

Dolby Laboratories Information

DOLBY PRO LOGIC SURROUND DECODER PRINCIPLES OF OPERATION

by Roger Dressler

INTRODUCTION

To many people, the term *surround* implies that something new has been added to a stereo audio signal – something requiring more than two speakers for reproduction. While this is true, it should also be realized that *stereo* itself is not confined to just two channels of sound; it can mean three, four, six, or any number so desired. Somewhere along the way, home stereo became synonymous with two channels, while at the movie theatre several multi-channel stereo formats appeared – and disappeared – in the 1950s and 60s.

By the latter 1970s, Dolby Stereo was established as a stereophonic reproduction system having from three to as many as six channels of sound to enhance the action and drama of theatrical presentations in ways only approached by two-channel systems. The most obvious feature of Dolby Stereo is that an additional channel of sound is reproduced along the sides and back of the theatre to "surround" the audience with sound.

Dolby Laboratories then devised a simple method to emulate the overall effect of Dolby Stereo in a home environment by recovering these extra surround sound effects. Introduced in 1982, Dolby Surround products today enjoy widespread popularity, with millions sold worldwide and the pace accelerating due to technical breakthroughs in home audio/video delivery formats, especially stereo television and satellite.

The advanced Dolby Pro Logic Surround system appeared in consumer products in the fall of 1987, quickly gaining favor with home theatre enthusiasts for its improved spatial articulation and expansive listening area. This leap in performance, however, came at a price; the decoder circuitry was significantly more complex, in effect limiting Pro Logic technology to high-end A/V products.

Just one year later came the breakthrough needed to make Pro Logic decoding as economically attractive as it is sonically: a custom integrated circuit was developed, consolidating a significant number of processing circuits into one package [everything shown in Fig. 10 and more]. Since that time several more analog and digital integrated circuits have become available, providing high performance multi-channel sound in a wide range of cost-effective home systems.

This paper is intended to help the reader understand how Dolby Pro Logic works. If you are familiar with the general principles of Dolby Surround, you may wish to scan ahead to Part 2 for the discussion of Pro Logic decoding. If you are new to the subject of Dolby Surround, it is suggested that you read the background material in Part 1.

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1.0. ORIGINS OF DOLBY SURROUND

Dolby Stereo movies and Dolby Surround video and television programs include an additional sonic dimension over conventional stereo productions. They are made using a Dolby MP (Motion Picture) Matrix encoder, which encodes four channels of audio into a standard two-channel format, suitable for recording and transmission in the same manner as regular stereo programs.

To recapture the dimensional properties brought by the additional channels, a Dolby Surround decoder is used. In the theatre, a professional decoder is part of the Dolby Stereo cinema processor used to play 35 mm stereo optical prints. The decoder recovers the left, center, and right signals for playback over three front speakers, and extracts the surround signal for distribution over an array of speakers wrapped around the sides and back of the theatre. (These same speakers may also be driven from four discrete tracks on 70 mm Dolby Stereo magnetic prints, but in this case no decoder is needed.)

Home viewing of movies on video has become extremely popular, and with the advent of hi-fi stereo VCR's, stereo television and laser discs, the audio side of the video presentation has improved considerably, inviting the use of full-range sound reproduction. The ability to deliver high quality audio in these formats made it easy to bring matrix-encoded surround soundtracks into the home as well, thus establishing the foundation for Dolby Surround.

1.1. The Dolby MP Matrix

One of the original goals of the MP Matrix was to enable Dolby Stereo soundtracks to be successfully played in theatres equipped for mono or two-channel stereo sound. This allowed movies to be distributed in a single optical format, and has furthermore resulted in complete compatibility with home video media (without requiring separate mixes). Since the three front channels of the MP Matrix are assembled in virtually the same way as a conventional stereo mix—left in left, center equally in left and right, and right in right—playing a Dolby Stereo mix over two speakers reproduces the entire encoded soundtrack. There is only one thing missing: the surround signal is not reproduced in its proper spatial perspective. When the first home decoder was developed in 1982, its goal was to recover this missing spatial dimension.

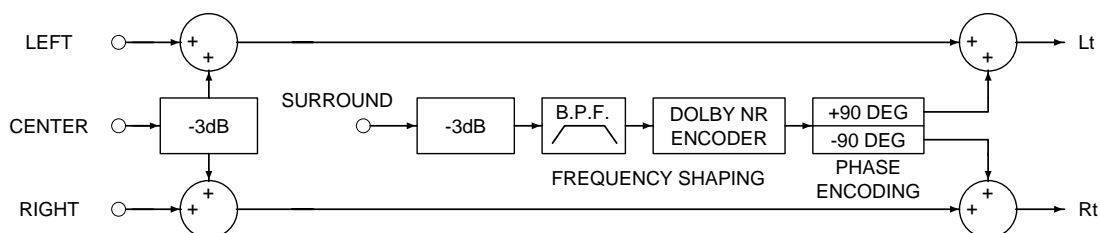


Fig. 1. Conceptual Dolby Stereo or Dolby Surround encoder.

Before we discuss decoders, however, it is necessary to see how the MP Matrix *encoder* works. Referring to the conceptual diagram in Fig. 1, the encoder accepts four separate input signals, left, center, right, and surround (L, C, R, S), and creates two final outputs, left-total and right-total (Lt and Rt).

The L and R inputs go straight to the Lt and Rt outputs without modification. The C input is divided equally to Lt and Rt with a 3 dB level reduction (to maintain constant acoustic power in

the mix). The S input is also divided equally between Lt and Rt, but it first undergoes three additional processing steps:

- Frequency bandlimiting from 100 Hz to 7 kHz.
- Encoding with a modified form of Dolby B-type noise reduction.
- Plus and minus 90-degree phase shifts are applied to create a 180 degree phase differential between the signal components feeding Lt and Rt.

It is clear that there is no loss of separation between the left and right signals; they remain completely independent. Not so obvious is that there is also no theoretical loss of separation between the center and surround signals. Since the surround signal is recovered by taking the difference between Lt and Rt, the identical center channel components in Lt and Rt will exactly cancel each other in the surround output. Likewise, since the center channel is derived from the sum of Lt and Rt, the equal and opposite surround channel components will cancel each other in the center output.

The ability for this cancellation technique to maintain high separation between center and surround signals requires that the amplitude and phase characteristics of the two transmission channels be as close as possible. For instance, if the center channel components in Lt are not identical to the ones in Rt as a result of a mismatch in channel balance, center information will come out of the surround channel in the form of unwanted crosstalk.

1.2. The Dolby Surround Decoder

This leads us to the original Dolby Surround decoder. The block diagram in Fig. 2 shows how the decoder works. The Lt input signal passes unmodified and becomes the left output. The Rt input signal likewise becomes the right output. Lt and Rt also carry the center signal, so it will be heard as a "phantom" image between the left and right speakers, and sounds mixed anywhere across the stereo soundstage will be presented in their proper perspective.

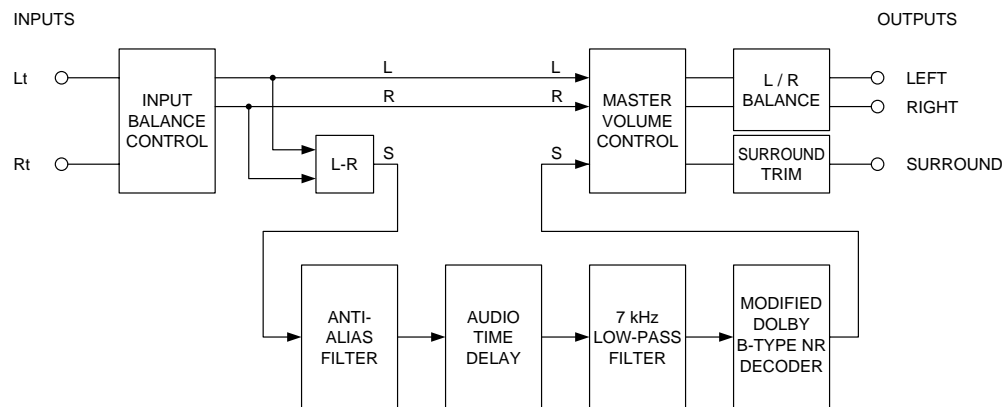


Fig. 2. Passive surround decoder block diagram.

The L-R stage in the decoder will detect the surround signal by taking the difference of Lt and Rt, then passing it through a 7 kHz low-pass filter, a delay line, and complementary Dolby noise reduction. The surround signal will also be reproduced by the left and right speakers, but it will be heard out-of-phase, which will diffuse the image.

Since the heart of the decoding process is a simple L-R difference amplifier, it is referred to generically as a "passive" decoder. This is to distinguish it from decoders using active processes to enhance separation, which are consequently known as "active" decoders.

1.3. Separation Maps

A plot, or map, can be drawn to represent the measured signal separation between any pair of channels in a matrix encode-decode process. The term "map" is used because of its similarity to a compass; it uses a circle divided into 360 degrees and has four cardinal points, in this case, four signal channels. Since the use of the MP Matrix encoder is assumed, the differences in mapped values will be a function of the decoding process itself. The map in Figs. 4 and 5 are able to show how the separation between the opposite channels (left to right; center to surround) compares with the separation between the adjacent channels (left to center; right to surround, etc.).

It should be realized that even though the electrical signals are isometrically distributed in the transmission medium as depicted by Fig. 4, the physical arrangement of the speakers in the listening room compresses three of the four channels across the front, and spreads out the remaining channel around the sides and rear (see Fig. 3.). This concentrates fully half of the spatial resolution of the system to only about one-fourth of the area of coverage, enabling a high degree of sound placement accuracy to complement on-screen images.

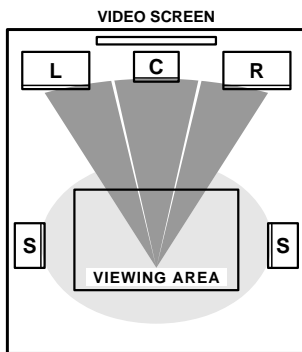


Fig. 3. Typical room layout.

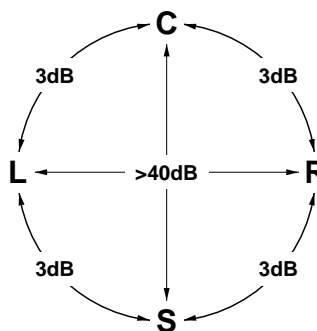


Fig. 4. Four output separation map.

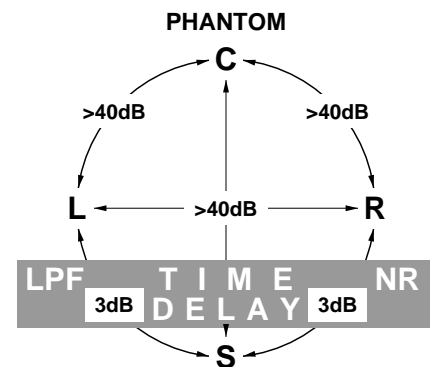


Fig. 5. Three output separation map.

The map in Fig. 4 shows the measured separation of a four-output passive decoder, which gives 3 dB of separation between any pair of adjacent channels. However, most passive decoders omit the center output, depending on the main left and right speakers to create a "phantom" center signal, which is actually the ideal arrangement for the front channels. To see why this is so, the system will be analyzed assuming that the listener is positioned equidistant from the left and right speakers such that monophonic (center channel) signals create a focused phantom image mid way between them.

The map in Fig. 5 shows the perceived separation for each of the four cardinal input signals. Given a left-only signal, the sound comes from the left and surround speakers together. An analogous result occurs with a right-only signal. With a center-only signal, the sound comes from the left and right speakers but is perceived as coming only from the phantom center position. Therefore, considering just the front channels, the passive decoder can reproduce left,

center, and right input signals with *no* perceived loss of separation. This should not be surprising—the identical psychoacoustic principle is the basis for all two-channel stereophonic reproduction, and provides the motivation for audiophiles to carefully arrange their listening room seating squarely in front of the speakers.

Additional techniques described below are employed to increase the perceived separation between the front and surround channels.

1.4. Controlling the Effects of Channel Crosstalk

The fact that some of the surround signal will continue to come from the left and right channels is actually of little consequence. One reason is that sounds are expected to come predominantly from the screen direction, since that is where we see the action. Another reason is that signals assigned to the surround track usually are not associated with specific source locations. We might see a lightning bolt on screen, but we'll hear the thunder, rain, and wind all around.

The ability of the surround channel to project its sonic image into the listening room does not rely on perfect signal isolation. However, that does not mean it is acceptable to allow crosstalk between the front and rear channels to exist unimpeded. It has been shown that the effects of crosstalk from the front channels—especially from dialogue in the center channel leaking into the surround speakers—represents the greatest potential for detracting from the presentation. By combining the effects of time delay, limited frequency bandwidth, and complementary noise reduction in the remainder of the surround decoding chain, we can invoke the psychoacoustic principles of the Haas effect, spectral modification, and signal masking, which together act to mitigate the effects of such leakage.

- **Time delay** ensures that any front channel sounds that happen to leak from the surround speakers will arrive at the listener just after the front channel sounds. This will help prevent the leakage from pulling the sound image away from the screen. Such leakage may be caused by phase or amplitude differences present in the input signals caused by azimuth misalignment or frequency response errors, or simple balance errors in the stereo program.
- The **7 kHz low-pass filter** is used for several reasons; the main one is that for a given azimuth error between the two audio channels, the leakage signal magnitude will increase with frequency, making separation at the high frequencies much more difficult to achieve. Dialogue sibilance, for example, could become quite strong and distracting from the surrounds without use of the filter. Reducing the high-frequency content also has the effect of making the surround speakers sound further away and more difficult to localize, two characteristics which benefit the persons seated closest to the surround speakers.
- **Modified Dolby B-type NR** is used to reduce noise as well as front channel signal leakage. The amount of processing was relaxed from 10 dB to 5 dB to prevent the encoded surround signal from significantly altering the nature of the left and right channel signals.

The shaded line in Fig. 5 represents that the perceived front-to-back separation is subjectively improved by the techniques just described. Together with the use of a phantom-center image, the passive decoder is quite capable of producing the intended spatial effects. Some passive decoders do offer an output for a center speaker intended to improve dialogue imaging for off-axis viewers. However, this benefit comes at the expense of a narrowed soundstage since the "passive" center output includes not just the center signal but the left and right signals as well.

From the above, it is clear that psychoacoustics plays a significant role in the success of Dolby Surround. This is equally true of the Pro Logic process, as discussed next.

2.0 PRO LOGIC DECODING

Today, A/V systems have taken on new dimensions; televisions with 27- to 35-inch screens are popular, with a move toward even larger screens and 16:9 "home theatre" aspect ratios underway. Rear projection sets from 40 to 60 inches are becoming mainstream products, as are projection systems with 6- to 12-foot screens. Larger screens coupled with increased video resolution bring the home viewer more of the movie theatre experience, and benefit greatly from expanded sonic dimensions to balance the presentation. These factors led to Dolby's introduction of Pro Logic, the second generation in Dolby Surround decoding technology.

Pro Logic is an active process designed to enhance sound localization through the use of high-separation decoding techniques. The system is a direct descendant of the one used in Dolby Stereo cinema processors, and features a center channel speaker along with the left, right, and surround outputs.

2.1. Concept of Active Decoding—Directional Enhancement

Passive decoders, as noted earlier, use a simple differential stage to extract the surround signal from the left and right input signals. The decoders maintain high channel separation across the front, but localization is proper only within the particular seating area where a phantom-center image is effective. Furthermore, even with the effects of surround channel processing, it is not possible to obtain a total degree of isolation from front to back since the surround speakers reproduce any difference information in the Lt/Rt composite. It is due to these factors that passive decoders are limited in their ability to place sounds with ultimate precision for all viewing positions.

Directional enhancement is a term referring to any technique that attempts to remove the effects of the matrix system crosstalk by manipulating the output signals of the decoder. Its goal is to create sharply focused sound images and to recreate directional cues covering a wide listening area. An active decoder can most easily be thought of as a passive decoder followed by an enhancement circuit. To illustrate the concept, we will first describe the simplest technique for directional enhancement—gain riding. Fig. 6 shows how the gain of each output is varied with a voltage-controlled amplifier (VCA).

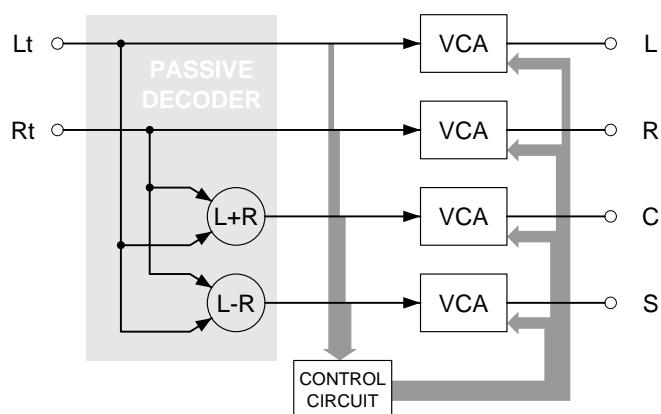


Fig. 6. Gain riding concept.

Consider the case where a single sound—dialogue—is present in the center channel ($L_t=R_t$). The passive decoding map in Fig. 4 shows that the center output will carry the dialogue, but so will the left and right outputs only 3 dB down. With control circuits, the decoder can decide to eliminate the unwanted dialogue leakage by turning down the gain of the left and right speakers, leaving only the desired center output audible. The same procedure may also be used to isolate the left channel output by turning down the center and surround outputs when only an L_t input signal is present. In fact, for an input signal encoded at any position in the 360-degree circle, the separation can be completely restored with this simple method to equal that of a discrete four channel system.

2.2. Problems with Gain Riding

Unfortunately, soundtracks are rarely composed of single, isolated sound elements. Consider how the described gain-riding decoder would work when the dialogue is mixed together with stereo background music. We would like to hear the music from the left and right speakers and the dialogue from the center speaker. First, let's examine the passive decoder outputs before separation enhancement. Just as before, dialogue comes from the center, with dialogue leakage in the left and right. The stereo music comes from left and right, with music leakage in center ($L+R$) and in surround ($L-R$).

Since dialogue is dominant in the mix, it controls the gain-riding circuit to reduce the dialogue leakage in the left and right outputs. But this action also cuts off the stereo music, leaving only the monaural sum in the center channel and the difference signal in the surround channel. When the dialogue stops, the control circuit will restore the gain in the left and right channel outputs, allowing the music to be heard in stereo once again. The hope is that when dialogue is present, it will mask the fact that the music has been mostly removed.

Not only does this fail to work, but consider what happens during the transition when the dialogue starts and stops: the music coming from the left and right speakers goes off and then back on. This is quite audible, resulting in "pumping" of the non-dominant sounds (the music) in response to the dominant sounds (the dialogue). Such gain-riding techniques were used during the quadraphonic era in some SQ decoders having "logic enhancement" circuitry, and were considered aesthetically inadequate for musical reproduction because they gave the sound an unstable character and caused the soundstage to wander. Whatever measurable benefits the gain-riding decoder made on the test bench were lost in the realities of the listening room.

2.3. The Cancellation Concept

Another way of eliminating dialogue leakage in the left and right outputs is shown in Fig. 7. By taking the right output signal, inverting its polarity and blending it into the left output, the center signal components—being equal and opposite—cancel each other completely, thus eliminating the dialogue leakage in the left output. This is the basic cancellation principle, which can be implemented in several ways depending on the final performance goals of the system.

In this example, though, the music did not emerge unaltered from the left channel after the center signal was removed; the inverted right signal was mixed with it. The right output will also receive the inverted left channel signal in the process of canceling its center leakage components. This is an unavoidable consequence of the cancellation process. Furthermore, there will continue to be some music from both channels mixed together ($L+R$) in the center channel output. However, the fact that the music signal has been redistributed spatially *does not* mean that its overall level has been affected, as occurred unavoidably in the gain-riding system. With careful

design, it is possible to maintain constant signal power for all components of the soundtrack as they are redistributed to the various speakers.

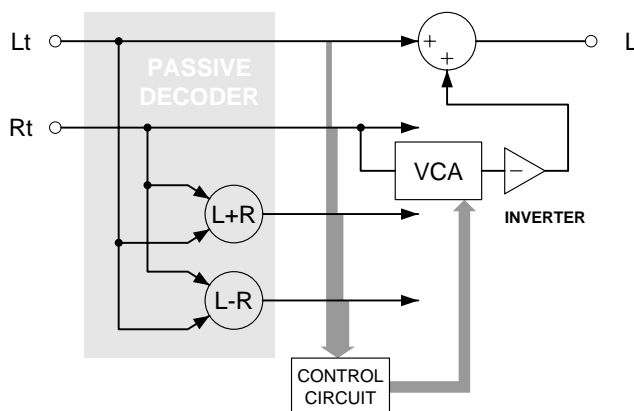


Fig. 7. Signal cancellation concept.

2.4. Constant-Power Concept

As the decoder operates, the dominant sounds are focused to the main point of origin within the 360-degree range, while the non-dominant sounds are redistributed among the remaining speakers. The principle here is that the dominant sounds will temporarily limit the listener's ability to detect a change in directionality of the non-dominant sounds. Since no overall change in loudness occurs for any of the signal components, no pumping or signal modulation will be perceived.

The concept of one signal masking another is well documented and effectively used in Dolby noise reduction systems. In these systems, the dominant signal (the music on the tape recording) is some 20 to 40 dB greater than the tape noise level in the range where the NR circuit is dynamically active, but even so, the noise must be handled carefully to prevent audible "noise modulation." In movie soundtracks, however, the level difference between concurrent sound elements is often less than 20 dB, and there may be little or no difference in level among them for sustained periods. Therefore, dominant sounds are much less effective as a tool for masking level changes in non-dominant sounds, since the non-dominant sounds can be plainly heard.

In the example, we assumed the dialogue was much louder than the background music, thus calling for maximum action from the decoder. More frequently, however, the differences in sound levels are not this great. As the sounds become closer in level, each one begins to mask the crosstalk of the other, concealing the fact that separation is not perfect. When directional enhancement is then applied, less of it is needed to improve the sense of localization, so less redistribution of the non-dominant signals takes place.

It may also be desirable to suspend directional enhancement altogether, with the decoder becoming passive at certain times. For instance, the sounds of rain or wind, which are so involving on a subliminal level, are intended to come from all speakers simultaneously. This requires no sound source localization, and hence, no directional enhancement.

By maintaining constant signal power and by applying directional enhancement only as needed to preserve good localization, we can rely upon a modest degree of masking to be sufficient to cover the fact that the non-dominant signals have been directionally redistributed.

2.5. Nature of Signal Dominance

A dominant sound is simply that—the sound that is most prominent in the mix at any given instant in time. It is necessary to be able to sense when a dominant sound occurs because it will have the greatest influence on the perception of "discreteness" or the effective separation of the soundtrack.

The highest degree of dominance occurs when all sounds are placed in one location. Remember that in a passive decoder such a signal will be reproduced from its intended output location and, undesirably, from its adjacent channels. There will be no other sounds present to help mask the leakage. The condition of a purely dominant signal thus exposes the leakage most clearly, but this is also a condition where the side-effects of directional enhancement can be most easily and completely suppressed.

At the other end of the scale, sounds with similar intensities tend to confuse the listener's ability to pinpoint their individual locations, thus needing little or no directional enhancement. However, while two different sounds may seem to have the same average loudness, it is likely that, on an instantaneous basis, one of them will be dominant over the other and that the dominance will continuously alternate between them. Depending on the peak-to-average ratios of the sounds over time, it may or may not be necessary to provide directional enhancement.

This suggests that the decoder must include two additional characteristics in order to work effectively. The decoder first must be fast enough to provide enhancement on an instantaneous basis between two or more encoded positions when the signal peaks are prominent enough to be heard as individual events; in effect, time-division multiplexing its action among several individual sounds occurring in rapid succession. Even though the decoder is essentially providing directional enhancement for sounds at only one position at a time, all of them are perceived as being separate from each other. The second characteristic is the ability to sense when the relative dominance falls below the point where it is no longer necessary for the decoder to provide any substantial directional enhancement, but, were it to continue its rapid operation, might create an indistinct or audibly "nervous" soundfield.

For these reasons, the Pro Logic decoder has been designed to sense the level of dominance in the soundtrack. When dominance is at very low values, the decoder stays in a "slow" yet fully operative mode to give a stable feeling to the soundfield image. Above a certain point, the decoder can ideally shift to a high speed mode to quickly process each individual sound element.

2.6. Detecting Soundtrack Dominance

Since it is the relative level of one sound to another which determines the perception of separation, it is desirable to have sensing circuits that ignore absolute signal level in favor of being responsive to the *difference* in level between two signals (the equivalent of taking their ratio). Electrically, this is no simple task, but, by taking the logarithm of each signal and subtracting one from the other we, we can obtain a measure of relative dominance.

The resultant control voltage—in this logarithmic form—closely follows the way loudness is perceived. Consequently, the final process provides separation enhancement directly corresponding with the amount needed to prevent crosstalk from becoming audible, and proportional to the ability of the dominant sound to mask the spatial redistribution of the non-dominant signals.

2.7. Sensing Direction of Dominance

Knowing which signal is dominant includes knowing the encoded position, or angle, of the signal. It is in this direction that enhancement must take place, and may encompass any point in a 360-degree circle, not just one of the four cardinal positions.

By definition, dominance can only occur in one place at any instant in time; it cannot exist in two places simultaneously, since their equality of magnitude would mean that neither is dominant. (These two signals may together constitute a single dominant quantity, however.) Therefore, it is sufficient to be able to detect a single direction of soundfield dominance, no matter how rapidly the soundfield changes. With two independent, orthogonal signal pairs available in the encoded soundtrack (the left/right pair and the center/surround pair), it is possible to identify any point on an X-Y coordinate plane within a given boundary area.

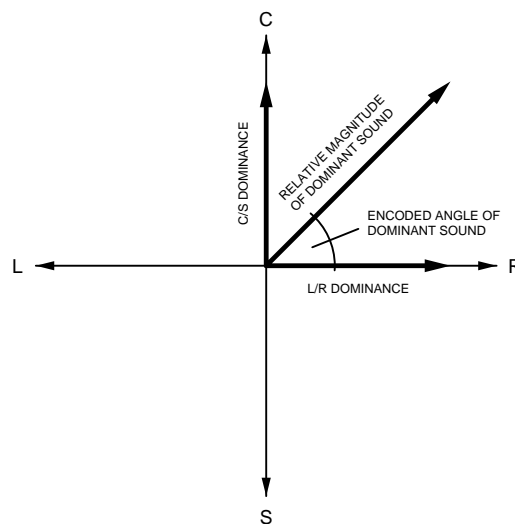


Fig. 8. Resolving the dominance vector.

In Fig. 8, the left/right pair rests on the x-axis, and the center/surround pair is on the y-axis. Resolving the magnitudes of the signals along each axis and converting from rectangular to polar coordinates then gives the necessary information. Dominance is now shown as a vector quantity; the angle represents the encoded angle of the dominant sound, and the magnitude represents its relative dominance.

2.8. Pro Logic Adaptive Matrix

Just as the L-R differential stage is the heart of a passive decoder, the adaptive matrix is the heart of a Pro Logic decoder. Two main signals go in (Lt and Rt), and four resultant signals emerge (L, C, R, and S). Compare the block diagram of the Pro Logic decoder in Fig. 9 with the passive decoder in Fig. 2. Except for the matrix stage (and center output), they are virtually the same.

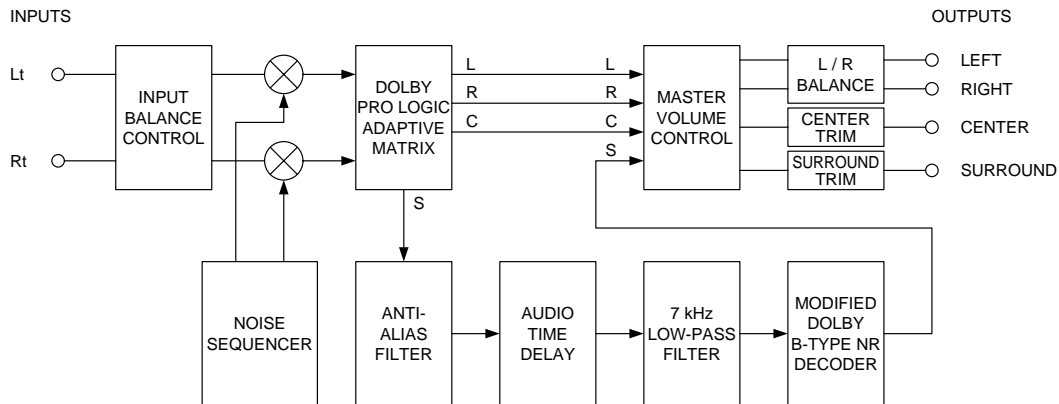


Fig. 9. Pro Logic decoder block diagram.

To summarize, Pro Logic operates by continuously monitoring the encoded soundtrack, evaluating the inherent soundfield dominance, and applying enhancement in the same direction and in proportion to that dominance. To see how the circuit works, we will examine the adaptive matrix block diagram in Fig. 10.

Notice that the adaptive matrix employs two parallel paths: a relatively direct audio path (L and R inputs go straight to the combining networks) and a complex control path. Most of the decoder's electronics are used to condition and analyze the input signals rather than to actually process the audio itself.

As you may guess, the main order of business is to generate the signal dominance vector. The first step in this process is to condition the incoming signals to prevent decoder errors. This is done by bandpass filtering the Lt and Rt signals to strip off strong low-frequency signals which do not provide directional cues, and to attenuate high frequencies that may contain uncertain phase or amplitude characteristics.

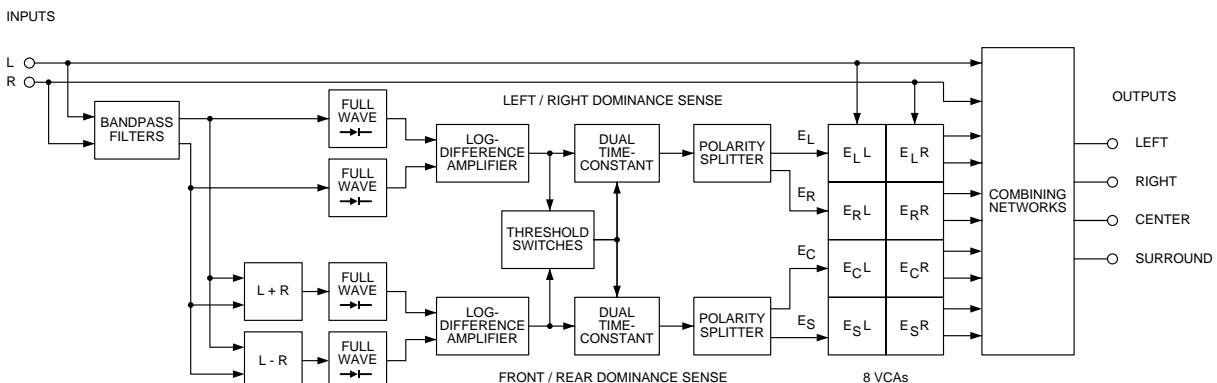


Fig. 10. Pro Logic adaptive matrix.

The next step is to determine the magnitudes of the two orthogonal signal pairs, which is done by first full-wave rectifying each cardinal signal, then subjecting the resulting DC voltages—in pairs—to log conversion, and finally taking their difference. Two independent control signals are now

available; one represents dominance along the left/right axis, the other along the center/surround axis. Even though there are only two control voltages at this point, they are of dual polarity, or bipolar. For instance, when the left/right voltage deflects upward, the dominance is to the left; when it goes downward, the dominance is to the right. At the midpoint, no dominance is indicated.

Each of these control voltages is evaluated continuously to determine if their relative dominance exceeds a certain threshold point. If either one does, the control circuit switches to the fast mode of operation.

A polarity splitter resolves the two bipolar dominance signals into four unipolar control voltages, called E_L , E_R , E_C , and E_S . They now represent the soundtrack dominance in electrical terms embodying psychoacoustic properties, so are ready to be applied to the signal-canceling VCA stages. Since there are two input channels (Lt and Rt) and four control voltages, eight VCAs are used to generate eight variable sub-terms. When added with the Lt and Rt inputs, ten individual terms are available. To construct a decoded output signal, portions of each of the ten terms are added or subtracted with a predetermined weighting factor in the combining networks. Selection of the appropriate magnitudes and polarities for the forty summed components gives the desired directional enhancement and non-dominant signal redistribution, all the while maintaining constant acoustic power for the signal components.

2.9. Channel Separation Specifications

In this era of digital audio systems where channel separation may be on the order of 90 dB, it is useful to try to understand just how much separation is necessary in a surround system. The matrix encode/decode transmission path is not intended to convey unrelated signals, such as dialogue in four languages, each meant to be heard by itself. Rather, it will carry sound elements expressly assembled to be heard together, plus enough information to reproduce a coherent and controllable soundfield. These are very different requirements.

On the one hand, any degree of channel crosstalk will result in just that—dialogue becoming audible from one channel while listening to another, something difficult or impossible to ignore. On the other hand, a surround system is only required to provide enough apparent separation to create unambiguous directional cues. Furthermore, the crosstalk in a surround system is always correlated with the main signal of interest. For example, the crosstalk from center channel dialogue is just this same dialogue leaking from the adjacent channels; it is not some signal unrelated to the program.

Another point to remember is that the ability for correlated crosstalk to cause a shift in apparent sound source location diminishes as the angle between the speaker locations reduces. If a sound is intended to come from one speaker, but it is also reproduced in another speaker placed right next to it, the sound will still seem to be coming from the correct location. Only as the second speaker is moved away from the first will the image shift away from the intended position.

From this, it may be concluded that it is important to provide more separation between the opposing speaker locations rather than the between the adjacent positions, since the greater spacing increases the possibility for the leakage to disrupt the soundfield. It has already been shown that the MP Matrix has much higher separation between the opposite channels than the adjacent channels. The critical surround channel, which is diametrically positioned relative to the three front channels, has additional signal processing to give an extra degree of separation to maintain a forward-focused soundfield.

In order to get a feel for how much separation is needed to prevent the correlated crosstalk in a surround system from disrupting image placement, you can try the following experiment with a two-speaker stereo system. Select any signal source, music or dialogue, and play it in monaural from both speakers. While seated in the normal listening position, have someone turn the balance control completely to the left, so that the right speaker is off. Then, while the control is rotated slowly toward the center position, note the point where you first hear the right speaker come on, or where you can detect that the image has moved away from the left speaker. If you then unplug the left speaker, you will hear how loud the right speaker had to be to make this happen. Depending on the angle between the speakers and the listener's acuity, the right speaker will be only about 10 to 20 dB lower than the left one, substantially less than what is needed to prevent *unrelated* signals from interfering with each other.

The results of the above experiment correlate with the separation needed between the left and right channels of a surround system; somewhat less is needed between the center and the left or right channels since the distance between them is only half as much. The separation map for a typical Dolby Pro Logic decoder is provided in Fig. 11, showing that about 30 dB of separation is available between any pair of channels. It should be noted that the results of the above experiment represent the separation needed for a purely dominant sound source. The map figures similarly represent separation measured with a purely dominant, steady-state test tone signal.

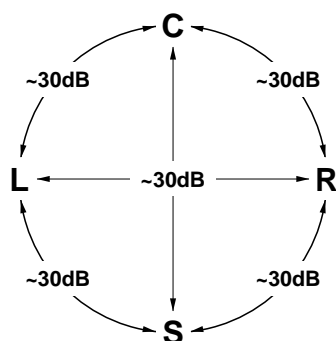


Fig. 11. Typical Dolby Pro Logic separation map

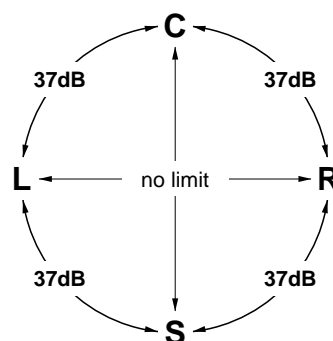


Fig. 12. Theoretical Dolby Pro Logic separation map

The map of a theoretical Dolby Pro Logic decoder is shown in Fig. 12, which, under these same test conditions, shows that the worst-case adjacent channel separation reaches 37 dB, and the opposite channel separation is not limited at all. Due to the precision and repeatability of digital signal processing, these separation figures can be approached in DSP-based Pro Logic decoders, but as was pointed out earlier, there is no improvement in perceived sound quality brought by these sometimes dramatic, measurable differences.

2.10. Spectral Partitioning

It is possible to assign certain portions of the audio signal to different speakers or locations without detrimental side-effects. One popular technique is to use a single woofer to carry the bass for both channels of a stereo system. Even though the two satellite speakers do not reproduce the bass, so long as it is present in the room, we are satisfied that the system is reproducing the entire frequency spectrum. A relatively small center channel speaker can be made to sound much larger than it actually is by using this same technique, either with a separate woofer or simply by diverting the bass to full-range left and right speakers. Since it is of

fundamental importance to include the center speaker, a bass-splitting function is included in all Pro Logic decoders.

This principle also applies to the surround speakers. With the bass being well reproduced in the listening room by the front speakers, there is little value in having the surround speakers attempt to duplicate it. Furthermore, since the front three speakers are also reproducing the complete high-frequency spectrum, it will not be readily apparent that the surround speakers are rolling off above 7 kHz. Only if the surround speakers were playing a wideband signal in total isolation would their limited frequency coverage be exposed—something that is rarely, if ever, done.

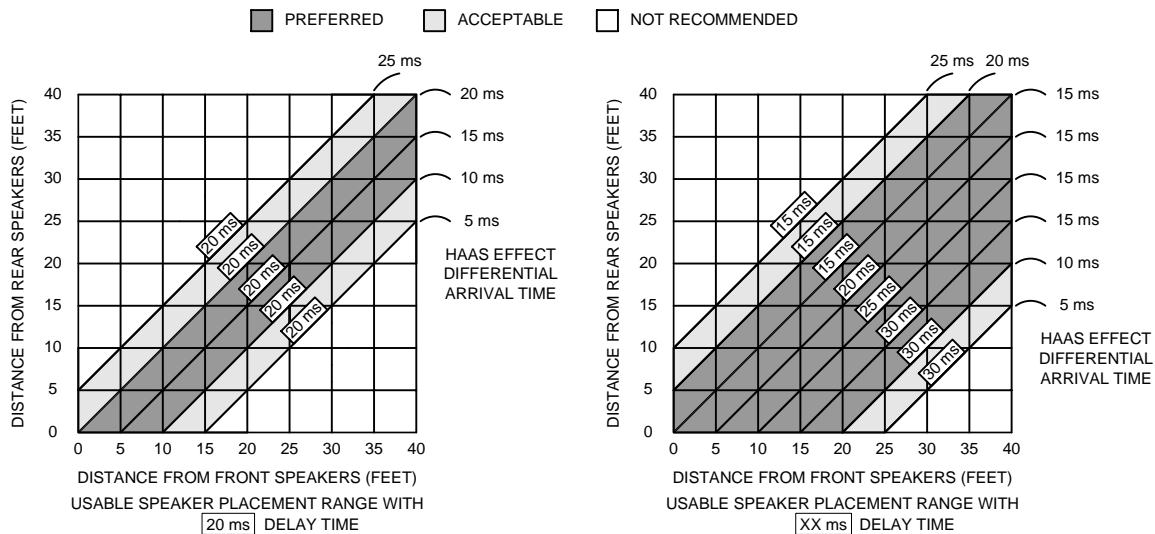
CONCLUSION

The Dolby Pro Logic Surround decoder system is the final link to bringing soundtracks with greatly expanded spatial capabilities to the consumer. Due to the parameters of the Dolby MP Matrix, the encoded soundtracks are fully compatible with two-channel stereo reproduction, and also give outstanding results with monophonic reproduction. With advent of Pro Logic decoding, conventional two-channel media can now offer several levels of performance to the consumer, all from a single version of the soundtrack. The MP Matrix thus utilizes two-channel media more effectively than any other system, and the Pro Logic decoder provides the means to extract multi-dimensional properties with an accuracy previously unattainable in consumer equipment.

Appendix

Time delay is often used to create an echo effect, which can help give music reproduction a feeling of greater spaciousness. However, that is *not* why it is used in Dolby Surround decoders. The real reason is to improve the sense of clarity and directionality of front channel sounds. This is done by taking advantage of the "Haas" or precedence effect, which enables the main frontal sound to arrive at your ears before the surround sounds. The time delay stage compensates for the travel time of sound through the air, which takes about 1 millisecond per foot distance. By knowing how far the listening position is from the front and the surround speakers, it is possible to adjust the time delay for optimal results.

There are generally two kinds of time delay available in Dolby Surround products: adjustable, or unadjustable (fixed delay). As would be expected, the adjustable delay allows a wider range of distances to be used than the fixed delay. By using the appropriate graph below, the delay time adjustment setting or the physical speaker-to-seating arrangement can be brought into the ideal relationship.



Graph for **fixed**
time delay decoders.

Graph for **variable**
time delay decoders.