost Frequency	DIFF Diffusion			
Page Four					
ELD Early Delay	ELS Early Spin	ELR Rear Early Rolloff	ELRA Rear Early Reflection Attenuation	RDL Rear Delay Offset	

The Surround/HD Reverb Cartridge contains two algorithms, the surround algorithm and the two-channel algorithm. Both algorithms use all the resources of the 480L, and run in "Single" mode. The two-channel algorithm is essentially the same as the surround algorithm, but the four outputs of the surround algorithm are summed internally to create a two-channel output with twice the echo density. Neither algorithm includes the "Mix" control common to most 480L reverbs. The new algorithms are intended for use with an external mixer.

Both algorithms create realistic (sometimes better than realistic) room impressions. They make extensive use of random variation to create a sound field with extremely low coloration. In the surround algorithm all four outputs – intended to be used with the front and rear speaker pairs – are separately derived. Thus all outputs are independent and uncorrelated at all times. Although the outputs are all completely independent, the outputs are coupled internally to the inputs, so a sound in one input will appear in all the outputs, just as would happen in a real room.

The algorithm is divided into two parts: an ultra low coloration reverberation algorithm, and a low coloration early reflection algorithm. Both algorithms use random variation to reduce coloration. The most important controls for each algorithm are the first two sliders on page one: Early Level and Reverb Level.

About the Parameters

Page One

ELV (Early Level)
Controls the level of the early reflection pattern – usually in the time range of 20 to 50ms. Setting this control to zero results in no output from the early reflection generator. This control affects primarily the sense of distance from the source by removing the "close-miked" sound. Usually this control should be set to maximum, and the levels of the reverberation returns should be set on the user's mixer for the best result. Note that the ELV level for the two channel program is 3 dB hotter than the surround program, so the two channel presets will typically use a value of -3 dB.

RLV (Reverb Level)
Controls the level of the late reverberation. Once the level of the returns has been found that produces the desired distance effect, the reverb level can be adjusted to create precisely the amount of reverberance and envelopment desired.

RTM (Mid Reverb Time)
Sets the decay time of the reverberation algorithm at mid frequencies. Because low-frequency reverb time (BAS) is a multiplier of RTM, RTM acts as a master control for the decay time. Similar to RT MID in the regular 480L programs.

SPR (Spread)
The time stretch in the initial decay of the reverberation. Setting the control to longer times results in a larger apparent size of the space being emulated. This control is more effective at creating a large size impression than the "size" control.

HFC (High Frequency Cutoff)
Sets a 6dB/octave low-pass filter on the outputs of the reverb algorithm. This control is extremely important for creating a realistic, unobtrusive reverberation. Typical values are around 2kHz.

LFB (Low Frequency Boost)
Controls an added boost to the low frequencies. A setting of 128 results in about a 1.5dB boost. A setting of 256 gives about a 3dB boost, and a setting of 512 gives about a 6dB boost. The delay of this boost, and its crossover frequency are adjusted with the controls on page 3.

Page Two

BAS (Bass Multiply)
The reverb time for low frequency signals, as a multiplier of the RTM parameter. For example, if BAS is set to 2X and RTM is set to two seconds, the low frequency reverb time will be four seconds. For a natural sounding hall ambience, we recommend values of 1.5X or less.

XOV (Crossover)
The frequency of the low frequency reverberation time increases as set by the BAS control. Typical values are below 180Hz for real halls.

RTC (RT HF Cut 1)
Reduces the reverberation time at high frequencies. Similar to RT HF CUT in the regular 480L programs. Typically set to about 4kHz.

SPN (Spin)
Controls the randomization rate of the reverberation – NOT the randomization rate of the early reflections. Typical value is 46 to 49. Pitch problems are generally inaudible at 46, and speech and other uncritical music may sound better with Spin values as high as 49.

Note: Spin values greater than 49 will cause audible artifacts. Unless these artifacts are desired, we recommended lower values of Spin.

SIZ (Size)
The overall delay scaling in the entire algorithm. Typical value is about 31. Smaller sizes will produce a tighter but more colored sound. Larger values will have a big sound, but may not be sufficiently diffuse for many applications. It is best to use the "spread" control to increase the apparent size of a space.

RPD (Predelay)
The delay offset of the reverberation. Typical value is 10ms. It is not recommended to use this control except for special effects. It is better to achieve the same effect with the "spread" control.

Page Three

LBD (LF Boost Delay)
The delay of the low frequency boost. Typically 80ms. Smaller delays are not as effective, and larger delays may cause the reverberation to sound "lumpy".

LBF (LF Boost Frequency)
The frequency cutoff of the low frequency boost. Typically 180Hz or lower.

DIFF (Diffusion)
Controls the very early increase in echo density. High settings of DIFF result in high initial buildup of echo density, and low settings cause a low initial buildup. To enhance percussion, use high settings of DIFF. For clearer and more natural vocals, mixes and piano music, use low or moderate settings. This control affects both the reverberation and the early reflections.

Page Four

ELD (Early Delay)
The "Early Delay" control is similar to the predelay control in the reverb algorithm, and is just as dangerous. This control should almost always be set to the preset (minimal) value. It is quite easy to push the early energy out of the time delay range that gives the most believable distance effect, and into the time delay range that reduces intelligibility.

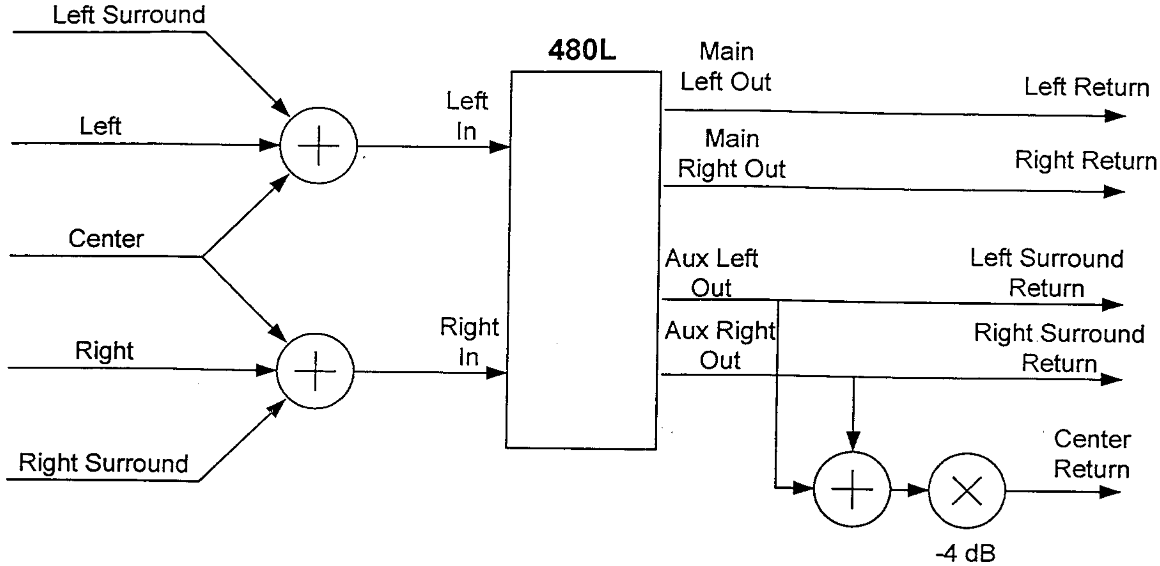
ELS (Early Spin)
The spin control for the early reflections.

ELR (Rear Early Rolloff)
This control sets a 6dB/octave low-pass in the early reflections in the rear channels. In the two-channel algorithm these reflections are directed to the front channels, but they are usually later in time than the first early reflections. ELR is used to create the sense of a larger space, where the reflections from behind have traveled a greater distance than in a small space, and thus have fewer high frequencies. Typical values for a large space are about 4kHz.

ELRA (Rear Early Reflection Attenuation)
This control reduces the level of the early reflections coming from the rear, relative to the reflection level from the front. It is used to create the sound of a large space, where the reflections from the back are less strong. In the two-channel algorithm it affects the later reflections.

RDL (Rear Delay Offset)
This control sets an additional delay for the early reflections in the rear channels. Once again it is useful for creating a larger space. However, it is just as dangerous as the ELD control. Values higher than about 20ms are not likely to work well.

Using the Surround Algorithm



When making a surround mix, the echo send busses of the mixer should be used to create a two-channel mix of the front and rear channels. Try to keep the send mix in stereo – with full left/right separation. If you are starting with a pre-made 5 channel mix, the send for the center channel should be panned to the middle, so the center goes to both inputs of the 480L. It is almost always the case that the send level of the center channel should be lower than the level of the other channels. Usually instruments and vocals panned to the center sound better if they appear to be somewhat closer to the listener than the other elements in the mix.

The outputs of the 480L should be routed to the front and rear output channels of the mix. The 480L main outputs should go to the front mix output channels, and the 480L aux outputs should go to the rear. It is usually not necessary to put reverberation into the center channel of a 5.1 mix, but if you feel this is important, the best signal to use is an equal blending of the 480L aux (rear) channels. The level of the reverberation in the center output channel should be lower than the level to the other channels – by at least 4dB.

The subjective level of the different outputs of the 480L in the mix should be equal – that is there should be approximately the same reverberation level coming from the front and the rear when you listen to the 480L outputs alone. In the "large surround" preset there will be slightly less level from the rear, as the preset emulates a large space. However the level difference is only in the early portion of the reflected energy. The later reverberation should be equally distributed around the room. As a test you may want to turn the "early level" control all the way down, while leaving the "reverb level" control up. The sound should be equal in all directions.

To use the algorithm, start by selecting an appropriate preset, and then bring up the returns on the mixer until the desired distance effect is achieved. The early reflections should remove the "close-miked" sound, giving the whole mix a sense of smoothness and acoustic reality. At this point you may want to try adjusting the "reverb level" control, to increase or decrease the reverberance. Once you have found the balance between the early energy and the late energy that works for your music, you can adjust the overall effect by changing the level of the send or the returns.

Operation of the High Definition Reverb algorithm is identical to that of the other currently existing stereo algorithms of the 480L which operate in Single mode.

About Presets

Loading Presets

The first bank of Presets is always available as Register Bank 6. To make all 50 Presets available, you must load them from the Cartridge to internal 480L registers.

Note: The following procedure will erase all Registers in the internal user Register Banks. If you have internal registers of your own and you don't want to lose them, save your own registers to a non-volatile memory RAM Cart (NVM Cart – Lexicon Part # 750-04718).

1) Power down the 480L mainframe and insert the Surround Cart. (Remember to power down before inserting or removing Carts).
2) Power up, press CTRL (Control), and go to Page 2 on the LARC.
3) Using the first fader, select the function "CART TO INT".
4) While holding the "STO" button down, Press the "REG" button.
5) When this is done, Banks 1-5 will contain all 50 presets. Bank 6 will still have the same contents as Bank 1.

Bank	1	2	3	4	5	6	7	8	9	0
Bank 1 *Combo.*	Large SR *	Small SR *	Opera SR *	Acoustisolo *	Lil Bigger *	Bigfoot *	Voc Enhance *	Large HD =	Small HD =	Acoustisolo =
Bank 2 *Surround*	Slap Hall *	Slap Room *	Chamber SR *	Church SR *	Cathedral SR *	Nice Place *	Percusspace *	Snare Amb *	Gately *	Lead Solo *
Bank 3 *"Classic" Surround*	Large Hall *	Medium Hall *	Small Hall *	Studio A *	Contemplate *	Sm Woodroom *	Lg Woodroom *	Large Chamber *	Snare Plate *	Fat Plate *
Bank 4 *HD Reverb*	Slap Hall =	Slap Room =	Chamber HD =	Church HD =	Cathedral HD =	Nice Place =	Percusspace =	Snare Amb =	Gately HD =	Lead Solo =
Bank 5 *HD Reverb*	Large Hall =	Medium Hall =	Small Hall =	Studio A =	Contemplate =	Sm Woodroom =	Lg Woodroom =	LG Chamber =	Snare Plate =	Fat Plate =
Bank 6 *Combo.*	Large SR *	Small SR *	Opera SR *	Acoustisolo *	Lil Bigger *	Bigfoot *	Voc Enhance *	Large =	Small =	Acoustisolo =

Note that in the program names, both in the table above and the LARC display, ' * ' at the end of the name denotes a surround program and ' = ' denotes a high density two-channel program.

Preset Descriptions

Large, Small, Large Surround, Small Surround
Very natural spaces with Early Reflections and subsequent Reverb in proportions that would occur in real concert halls. These are excellent for classical music.

Opera
Similar the Large and Small Surround, but optimized for voice.

Acoustisolo
Tailored to acoustic string instruments, the Early Reflections reveal the subtle nuances of the performance while the Reverb component adds spaciousness. Great for acoustic guitar, mandolin, harp, etc.

Lil Bigger
All Early Reflection, this preset does exactly what the name implies -- makes everything a little bigger. Also makes close-miked sources sound more lively without making them too wet.

Bigfoot
This specialized preset adds "bottom." Beefs up kick drums, toms and snares. Also great to run your whole mix through.

Voc Enhance
Early reflections bring out the sizzle. Warm, large hall adds body and spaciousness. Complements most singers right out of the box. Ideal starting point for any pop vocal.

Slap Hall
Delaying some of the Early Reflections in this medium sized hall helps to add presence. Great for horns, pop and gospel choirs, and backup vocals.

Slap Room
Similar to Slap Hall, but with a smaller sized Reverb and longer delay of Early Reflections. Use in place of Slap Hall when you want a smaller room. Also useful for slap-echo vocal sounds. Since the "slap back" is really a complex set of Early Reflections, the resulting effect may be more interesting to the ear.

Chamber Surround, Church Surround, Cathedral Surround
Very natural sounding, these presets are great for any application and they stand up to the scrutiny of serious classical recording.

Nice Place
Useful for fattening synth tracks. Makes them sound more lively and organic.

Percusspace
Highs are cut in the Reverb component, preserving clarity in the attack transients of percussion instruments. Ideal for multi-instrument or combined percussion tracks. The congas, bongos and other hand drums get the air that they need without the maracas, guiro or agogo's getting out of control.

Snare Amb
Originally designed for snare, this preset works well on other drums, or anything that needs added ambience.

Gately
Although not a true gate, this preset provides a very abrupt decay that is useful for coloring the aggressive sound of many pop recordings.

Lead Solo
Although not a true delay, the Early Reflections are spaced in this preset to emulate a delay effect. Makes distorted rock leads sound HUGE. Manipulation of Early Reflection times yields many interesting guitar effects.

<u>**Bank 3**</u>

These presets are derived from "classic" Lexicon stereo reverb processors and adapted for surround. The first five are from the 300/300L and the second five from the 480L.

Large Hall
Used as a great standard reference point. The hall size is big, with a gentle bloom in the reverberation envelope.

Medium Hall, Small Hall
Similar to Large Hall, but the sizes are not as big and the reverb times are proportionally smaller.

Studio A
This preset sounds like the "big" rooms ($125/hour).

Contemplate
Bright and splashy sounding. Should work well to brighten a track without getting too washy sounding.

Sm Woodroom, Lg Woodroom
Similar to Large Room programs, but with a lower BAS, simulating rooms with thin wooden paneling, or a cheaply made warehouse or auditorium.

Large Chamber
Large Chamber has few size cues. It produces a sound similar to a good live chamber with non-parallel walls and hard surfaces. Large Chamber can be used wherever a plate would normally be used, but with a more subtle acoustic sound.

Snare Plate
Snare Plate has its HFC and RTC parameters set to full range, resulting in a rapid buildup in high-frequency information. As its name implies, it has been tuned for optimal results with snare drum.

Fat Plate
Fat Plate produces the sound of a very large, highly colored plate.

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