Old and New Techniques for Artificial Stereophonic Image Enhancement 4371 (K-6)

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Abstract

Artificial stereophonic image enhancement techniques for loudspeaker reproduction are well known in the literature. The general goals of such methods are to simulate the desirable properties of a natural listening environment, to reduce the dependence of the reproduced sound on the peculiarities of the listening room, and to broaden the perceived width and depth of the reproduced sound field--ideally to dimensions beyond the physical locations of the loudspeakers. In this paper a representative group of stereophonic image enhancement techniques are reviewed and several new strategies are proposed.

1. Introduction

For more than 60 years--especially since the widespread introduction of stereophonic sound reproduction in the 1950's--methods have been proposed to allow manipulation of the spatial characteristics of sound recordings [1-6]. These techniques range from microphone placement and simple panning to more complicated methods involving filtering and stereo reverberation. Similarly, there is often the need or desire to alter the stereo properties of an existing two-channel recording in order to reduce the shortcomings of conventional loudspeaker reproduction in typical listening environments. For example, if the listener is not located equidistant from the two loudspeakers or if one or both of the loudspeakers are located near reflective surfaces or other obstructions, the listener will generally perceive a degraded stereo impression. In either of these examples it may be desirable to forego the precise imaging

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properties of the original stereo recording in favor of a broader, less correlated stereo signal [7].

A new and potentially important application of artificial stereophonic image enhancement is in the area of multimedia computing. Personal computers with stereo sound reproduction capability often utilize small loudspeakers positioned immediately to the left and right of the computer's video display. The resulting sound field is generally very narrow and unrealistic. This situation is less than ideal for reproduction of either recorded music for entertainment or convincing sound effects for computer games. Spatial enhancement techniques can be applied to improve the listening experience for the computer user.

For mono to stereo conversion the primary requirement is to simulate the mutually incoherent left and right signals that would occur naturally if the input signal was recorded stereophonically in a reverberant room. Examples include a delay or phase shift between the channels, a stereo reverberator or decorrelator, a pair of complementary comb filters, and various combinations of techniques [6].

Artificial image enhancement can be *single-ended,* meaning that the input is a conventional mono signal or stereo pair, or *complementary,* meaning the input has been prepared (*matrixed* or *encoded*) to allow the recovery of several output channels from a smaller number of transmission channels.

Many single-ended image enhancement schemes with stereo input make use of the channel difference signal (L minus R) in order to emphasize the difference between the left and right signals. For example, if the L and R signals contain a substantial mono (common) component it is possible to express $L = M + L_o$, and $R = M + R_o$, where M is the mono signal and L_o and R_o are the left-only and right-only components. In this situation $L-R = L_o-R_o$, so adding L-R to L gives $M + 2L_o - R_o$, which boosts the proportion of L_o in the composite left signal. Similarly, subtracting L-R from R performs the same operation on the right channel. Furthermore, the presence of the inverted components ($-R_o$ in the left output and $-L_o$ in the right output) also serves to give a broadened spatial impression to the resulting stereo sound field.

More sophisticated spatial enhancement schemes involve frequency- and level-dependent processing to achieve a more noticeable change in the stereo image. Commercial examples of this procedure include products from SRS [8] and Spatializer [9] (single-ended), and the Dolby Pro Logic Surround system [10] (complementary).

Another type of spatial enhancement procedure that is suitable for use as a preprocessor involves the use of special transfer functions (filters) to generate a stereo output signal from a monophonic source. The special transfer functions are chosen to create the impression that the sound is emanating from a particular direction in space, and can incorporate a model of the directional characteristics of the human hearing apparatus known as head-related transfer functions (HRTFs) [11]. Because the processing is done prior to the creation of the stereo signal, the transfer function approach can be used either during the mixdown step of music production or on-the-fly in the reproduction system. Commercial examples employing transfer function processing are the QSound [12] and Crystal River Engineering [11] systems.

A technique specifically for two-channel loudspeaker playback in a single-ended configuration is *crosstalk compensation* (CTC) [13-15]. In CTC, the signals delivered to the loudspeakers are preprocessed in order to reduce or eliminate the crosstalk, or signal leakage, from the left loudspeaker into the right ear and from the right loudspeaker into the left ear. The actual cancellation occurs in the air by deliberate interference between the acoustical signals arriving at the listener's ears. Under ideal conditions the perceived stereo image can be placed anywhere over a 180 degree azimuth range.

The organization of this paper is as follows. First, a general survey of complementary and single-ended artificial stereo image enhancement schemes is provided. Second, specific examples of several existing techniques are presented along with a discussion of their areas of application. Next, a new artificial image enhancement technique employing interchannel adaptive processing is described. Finally, a summary and some recommendations for future work are given.

2. Complementary Image Enhancement

Complementary stereophonic image enhancement involves an encoding process where *N* (three or more) channels of audio material are combined into typically only two audio channels for distribution, followed by a complementary decoding operation in which approximations of the *N* original audio channels are reconstructed. The common nomenclature for these systems is *N*-2-*N*. Of course, it is not possible in general to perform a perfect reconstruction of the original audio channels under these conditions, but various methods can be employed to enhance the apparent separation of the output channels [6].

2.1 Quadraphonic matrix history

Among the significant complementary systems of historical interest are the analog four-channel matrix systems (4-2-4) proposed for quadraphonic consumer audio in the late 1960's and early 1970's. These systems were referred to as matrix systems because the encoding and decoding operations could be expressed mathematically as the product of a signal matrix and a coefficient matrix. The first such 4-2-4 matrix for quadraphonic sound was proposed by Peter Scheiber in 1969, and was quickly followed by the Sansui QS matrix and the CBS SQ matrix [6]. An important feature of the Scheiber, QS, and SQ matrix systems was that the two encoded channels were largely compatible with conventional two-channel stereo playback systems, meaning that a pleasing stereo mix could be enjoyed even if the 2-4 matrix decoder was not available.

2.2 Passive and active decoding

All of the quad matrix systems utilized some sort of signal-dependent gain mechanism in the decoder which detected the dominant program material and adjusted the channel gain or matrix coefficients to reduce the audibility of the inherent signal crosstalk between output channels. This signal-dependent adjustment is known as *active* decoding, as opposed to *passive* decoding in which the matrix coefficients and gains are fixed. Active decoding is more complicated--and therefore more expensive--than passive decoding, but the active systems provide the listener with a much stronger localization impression for the dominant program material.

2.3 Dolby Stereo and Dolby Surround

Despite the commercial failure in the 1970's of 4-2-4 matrix systems in the consumer marketplace, many of the important techniques introduced at that time are retained in the so-called surround sound systems of today. The most important commercial examples are the Dolby surround sound products [10, 16].

In the late 1970's Dolby Stereo was introduced for use in motion picture theaters. In its most basic form Dolby Stereo employs an analog 4-2-4 scheme, known as the Dolby MP (Motion Picture) Matrix, to encode left (L), right (R), center (C), and a processed version of surround (S_p) into two compatible stereo signals, L_t and R_t, forming the optical soundtrack on the film. The complementary MP decoder produces four output signals (L′, R′, C′, and S′) and delivers L′, C′, and R′ to the corresponding speakers in front of the audience and S′ to one or more speakers located at the rear and to the sides of the cinema [10]. An active decoding system is generally employed in Dolby Stereo theater systems.

A simplified and low-cost home version of Dolby Stereo, introduced in 1982, is known as Dolby Surround [16]. Dolby Surround initially used a simple passive decoder system to produce L′, R′, and S′ from two-channel input signals via encoded stereo VCR tapes, stereo TV broadcasts, etc., as shown in Figure 1. As with the early quadraphonic systems, the encoded Dolby two-channel signals are compatible with conventional reproduction systems, providing a pleasing stereo (and even mono) mix for playback on systems without decoding hardware. Subsequently, an active decoding scheme, Dolby Pro Logic Surround (Figure 2), was introduced in 1987 to give the consumer market the desirable active decoding features found in the earlier Dolby Stereo theater systems [16].

In summary, complementary stereophonic image enhancement is effective and has been embraced enthusiastically by the consumer marketplace, albeit primarily in the context of processed soundtracks for TV and home video. Complementary systems offer an important advantage that the audio producer knows exactly how the final product will be decoded, and can therefore take best advantage of the system in a creative sense.

3. Single-Ended Image Enhancement

Unlike complementary (encode-decode) systems, single-ended image enhancement schemes either involve preprocessing of the stereo signals prior to distribution and conventional playback, or postprocessing of a conventional stereo signal at playback time.

Preprocessing for image enhancement can be thought of essentially as an extension to the conventional studio production techniques (panning, delay, reverb, and so forth) commonly used to prepare two-channel stereo recordings for distribution on CD, cassette, etc.: the audio producer "packages" the sound as desired for playback on ordinary stereo equipment.

Postprocessing, on the other hand, is accomplished with circuitry only in the playback system and is therefore required to operate with arbitrary two-channel stereo input signals without any explicit cooperation from the audio producer. Now, given a two-channel stereo audio signal the question becomes: what can be done via postprocessing to increase spatial perception? Since no encoding is done, the postprocessing system must rely only on L, R, and derived relationships between L and R in order to enhance or exaggerate any spatial content of the stereo stream. Thus, the degree to which the impression of spatial enhancement is achieved depends largely on the cleverness of the algorithm designer and the validity of the designer's *a priori* assumptions about the signals of interest.

In short, preprocessing requires the audio producer to make use of special hardware or software during mixdown, but the resulting "enhanced" stereo mix can then be played on any conventional stereo system, while postprocessing operates with any conventional stereo signal pair, but without deliberate benefit from the intentions of the producer.

3.1 Mono-to-Stereo Conversion

Many techniques exist for generating a two-channel stereo signal from a monophonic (one-channel) source. The primary method, *panning*, simply involves an amplitude control that adjusts the proportion of the mono signal applied to the left and right stereo channels.

Image broadening can also be achieved through the generation of frequency-varying amplitude and phase differences between the two output channels. The interchannel differences can be introduced in the form of a short delay or phase shift, a pair of overlapping comb filters (Lauridsen's method) [17], a stereo decorrelation or reverberation scheme [7], or a variety of other strategies [6].

3.2 Crosstalk Compensation

An inherent physical characteristic of loudspeaker reproduction of two-channel stereo signals is *crosstalk*. Crosstalk refers to the acoustical paths that exist--in addition to the direct left-to-left and right-to-right paths--between each loudspeaker and the *opposite* ear (left-to-right and right-to-left) of the listener. The situation is entirely different for headphone listening, of course, because no acoustical crosstalk exists between the listener's ears. The crosstalk paths are a completely normal and expected aspect of conventional loudspeaker stereo, but the effect tends to prohibit creation of localized sound images beyond the physical extent of the loudspeakers.

It is possible (with some restrictions) to eliminate acoustical crosstalk with loudspeakers by inserting a linear time-invariant two-port electrical network prior to the speakers that is the inverse of the expected acoustical crosstalk function. In this way, recursive signals are generated such that the original crosstalk components are acoustically canceled at the positions of the listener's ears, as depicted in Figure 3. In other words, the crosstalk compensation (CTC) network can convert the normal loudspeaker listening experience into a binaural experience [14, 15].

In a properly assembled listening area the impression when listening to a CTC system can range from compelling to stunning. However, various practical constraints limit the universal appeal of such systems. For example, the CTC network depends to a great extent on knowledge of the geometrical relationship between the loudspeakers and the listener's ears. Any deviation from the assumed geometry has a deleterious effect since the acoustical signals intended to cancel the crosstalk will no longer arrive at the ears with the proper amplitude and phase. Moreover, reflections from the walls, floors, and other surfaces of the listening room conspire to interfere acoustically with the CTC effect.

However, the outlook may be significantly better for CTC in multimedia computing since the listener is typically located in a known position in front of the display screen and between the loudspeakers. Thus, provision for CTC in personal computer-based audio systems seems desirable and appropriate if the required hardware and/or computational resources are available.

3.3 Commercial Spatial Enhancement Methods

Several commercial examples of single-ended image enhancement systems are available in the marketplace. Three notable patented processes are considered next.

3.3.1 Klayman Patent (SRS)

U.S. Patent #4,748,669 (filed 12 Nov. 1986, issued 31 May 1988) [8] describes a stereo image enhancement scheme invented by Arnold Klayman and currently marketed by SRS Labs, Inc. In the SRS (Sound Retrieval System) process L+R and L-R signals are calculated from the two-channel stereo input. A signal-dependent equalization of L+R and L-R is performed in several frequency bands, then the enhanced output signals L_o and R_o are constructed by additively combining the unprocessed L and R signals with the processed L+R and L-R signals. The gain applied to the processed L-R signal is adjusted automatically by monitoring the level of L+R and L-R at the input and the level of the processed L-R signal prior to the output mixer. The equalization applied to the L+R and L-R signals is used adaptively to *whiten* the sum and difference signal spectra, i.e., to boost weak spectral bands relative to strong spectral bands. The invention is summarized in Figure 4.

The patent also discloses a "perspective correction" technique in which the L_o and R_o signals from the spatial enhancement process are subsequently passed through summing and differencing circuits to create L_o+R_o and L_o-R_o. In the case of conventional stereo playback with the loudspeakers located in front of the listener, fixed equalization is applied to the output difference signal to simulate the spectral cues that would be present in natural surroundings if the sound was actually emanating from the sides of the listener. Following equalization of the difference signal the left and right processed output signals are assembled via $L_p=0.5*\{(L_o+R_o)+EQ(L_o-R_o)\}$ and $R_p=0.5*\{(L_o+R_o)-EQ(L_o-R_o)\}$.

SRS has been widely licensed in both the consumer audio and multimedia markets as a standalone spatial enhancement module. The SRS technique is specifically designed for use in a postprocessor configuration, so ordinarily it is not possible for an audio producer to position individual sound effects in predetermined locations. As a stereo postprocessor this approach is appropriate, since the L and R input signals to the process are typically already a stereo mix, and the goal of the procedure is to enhance or exaggerate the spatial qualities of that mix.

3.3.2 Lowe and Lees Patent (QSound)

U.S. Patent #5,046,097 (filed 2 Sept. 1988, issued 3 Sept. 1991) [11] and subsequent patents #5,105,462 and #5,208,860 describe an invention in which one or more monophonic sound recordings are separately processed into pairs of left and right stereo signals, which are then additively combined into a two-channel stereo mix specifically for reproduction via two ordinary hi-fi loudspeakers. The processing consists of special empirically-determined amplitude and phase (delay) adjustments which produce the illusion of a highly localized sound source at a predetermined azimuth and elevation using a conventional stereo reproduction system. In this manner a plurality of individually localized sound images can be created and placed into motion by time variations in the amplitude and phase parameters. This scheme is summarized in Figure 5.

In the context of a single-ended preprocessing scheme the invention allows the audio producer to position sound effects or musical voices in virtual positions both between and outside the physical locations of the two loudspeakers. The procedure can also be used in a postprocessing configuration to broaden the stereo sound field by positioning the L and R images beyond the lateral positions of the speakers. Marketed by QSound Labs, Inc., of Calgary, Alberta, Canada, the various products based on this patent have been used for professional audio production, and more recently in multimedia computing.

3.3.3 Desper Patent (Spatializer)

U.S. Patent #5,412,731 (filed 9 Jan 1990, issued 2 May 1995) [9] describes a spatial enhancement technique that is most suitable for use as a postprocessor. The patent is assigned to Desper Products, Inc., a subsidiary of Spatializer Audio Labs, Inc. Spatializer Labs has licensed its products in both the consumer audio and multimedia computing markets.

Like the SRS patent described above, the principal embodiment of the Desper system is used to compute the sum (L+R) and difference (L-R) of the two-channel input. However, rather than equalizing the sum and difference signals, the Spatializer system is used to delay the L-R signal by a user specified amount, to filter and adjust the level of the delayed L-R signal (either manually or automatically), then to construct left and right output signals (L_o and R_o) by additively combining the unprocessed L+R and L-R signals and the processed L-R signal. The stated rationale for this processing technique is to adjust the relationship between the direct signal arrivals, assumed to be primarily in the scope of the mono signal (L+R), and the spatial or ambient signal arrivals, which are assumed to be primarily in the scope of the difference signal (L-R). An output detection circuit is provided to monitor the levels of $|L_o+R_o|$ and $|L_o-R_o|$

in order to adjust the gain applied to the processed L-R signal. The Desper invention is depicted in Figure 6.

Thus, unlike the SRS scheme described previously, the Desper (Spatializer) technique performs its processing action entirely on the L-R signal, not on the sum and difference signals separately.

3.3.4 Discussion

The three commercial single-ended spatial enhancement methods described above produce obvious and useful broadening of the stereo sound field for both music and sound effects. The Lowe and Lees (QSound) invention differs from the Klayman (SRS) and Desper (Spatializer) schemes by addressing the issue of specific localization of individual sound effects. In the multimedia PC environment the spatial enhancement procedures are typically implemented as a postprocessor module (analog circuitry or a programmed algorithm) inserted just prior to the output amplifier of the plug-in soundcard or audio hardware within the microcomputer. Work is presumably underway by these and other inventors to deal with the increasing need for both spatial enhancement and spatial localization in multimedia computing systems. In particular, Microsoft Corporation is currently formulating a standardized audio localization procedure in the Windows environment through the recently announced DirectSound3D application programming interface (API).

4. A New Single-Ended Stereo Broadening Scheme

Extending the wide range of artificial stereo image enhancement techniques in the literature, a new scheme involving an adaptive filter structure is proposed in this paper.

4.1 Description

In the proposed procedure two adaptive filters are used to process the left and right signals of the stereo pair. The "error" signal used to perform the adaptation is the difference between the filter outputs, $Y = H_L(L)$ and $Y = H_R(R)$, and the "desired" signal which is chosen to be the interchannel difference, L-R. By minimizing the error signal the filter outputs are steered toward the desired signal, or in other words, L and R are processed to become more like L-R. Thus, the processed left and right output signals are similar to the input signals, but with enhancement of the difference between the channels and therefore a perception of enhanced L, R dissimilarity.

The presumed advantage of the proposed adaptive system is a reduction in the presence of the alternate channel signal in the left and right outputs as would otherwise occur when simply adding L-R and R-L to the left and right input signals, respectively.

The most basic configuration of the proposed adaptive system is shown in Figure 7. The left and right input signals are passed through finite impulse response (FIR) adaptive filters H_L and H_R, respectively. In this basic configuration the filters are adapted separately via the widely known LMS algorithm using L-R as the desired response [18].

4.2 Discussion

An illustrative example of the proposed algorithm is depicted in Figure 8. The simple system used here involves 12 adaptive coefficients for each channel and a convergence time constant of approximately 50 ms. In this example the left channel consists of a 100 Hz sinusoid that is gated on and off, while the right channel contains the sum of an identical 100 Hz gated component and a continuous 1 kHz sinusoid: the difference signal, L-R, consists only of the 1 kHz component, as shown in Figure 9. Thus, the adaptive algorithm is expected to adjust H_L and H_R to attenuate the common 100 Hz component in both channels and to pass the 1 kHz component in the right channel. The raw system output is shown in Figure 10.

In a practical system the raw output could be mixed with a proportion of the input signal to avoid complete attenuation of any monophonic material. As mentioned above, the transient behavior of the adaptation process must also be carefully controlled to eliminate any audible artifacts as the left and right channel signals vary with time.

This adaptive configuration has been evaluated informally with a wide variety of two-channel musical input signals and several adaptive filter lengths and convergence rates. The results have been very encouraging. Still, the expected conflict between the desire for fast adaptation and the desire to avoid any timbral coloration or audible pumping is difficult to resolve with a computationally simple algorithm like LMS. With the careful tuning that is required for this or any other adaptive algorithm, the noticeable enhancement of interchannel differences achieved by this process can be a useful "front-end" for more sophisticated stereophonic image enhancement procedures.

5. Conclusion

In the years ahead it seems clear that the importance of spatial audio for multimedia entertainment and education on PC platforms will continue to grow due to the increasing awareness and interest of consumers in surround sound and multichannel reproduction. An initial step in this direction is already seen in the wide variety of personal computer products advertising support for "3D Sound" in some sense. The history of the field of stereophonic image enhancement and several notable commercial algorithms have been reviewed in this paper in order to put the impending multimedia computing developments into a broader perspective.

The adaptive enhancement algorithm proposed in this paper is an example of the digital processing that is now feasible in real time on programmable DSP chips and host processors. Thus, the range of solutions available for stereophonic image enhancement extends from discrete components or special-purpose analog integrated circuits to programmable DSP chips or general-purpose microprocessors.

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[8] Klayman, A.I., "Stereo Enhancement System," U.S. Patent #4,748,669, issued May 31, 1988.

[9] Desper, S.W., "Automatic Stereophonic Manipulation System and Apparatus for Image Enhancement," U.S. Patent #5,412,731, issued May 2, 1995.

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[18] Lim, J.S., and Oppenheim, A.V., eds., *Advanced Topics in Signal Processing,* Englewood Cliffs, NJ: Prentice Hall, 1988.

7. Other Information

Information on several of the commercial products mentioned in this paper is available via the World Wide Web at the following URLs:

Dolby Laboratories, Inc.
`http://www.dolby.com`

Spatializer Audio Labs, Inc.
`http://www.spatializer.com`

SRS Labs, Inc.
`http://www.srslabs.com`

QSound Labs, Inc.
`http://www.qsound.com`

Crystal River Engineering
`http://www.cre.com`

U.S. Patent and Trademark Office
`http://www.uspto.gov`

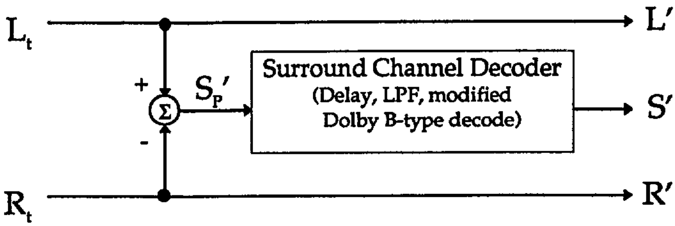

Figure 1: Dolby Passive Surround Decoder Concept

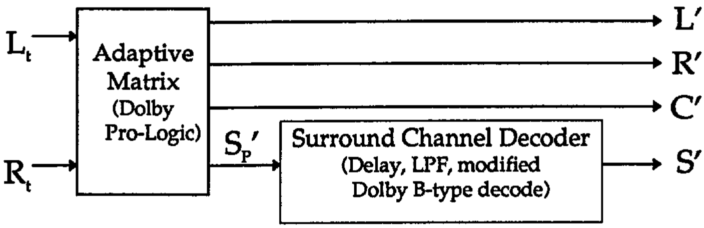

Figure 2: Dolby Pro-Logic (Active) Surround Decoder Concept

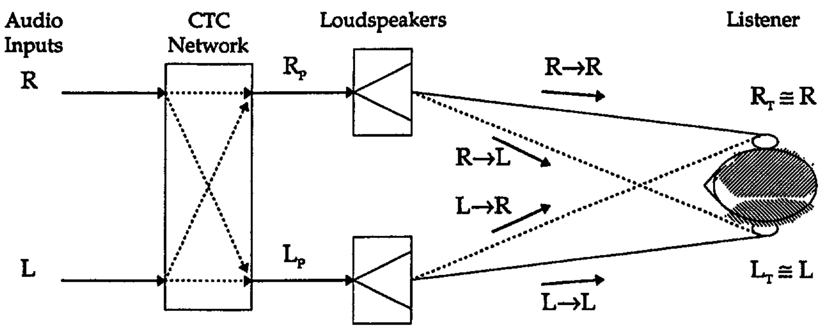

Figure 3: Crosstalk Compensation (CTC) Concept

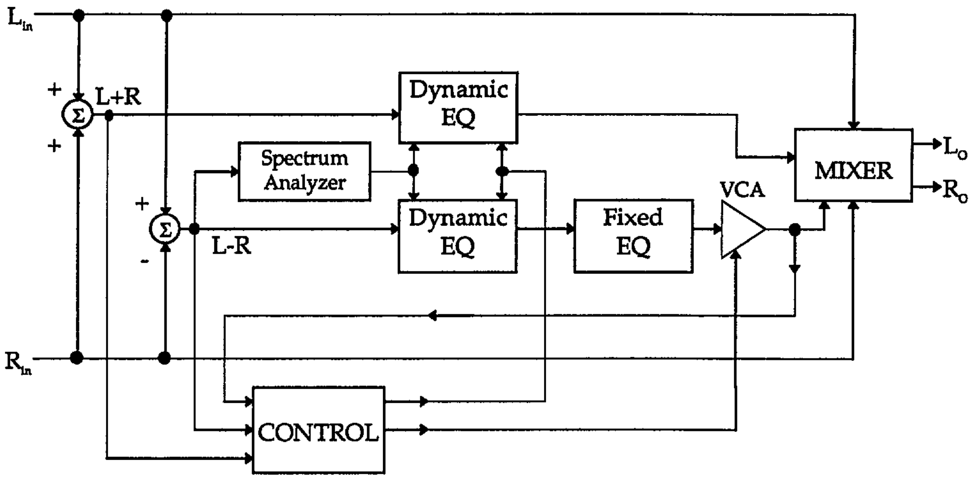

Figure 4: Klayman (SRS) Spatial Enhancement Concept
(based on Fig. 2 of Ref. [8])

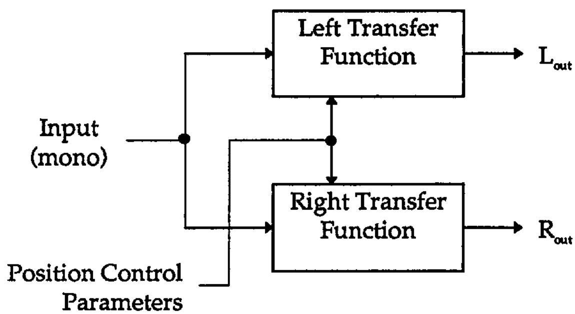

Figure 5: Lowe and Lees (QSound) Spatial Enhancement Concept
(based on Figs. 16 and 17 of Ref. [11])

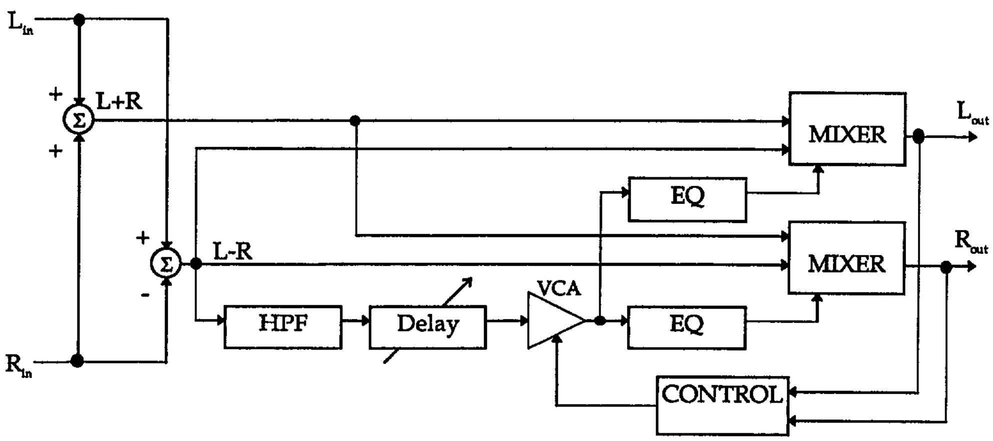

Figure 6: Desper (Spatializer) Spatial Enhancement Concept
(based on Fig. 5 of Ref. [9])

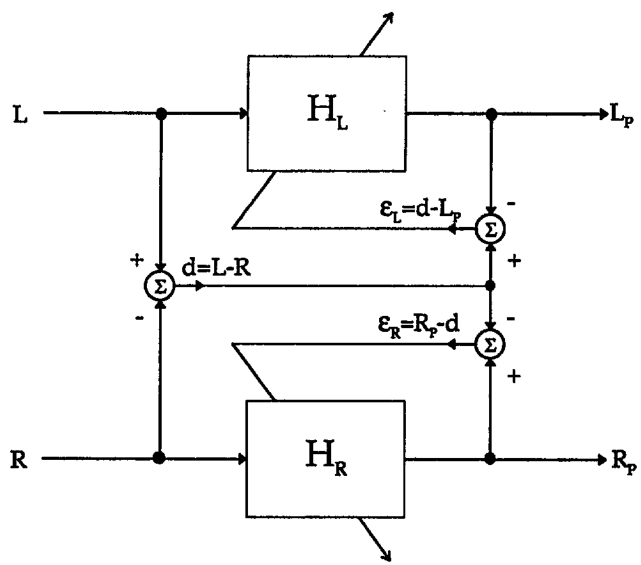

Figure 7. Basic Configuration of Proposed Adaptive Enhancement Procedure

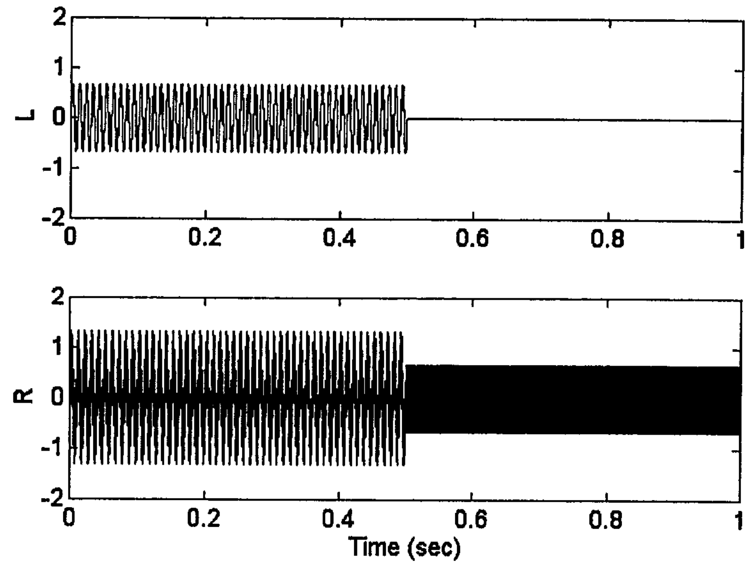

Figure 8: Left and Right Test Signals
(L: gated 100 Hz burst; R: 100 Hz burst + 1 kHz continuous)

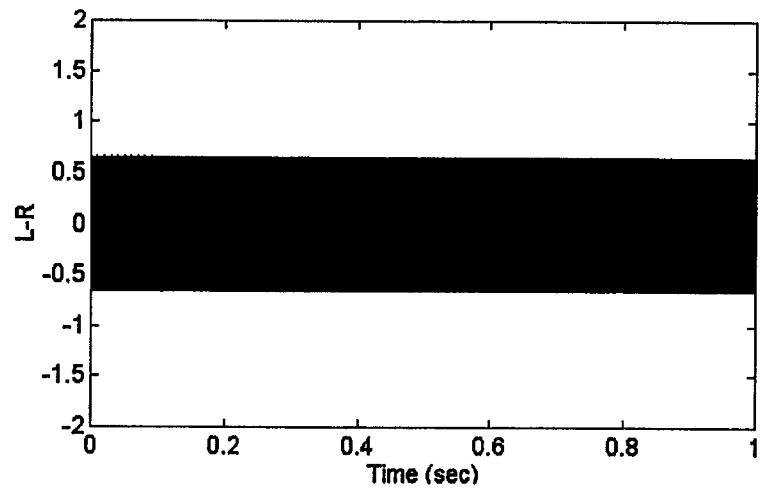

Figure 9: L-R (difference) Signal for Test Signals of Figure 8

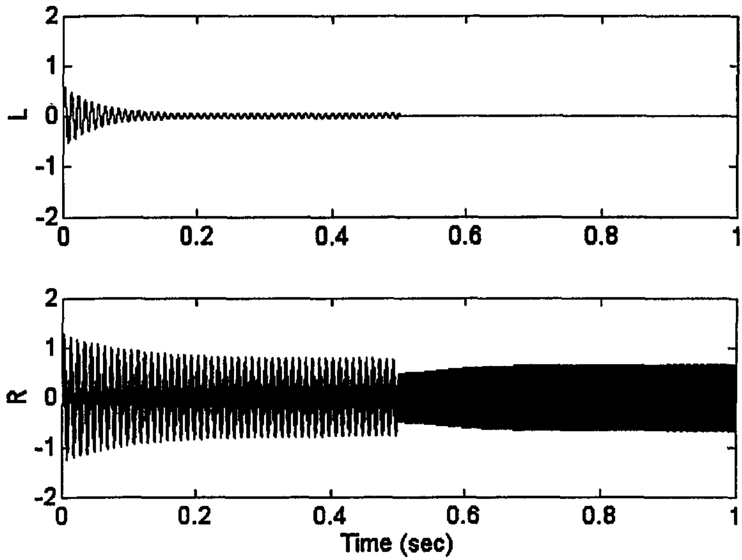

Figure 10: Output Signals from Adaptive Scheme Applied to Test Signals of Figure 8