

Single Bipolar Loudspeaker System for Stereo Reproduction

Fredrik Gunnarsson

CTO Embracing Sound Experience AB

Abstract:

The conventional stereo speaker system will be explained also with variations common in Consumer Electronics. Similarities but also distinct differences between the Single Bipolar Loudspeaker System and the conventional stereo system are shown. Theory as well as the embodiment of the Single Bipolar Loudspeaker System is put forward. The Single Bipolar Loudspeaker System is compared to a conventional system regarding the localization cues.

1. Introduction

This paper will show the benefits of the Single Bipolar Loudspeaker System compared to a conventional stereo system. The Single Bipolar Loudspeaker System set out in this paper is used by professionals in their daily work as a critical reference monitor system due to its accurate reproduction of the stereo image that translates well into any conventional listening situation at the consumer.

The stereo imaging of the Single Bipolar Loudspeaker System is of a quality previously only achieved in high-end speaker system. This imaging is maintained in ordinary consumer level products, even in very small devices such as mobile phone terminals etc. Everywhere it is implemented it will maintain both the tonality of the speaker drivers and provides true reproduction of the recording as it is reproducing the localization cues from the recording optimized to the present understanding of human localization of sound.

To give the reader a more complete picture of the background to the Single Bipolar Loudspeaker System, the conventional stereo speaker system will be explained. Similarities will be shown but also distinct differences between the systems.

2. Defining the Stereo Signal

The meaning of the word Stereo comes from the Greek word *stereos* meaning solid. A solid is implying the three dimensions x , y , z . (Ref.1) The word stereo is usually given the meaning *multidimensional*. One reason may be that the actual stereo system with two mono speakers does not have the ability to convey all three dimensions. The conventional stereo system is in the general case describing width (x - axis), and in optimal cases also depth (y - axis).

The common interpretation of the word stereo is connected to the two channels used that obviously differentiates stereo from mono. This definition is correct but should not be used in a limiting sense. Any number of channels more than one can be defined as stereophonic as long as its purpose is to provide a multidimensional experience to the listener. The definition of stereo is in

this paper chosen to be limited to the two channel incarnation that's without doubt most widely spread.

3. Visualizing the Stereo Signal

To aid the understanding of the stereo signal, images of a vectorscope will be used. Figure 1 show a stereo signal presented on a vectorscope. This signal contains both Inter Channel Level Differences (ICLD), and Inter Channel Time Differences (ICTD). The left and right channels are the carrier of the stereo signal.

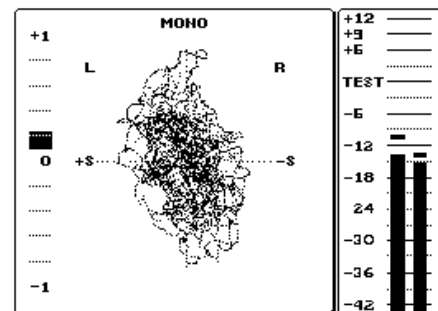


Fig.1 A by definition good stereo signal displayed on a vectorscope.

Figure 2a shows the stereophonic form of mono signal that constitutes of two separate channels being equal in amplitude and coincident in time/phase. This information can be carried in a single channel (*Sum* channel (S)).

If the said mono information is removed from a stereo signal such as shown in Figure 1, the remaining signals are identical in content and equal in amplitude but of opposite polarity relative to each other (Fig. 2b). This is the content of the stereo channels that differentiate stereo from mono. This information is the directional information that describes the lateral plane (x - axis). As the remaining signal in the two channels is identical, except for the polarity inversion, it is possible to use the signal in the Left channel only as the description of directional information (*Difference* channel (D)). The resultant S and D channels can be matrixed together to bring the stereo signal back in to its Left and Right form.

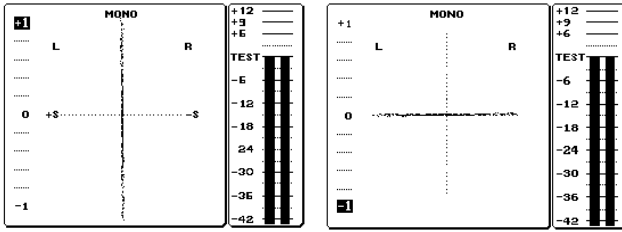


Fig.2a

Fig.2b

Fig.2a shows the Sum information of a stereo signal. Fig.2b shows the Difference information of the stereo signal.

This process adds gain to the signal that has to be taken care of every second time the linear transformation is performed. The relation is as follows;

$$\begin{aligned}
 L+R &= S \\
 L-R &= D \\
 (S+D)/2 &= L \\
 (S-D)/2 &= R
 \end{aligned}$$

4. The scope of stereophonic reproduction

The scope of stereophonic reproduction is to reproduce directional information of the recording venue to the listener's hearing sense.

A theoretically optimal way of fulfilling this scope of stereophonic reproduction is the binaural recording and reproduction method. This includes the utilization of a dummy head in the recording position and the use of headphones for the listener. It is then possible to perceive the dimensions x, y, z, of the recording venue. In spite of these theoretically correct preferences of the binaural reproduction method, there are practical aspects where this method comes to short.

The main problem is the need of similar head, torso, and outer ear shape of the artificial head in the recording position compared to the head of the actual listener. If these preferences are not fulfilled the reproduction will be altered so that the desired positions of localization will not be perceived correctly by the listener.

There are also attempts of binaural reproduction that utilizes loudspeakers to convey the binaural signal to the listener. This reproduction is limited by the lacking separation of the headphones. This can be compensated for by signal treatment (Ref. 2, 3) or a physical wall (Ref. 4). The signal treatment adds information in the time domain to cancel out the sound from the opposite speaker. The resultant signal is then more complex for the actual speaker driver to reproduce which degrades the sound quality. The wall solution does not have any of these problems but may be considered to be impractical. The binaural methods seems to provide steady localization in two dimensions x and y. The z - axis is the most difficult to realize as this depends most on the shape of the outer ear, something that is unique for every listener.

In recent time the recording venue is not as easy to define as there may be several venues that together form the art work of the recording. Some audio content is further more electronically created which simply makes the scope of

stereophonic reproduction set out above not to be applicable. This calls for a better definition of stereophonic reproduction that also can include a simulated stereo signal:

The scope of stereophonic reproduction is to reproduce the electrical inter-channel relationship, all information present in the recording, to the listeners hearing sense.

Even though a binaural approach is possible to utilize with the means of 3-D positioning systems in DSP it would be desired, out of a professional user perspective, that such a stereo system have similar functionality to how conventional stereo have worked in all times to facilitate the working procedure.

5. Definition of conventional stereo speaker systems

Stereophonic reproduction is implying at least two mono loudspeakers that can communicate with the listener's both ears. The trigonometric layout is defined to be an equilateral triangle where the centre of the listener's head is positioned at the apex and the two loudspeakers are positioned at the points at the base of the triangle, see Figure 3.

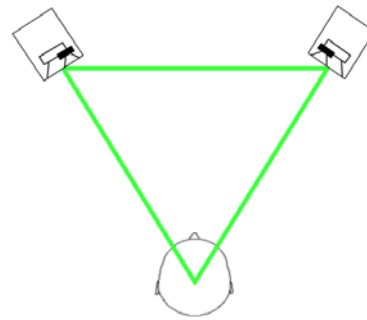


Fig.3 Trigonometric layout of conventional stereo.

6. Localization Cues in Conventional Stereo

The inter channel differences of the stereo signal mentioned above results in three different inter aural differences to the ears of the listener: *Interaural Level difference (ILD)*, *Interaural Phase difference (IPD)* and *Intraural Time difference (ITD)*

Perception of ICLD

The result of ICLD to the listener's ears is primarily ILD for short wavelengths. The ILD achieved by one of the two mono speakers playing defines the separation of the stereo channels to the listener's ears. Channel separation implies that the right ear should hear a bare minimum of what the left loudspeaker is reproducing and vice versa. This separation is the paramount of any stereo reproduction system as it sets the standard of how well the inter channel information is conveyed most logically from ICLD but also from ICTD. Simple measurements are used to show the source of the ILD. If measurement microphones are positioned at the points of the listener's ears (Fig. 4a, 4b), without a head as a sound shadowing barrier in-between, the measurement of one speaker positioned 30° to the left or right of the median plane of

the listener will show level difference of only 1–2 dB to the ears of the listener under pseudo anechoic conditions. With room reflections this difference is close to zero.

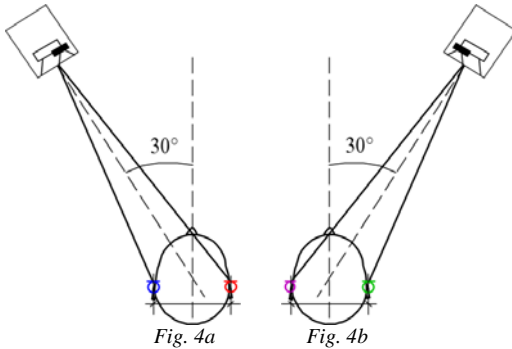


Fig. 4 Setup for measurements in Fig 5.

If a listener's head is positioned between the measurement points (Fig 4a, 4b), ILD is approximately 3 - 5 dB up to 1 kHz and is up to 10 dB above 1 kHz under pseudo anechoic conditions (Fig. 5). ILD is slightly reduced if room reflections are included in the measurements. ILD is improved if the listener is wearing spectacles as the energy that is to be diffracted around the head is reflected by the hard surfaces. The unsymmetrical ILD is the reference for this unique listener and is therefore correctly resolved.

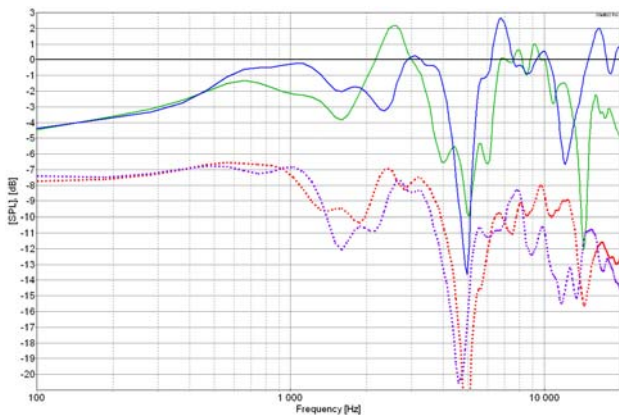


Fig. 5 Relative measurements to the trigonometric correct point at half the ear distance Black line. The corresponding colors of the plots are according to the colors of the microphones in Fig 4.

In the work of A. D. Blumlein (Ref. 5) it is stated that level difference between the two loudspeakers are reproduced as IPD to the listener's ears in the frequency range up to 700 Hz. This IPD is what results to the listener's ears from ICLD and does not reflect the Inter Channel Phase Difference (ICPD) (treated under *ICTD* defined below). This phase difference is due to the ITD that the two loudspeakers discrete positions introduce, as the amplitude between the two loudspeakers varies the IPD shifts accordingly. To put the definition straight it must be understood that this is a time shift that is independent of wavelength but that is small enough to fit inside the first part of the phase cycle of the reproduced wavelengths and thereby get its name. Through the work of Clark et. al. (Ref. 6) the IPD was termed ITD which is a more proper description. Clarks' work shows that the

ITD resulting to the listeners ears from ICLD below 700 Hz has the following relation to the perceived angle of a virtual sound source (Eq. 1):

$$[(L-R)/(L+R)]\sin \theta_0 = \sin \alpha \quad (1)$$

where,

α is the angle of the perceived virtual image,

$L-R$ is the directional information,

$L+R$ is the monoral information,

θ_0 is half the angle separating the loudspeakers (Fig. 7).

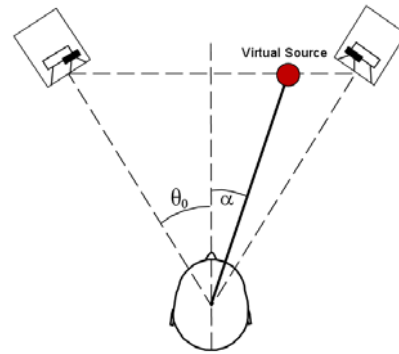


Fig. 7 The relation of the angle of the virtual source and the separating speaker angle in a conventional stereo system.

This equation shows that phase difference will not exist when $\theta_0 = \alpha$. The IPD present to the listeners ears at this point is the result of the position of a single speaker which is correct. However, the position where $\alpha = 0$ also shows that no IPD is present to the listeners ears. This must be questioned as identical IPD is present to the listeners ears. This symmetrical IPD introduces localization cues that adds proximity and width to the virtual source localized in front of the listener. The above described time dependant delay is alone at work as long as $L+R \geq L-R$. If $L-R < L+R$ the $L-R$ signal is also altered by a frequency dependent delay as referred to in the work of Sandel (Ref. 7). This frequency dependant delay has been found by the author to be approximately 90° independent of wavelength. A 90° offset between the $L+R$ information and the $L-R$ information results in a two way conversion of inter-channel relations. ICLD are converted into IPD and ICPD is converted to ILD. The most useful conversion is that from ICLD to IPD. This result in corresponding phase angles for all wavelengths but with the addition of the time dependant delay mentioned above. This 90° frequency dependant delay of the $L-R$ signal is what makes it possible to perceive an expanded sound stage outside the boundaries sidewise of the two speakers. This function is present up to the wavelength that equals the listeners separating ear distance (Fig. 8). Above this point IPD breaks down into chaos due to the trigonometric layout of the conventional stereo system. This out of speaker boundary localization can be better utilized the closer together the two speakers are positioned as seen in the work of Bauer (Ref. 8), even though equation 1 still applies. IPD improves as the ka ratio increases from the speaker drivers used. The larger sending surface the better defined IPD and the lower in frequency it is possible to be perceived. This favors the use of multi driver solutions

and is a possible explanation to the extreme stereo imaging that is possible to get from electrostatic speakers among other large surface piston related loudspeaker drivers.

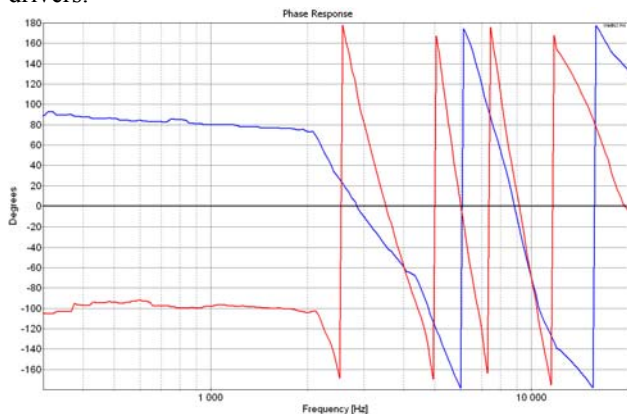


Fig. 8 The plot is showing the typical resultant acoustic phase relation of the L-R signal compared to L+R from a conventional stereo system. The test signal is inverted to the right speaker to create the L-R signal. The blue line is measured at half ear distance to the left of apex and is electrically in phase with the L+R signal (black) measured on axis of the speaker system (used as reference for this plot). The red line is the result of the inverted signal and is measured at half ear distance to the right from apex.

Perception of ICTD

The sensitivity in the time domain of the binaural hearing mechanism is taken out of order by the conventional stereo system. This is due to the previously mentioned ITD that the separate Left Right loudspeaker positions provide. The typical ITD for one speaker only is approximately 0,2ms. This is the natural ITD for a sound source 30° offset from the median plane of the listener. As seen in Figure 5 there are differences in how the channels are separated to the left and right side due to the lack of symmetry of a human head. The conventional stereo system don not rely on any natural detection of ITD but of the excess ITD from the speaker system (Ref.10) The work of the author has shown that this lack of symmetry results in a listener unique weighting of the localization angle depending on frequency.

Figure 9 is showing the ICTD generally needed to obtain various localization positions from a conventional +-30° stereo system. For band limited noise ITD is perceived corresponding to the channel separation obtained. When the channel separation is less, more ICTD is needed to obtain the desired localization angle. ICTD of long wavelengths is lost to our hearing sense as seen in the work of Bernfeld (Ref. 10). Wavelengths that approach a 180° relation between the speakers will result in polarity inverted ILD when altered by the aforementioned frequency dependent time delay. This results in localization to what seems to be random positions outside the boundaries of the speaker pair. This effect is sprung out of the conversion of inter channel differences from the frequency dependant time delay mentioned above. The summed impression of ICTD is a huge description of an acoustic event that is sprung out of the preferences of the reproduction system itself. The subjective effect is that the loudspeakers seem to disappear as the localization

cues are in a chaotic mixture and a vast number of ILD cues are pointing to positions outside the boundaries of the speaker pair.

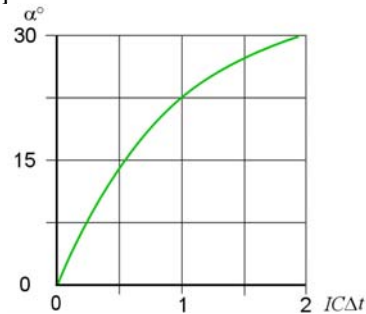


Fig. 9 Subjectively determined dependence of the localization angle [α] in relation to the Inter Channel time difference [IC Δ t] in ms for two channel replay. (Ref. 9).

7. Variations of conventional stereo

The conventional stereo system has been given many shapes over the years. Most commonly, loudspeaker designers that have little or no experience of the recording process, more specifically no experience of the recording venue of their reference recordings. With this lack of knowledge variations of the conventional stereo system has been invented that on its own create spatial experiences but with limited fidelity in comparison to the actual source. The conventional stereo system has further been implemented in products that by its normal intended use never invite the stereo speaker system to actually function properly as the Left and Right speakers cannot be placed far apart enough in the products form factor. The most important aspect of the conventional stereo speaker system is the angular relation. Any modification of this relation will result in alteration of imaging.

The most common in consumer electronic products is that the base of the stereo triangle is too narrow. If separate speakers are used it is up to the interest of the listener to position them correctly.

One approach to the conventional stereo system, even used in high end HiFi systems, is to use loudspeakers that are mainly defined as pressure points and minimally by directivity. The L-R signal is then vaguely separated from L+R and is reproduced from within the speaker boundaries. This single dimension that lacks localization cues outside the speaker positions is possible to perceive in a wide area but fails to fully reproduce the stereo signal. In order to perceive any noticeable depth to the image a proper triangular relation is needed.

8. Conclusions of conventional stereo

The conventional stereo system has two reproduction points that are totally impeccable, namely the position of the left and right loudspeaker. Any other position result in more complex localization cue to the listener's ears compared to the original inter-channel relation of the source signal.

One explanation of the many diverting opinions of what good stereo really is may be traced back to the fact that the perceived level difference is the result of the shape of the listener's unique head and its acoustic shadowing preferences. The level difference achieved by the head is

furthermore diminished by room reflections and is dependant on attributes like hats, spectacles, etc.

A high-end conventional stereo speaker system provides the best possible preferences for stereo to be perceived. However, the implications put forward above in regard of the added time difference of the speaker positions can *never* be overcome, only made less obvious.

The reproduction of sound from two speakers demands certain directivity of the speakers to convey all inter-channel differences as inter aural differences. This is given that we allow the stereo signal to be defined as both $L + R$ and $L - R$. Any piston related loudspeaker driver can be used for this purpose, for example, electrostatic, ribbon, magneplanar, domes etc. Distributed Mode Loudspeakers (DML) or the like show only pressure and no directivity and are therefore not possible to use if more than a single dimension (x - axis) is desired.

In the work of A.D. Blumlein (Ref. 5) the conventional stereo system is based on the duplex theorem put forward by Lord Rayleigh (Ref. 11) that states that the hearing sense uses ILD as the main localization cue for short wavelengths and IPD as the main localization cue for wavelengths below 700 Hz. In the work of Clark et. al. (Ref. 6) IPD was thought to be ITD based on the Jeffress model (Ref. 12) that states that localization is resolved in the time domain by a neural coincident network where different length of the neural paths from the two ears is used to determine ITD in the Olivary Complex (SOC). The model implies that time differences larger than the head size is not resolved.

This is all logically good assumptions but is not up to date with current research and understanding of the hearing sense. The weighting of the different cues of the duplex theorem is valid but does not set any limits in the capabilities of the binaural hearing sense. The Jeffress model has been shown not to fit the actual physiological system in the mammal brain (humans included) according to contemporary knowledge of the mammal physiology (Ref. 16, 17).

9. Introducing the Single Bipolar Loudspeaker System

This Single Bipolar Loudspeaker System for stereo reproduction is designed to overcome the limitations of conventional stereo and also better utilize the human perception of sound.

The entire concept of Binaural Masking Level Difference (BMLD), as first described by Hirsh (Ref. 13) and Licklider (Ref. 14), shows that the human perception is of a summing and differencing nature. In the experimental work of Koehnke et. al. (Ref. 15) the BMLD clearly shows that individuals can determine phase relation up to 4 kHz. It may be concluded from this that the time domain is not entirely limited to the region below 1,5 kHz where previous the upper limit of IPD has been understood to be.

In the work of Fitzpatrick et. al. (Ref. 16) it is explained how time resolution for mammals rely on there types of neurons namely the peak type, through type, and the intermediate type, all sensitive to ITD near the endpoint of ITD processing, in the primary auditory cortex. This

research shows that mammals are sensitive to far greater time differences than previously assumed and also that shorter wavelengths are resolved by the locking to the envelope of sound. In the work of D. McAlpine (Ref. 17) it is further shown that the sensitivity of the mammal brain for time differences is resolved out of the phase locking to the phase angle in $+ - 180^\circ$.

With this background it is easy to conclude that a new approach to sound reproduction is needed, an approach that does not alter the envelopes of the source signal and that reproduces ICTD for the human auditory sense to function with maximum resolution.

If these terms are met this will result in a stereo system that better match the auditory perception of man and can rely on IPD that have a capability to reproduce a sound image up to $+90^\circ$ in front of the listener.

10. Theory behind Single Bipolar Stereo Reproduction

In order to avoid the introduction of time differences of the loudspeaker system itself and by that maintain the integrity in the time domain, the speaker system must reproduce the sound image from a single point that is aligned with the median plane of the listener.

It is not until this is accomplished that any other positions but the loudspeaker position can be truthfully reproduced as the speaker positions by them self do not result in any inter aural differences.

Based on the integrity of the time domain, there are major acoustic advantages. The interference pattern from two or more separate speakers of a conventional stereo system is complex. For a conventional stereo system the listening space have to be treated with great care in order to provide the best possible frequency response over time to the listener's ears.

By using a single source these problems are eliminated as the spectral content is maintained in the direct sound from the speaker system. The single unit approach also fits the means to reproduce the L-R and L+R content of the stereo signal to the ears of the listener. The L+R signal will form a monopole dispersion pattern (Fig.10a) and the L-R signal will form a dipole dispersion pattern (Fig. 10b) where separate lobes reach each ear of the listener.

It is generally known that the expression for obtaining a dipole dispersion pattern from two monopoles in anti phase is:

$$kd \ll 1 \quad (2)$$

where,

k is the wavenumber derived from $2\pi/\lambda$ where λ is the wavelength and,

d is the separating distance of two monopoles in anti phase.

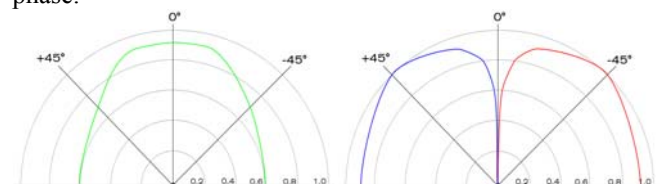


Fig. 10a

Fig. 10b

Dispersion pattern of a Single Bipolar Loudspeaker System Monopole Fig. 10a and dipole Fig. 10b @ 1 kHz.

This relation is definitely true if the dipole is constituted of two perfect monopoles in anti phase. However, if loudspeakers drivers are used the directivity of the separate speaker drivers must be considered. The relation that defines the efficiency of the piston links the size of the piston to the wave number reproduced:

$$ka \quad (3)$$

k is the wavenumber as previously defined and, a is the radius of the speaker driver.

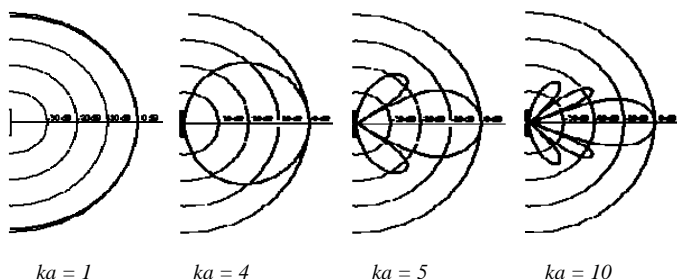


Fig. 11 Dispersion patterns for a selection of ka values.

This is describing the gradually increasing directivity as the ka value rises, see Figure 11. When the dipole is constituted by speaker drivers from which ka values can be derived, the $kd \ll 1$ relation is far from true. This is given that the speaker drivers are positioned on the same baffle to be able to work in parallel side by side of each other (Fig. 12). The relation between kd and ka is fixed throughout the spectra. This means that even if the kd value is increasing the directivity of the speaker drivers defined in the ka value makes the speaker pair to still constitute a dipole dispersion pattern. This is true up to a point where the dipole gradually breaks down due to the separating distance being too large compared to the dispersion pattern of the separate drivers. At this point the dipole action is gradually transformed into a bipolar action with polarity inverted poles as no cancellation of energy is taking place.

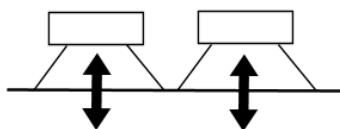


Fig. 12 Visualizing the parallel placement of loudspeaker drivers that is capable of both L+R and L-R reproduction.

The dipole is effective on axis from the speaker system though limited in the horizontal plane due to the directivity of the speaker drivers. The work of the author has shown that the following ratio is desired:

$$kd / ka \geq 2 < 3 \quad (4)$$

The difference in phase between the monopole and the dipole increases with frequency if the two drivers are contained within the same cavity. If the net pressure flow of the dipole is reduced by sufficiently separating the drivers on the back of the baffle, the phase relation

between the monopole and the dipole will reach approximately 90° in the entire spectra. This is identical to the frequency dependant time delay found in the conventional stereo system up to 2 kHz.

Two purposes must be served by signal treatment in order to make the Single Bipolar Loudspeaker System to function; to compensate for the less output power of the dipole in relation to the monopole and, to compensate for the frequency dependant time delay introduced by the speaker system.

11. The Single Bipolar Stereo Loudspeaker System Defined

The Single Bipolar Stereo Loudspeaker System has a number of theoretical similarities on the microphone side with the well known M/S stereo microphone principle. In the work of Woodbury (Ref. 18) and Fixler (Ref. 19), among many others, it is seen that the L-R signal is reproduced by separate speaker drivers that are angled to the sides of the speaker unit. The purpose is to disperse the L-R signal to the side of the speaker in order to exactly invert the working principle of an M/S microphone. This results in a speaker system which only shows a defined near field for the L+R information and has to rely on reflexes from the room boundaries in order for the L-R information to reach the listener's ears. The proposed Single Bipolar Stereo Loudspeaker System is instead targeting the ears of the listener. This makes the system less room dependent.

The Single Bipolar Stereo Speaker System comprises (see Fig 13);

Speaker system, including;

- Positioning of the speaker drivers,
- Separation of the drivers on the back of the baffle,
- A diffracting plate element that minimally protrudes in between the speaker drivers on the front of the baffle.

Stereo signal adaptation;

- Performed in the analogue domain with discrete components or in the digital domain by DSP.

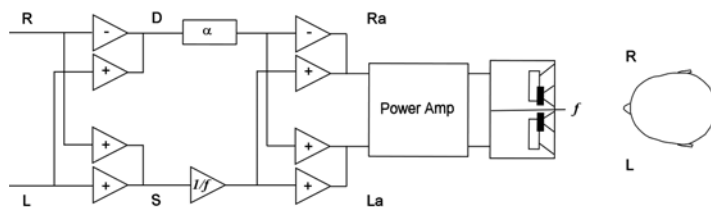


Fig. 13 Overview of the Single Bipolar Loudspeaker System.

12. The sound field defined

The Single Bipolar Loudspeaker System has a near field and a far field. These parts of the sound field are defined by two factors: (a) the effective distance of the dipole dispersion pattern relative to the monopole (Fig. 10b), and (b) the relation of the room boundaries to the listener and speaker system. The effective distance of the dipole is

dependant on the speaker driver size and is found by the following equation:

$$n\pi/2 = S_l \quad (5)$$

where,

n is the ka value at 1,5kHz for the largest driver.

S_l is the maximum listening distance where still the sides beyond +/- 30 degrees is perceived from the direct sound field.

The work of the author has shown that this relation also includes drivers that are crossed over at a much lower frequency than 1,5kHz as the larger surface added by these drivers contributes to the dipole action also for shorter wavelengths. The second factor is the relation of the speaker and listener to the room boundaries (Fig. 14). This is defined as follows:

$$SL \geq RL \quad (6)$$

where,

SL is the Speaker to Lister distance, and

RL is the distance from listener to the nearest room boundary.

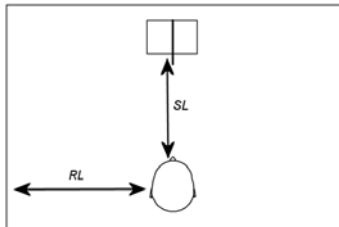


Fig.14 Speaker to listener distance to room boundaries.

13. Description of the speaker system

The speaker system consists of two identical speakers or sets of speakers, positioned in immediate proximity, mirrored to a central line Fig. 15.

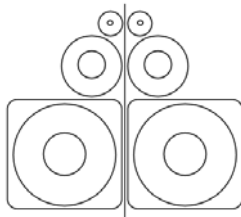


Fig. 15 Example of positioning of the speaker drivers.

The loudspeaker driver type used must show directivity to be used. The drivers are positioned side by side without displacement in any plane. This is done in order for the drivers to work together in the same direction when an identical signal is applied to the speaker drivers (L+R). When this is achieved, the optimal preference for a polarity inverted signal (L-R) is also obtained.

The aforementioned separation of speakers on the back of the baffle depends on the air tightness of the entire housing of the speaker unit. The author has found that the separation between the drivers on the back has to provide identical or better acoustic isolation than the entire enclosure. This is important to understand when the speaker system is implemented in i.e. Consumer

Electronics where air tightness varies due to production tolerances.

When this relation is achieved the L-R signal exhibits frequency dependent phase difference of approximately 90° in the entire spectra (Fig 16), This is identical to what the conventional stereo system is capable of up to around 2 kHz.

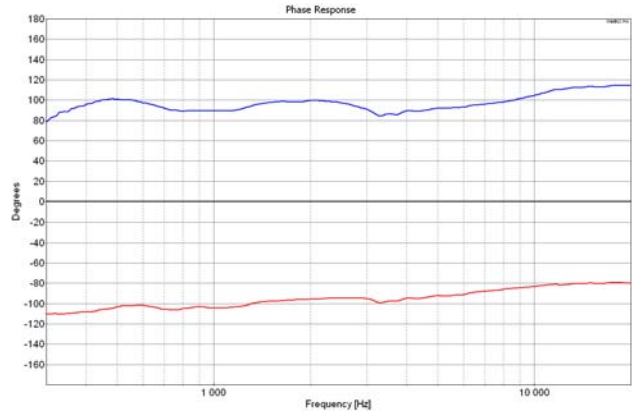


Fig. 16 the resultant acoustic phase difference between L-R and L+R measured at ear distance. L+R is used as a reference for this phase plot

As the two speaker drivers still do not form a perfect monopole source due to their size it is necessary to implement a protruding plate element between the speaker drivers. This plate element (Fig.17) acts as a secondary transducer by diffraction. A direction to the height over the baffle of the protruding plate element is defined to be equal to or shorter than one quarter of the speaker driver diameter that it is protruding between.

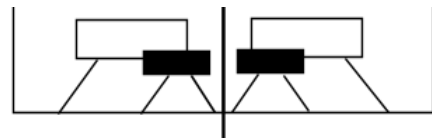


Fig. 17 The plate element shown in its position.

This maximum height is set in order to minimize the effect of reflection of the plate element. Its shape is however not of critical importance.

14. Description of the signal adaptation

The signal adaptation is performed on the line level signal or alternatively in DSP. There are several ways to implement the circuit depending on the actual application. The effectiveness of the dipole relative to the monopole differs from each other by nature. The two signals need to be compensated in gain to acoustically restore the relations of the stereo signal. There are two different approaches to this: to equal the level relation left, mono, right, or, to equal the level relation of L+R, and L-R.

Another object of the signal adaptation is to *correct* the introduced frequency dependant time shift of the L-R component from the speaker system. The reason for this need of correction is that the sensitivity of IPD in the auditory sense decreases as the wavelength gets shorter.

15. Localization cues in the single bipolar stereo system

The Single Bipolar Loudspeaker System basically relies on the same equation (1) as the conventional stereo system. The major difference is that the Single Bipolar Loudspeaker System does not define any angular relation of separate loudspeaker positions as the speaker drivers of the two channels are defined to be as close together as possible, yet still side by side of each other. The following equation is valid for the single bipolar stereo system and is valid up to approximately 4 kHz, namely:

$$(1-f)(L-R)/(L+R) = \sin \alpha \quad (7)$$

where,

f is an attenuation factor allowed to be $0 > f < 1$.

α is the azimuth angle of the virtual sound source, and

L and R , are the signals applied to the left and right system input respectively see Fig. 18.

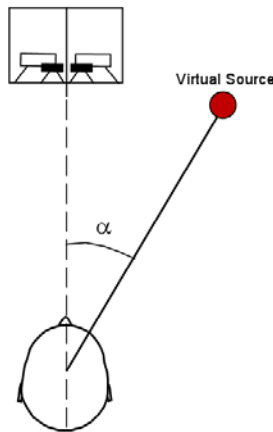


Fig.18 The angular relation to the median plane of the listener and the virtual sound source.

When understanding the difference in efficiency between the monopole and dipole it is clear that this equation shows that if f equals 1 the localization will be dominated by the $L+R$ information. If f equals 0.5 the reproduction system will show an angular relation equal to that of a ± 30 degree speaker setup but without the time differences introduced from any discrete speakers.

The Left and Right signal is of a constantly varying kind and can thereby result in that $L-R$ holds higher amplitude than $L+R$ either momentarily or statically. If $L+R$ equals zero Eq.6 shows the division by zero that will occur. No phantom source is possible to localize to any discrete angle for this signal state. The $L-R$ information is commonly localized to an undefined position to the left and right side of the listeners head up to $\pm 120^\circ$. The Single Bipolar Loudspeaker System corresponds theoretically very well with the function of the stereo microphone configuration consisting of two crossed figure of eight microphones. The equation that follows are describing the working principle of the microphone configuration.

$$[L-R]/[L+R] = \tan \theta \quad (8)$$

where,

θ is the angle of the real sound source to the microphone system (see Fig. 19), and

L and R are the resultant electrical signals from the microphones.

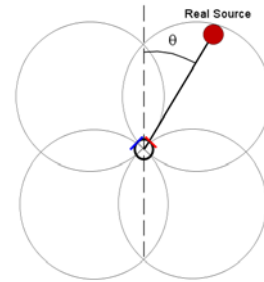


Fig.19 The angular relation of a real sound source to a pair of crossed figure of eight microphones.

It is clear by comparing equation 7 and 8 that certain angular distortion exists between recording and reproduction. This is due to the reproduction systems "stereophonic law of sines" (Ref. 8) whereas the microphone system is more linearly increasing its pickup angle up to $\pm 45^\circ$. It must also be remembered that an actual signal from a microphone system described above is far from perfect in that the transducers on the microphone side also has directivity flaws as the wavelength gets shorter.

After this general description of the localization function from a Single Bipolar Loudspeaker System, a detailed description will now follow, investigating the result of different inter-channel relations of the stereo signal to their results to the hearing sense of the listener. Initially the near field will be described but later also the far field.

ICLD

If ICLD is reproduced IPD will result due to the aforementioned frequency dependant time shift. This IPD is resolved by our hearing system and results in a close to $\pm 90^\circ$ imaging according to equation 7. This is given that ICLD is defined to be a coincident signal that can vary in amplitude but also alter in polarity, (a sharp line at varying angle between 0 and $\pm 90^\circ$ on a vectorscope). IPD is the basic localization cue from the Single Bipolar Loudspeaker System as ICLD is a very common kind of inter-channel difference. The up to $\pm 90^\circ$ imaging is valid up to 1.5 kHz. The $L-R$ information is still resolved by the listener up to approximately 4-5 kHz but the phase relation between $L+R$ and $L-R$ is not resolved and results in poor resolution around $\pm 30^\circ$. Above 5 kHz IPD is not perceived by most listeners. By compensating aforementioned frequency dependant time delay of the wavelengths from 1 kHz and above, ILD is restored to the listener's ears. The measured response is shown in figure 20. The ILD reproduced from a Single Bipolar Loudspeaker System does not depend on the acoustic features of the listeners head.

If ILD is desired to be reproduced in the entire spectra the frequency dependant time delay needs to be compensated for. The outcome after such compensation is that ILD will

resemble the outcome of IPD for any angle. This means that equation 7 is still valid.

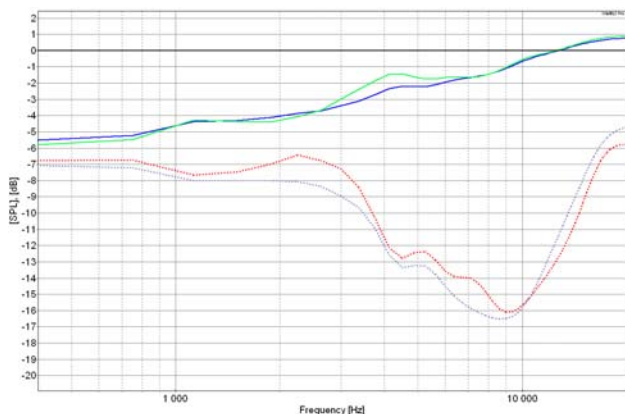


Fig. 20 Typical Level difference to the ears of the listener from a Single Bipolar Loudspeaker System, the colors of the plotted lines correspond to the microphone colors in Fig 4.

Above 1,5kHz ILD improves over IPD in that from this point ILD also provides good localization at $\pm 30^\circ$. In the frequency range from 5 kHz and above the L-R point is not resolved and ILD is limited in this part of the spectra up to $\pm 30^\circ$. The corresponding reproduction of IPD and ILD is due to the two least common denominators of the two localization cues namely, L-R and L+R which are identical for the both inter aural differences. There is no particular difference perceived in the frequency range up to 1,5kHz if the signal shows identical IPD for all wavelengths at the $\pm 30^\circ$ position or if ILD is present to represent said positions. This is also an explanation to why the same models for the perceived angle is applicable to these two inter aural differences.

ICTD

ICTD may be analyzed to result in IPD when the run time of the wavelengths is long compared to the ICTD. When the ICTD exceeds the run time of the wavelength the result is ITD to the ears that rely on the channel separation achieved as seen above. At this point and above ITD is reproduced with a resolution that is on par with the ITD introduced to the listener from natural sources up to $\pm 30^\circ$. To extend beyond these points a slight exaggeration of ICTD is needed. The ICTD is then gradually aligning with the time resolution similar to that of a conventional stereo system.

16. The far field

Up to this point the near field has been discussed. The far field is of a more complex nature and will in a simplified form be treated here to gain basic understanding on the subject for the reader.

The far field is defined as the area where reflections from the room boundaries are more dominant than the direct sound from the speaker. The trade off compared to the near field is that in the far field, the stereo image is single dimensional but can be enjoyed over a large area similar to a high end conventional stereo system using mainly pressure points. The localization cue at work is then ITD

rather than ILD and IPD due to the different path lengths of the reflected sound see Fig. 21.

This is due to the reflections of the resultant left and right lobes that are dispersed at $\pm 45^\circ$ from the speaker system that shows ITD after it is reflected by the room boundaries due to different path lengths. It is then the precedence effect that judges the localization of sound. The only feature needed from the speaker system is that the driver size is large enough in order to generate strong lobes that can be reflected.

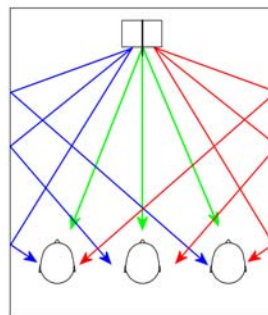


Fig. 21 Overview of the far field from a Single Bipolar Loudspeaker System.

17. System benefits

The main benefit of the Single Bipolar Loudspeaker System is the uncluttered time domain that makes the reproduction of phase differences meaningful. Another benefit is that level and phase difference finds a common ground through the L+R and L-R information reproduced from the speaker system. The uncluttered time domain provides the human with a more natural and relaxed way to discriminate and localize sound sources. Other benefits are:

- ICTD is reproduced on par with the resolution of the human localization.
- One localization cue is at work from 0 degrees to ± 90 degrees. (IPD all the way instead of ITD up to the speaker positions to be extended by IPD beyond these points as in a conventional stereo system).
- Less room dependant due to the direct sound wave containing the entire spectra.
- Driver size in relation to the listening distance is the only limit of how low in frequency the stereo information is perceived.
- The frequency response is not affected by any time dependant delays that results in cancellation of certain wavelengths.
- Improved phase coherence makes overtones easily detected and through this lower bass notes can be perceived, even if the fundamental of the tone is lower than f_0 .
- The most important information for the human perception, L+R and L-R, is always reproduced with the highest fidelity.

18. Disadvantages

The distinct difference between the Single Bipolar Loudspeaker System and conventional stereo is the reproduction of the former loudspeaker positions $\pm 30^\circ$.

This is the two points where the listener accustomed to the conventional stereo system is used to focus for good localization. Though it may be discussed what the scope of the reproduction is, if the position of the discrete speakers are of importance or if any other positions *but* the speakers should be localized. The use of one single source also attracts unnecessary visual focus as it is in fact one physical box in front of the listener. This is however a question of accustomation and may only be an issue for separate speaker systems utilizing the technology. This disadvantage does not apply to any product that desires to include the Single Bipolar Loudspeaker System in order to improve the stereo reproduction from a small form factor.

19. Conclusion

The Single Bipolar Loudspeaker System conforms to the new definition of stereo set out above, namely to reproduce the electrical inter-channel relationship - all information present in the recording - to the listeners hearing sense. The Single Bipolar Loudspeaker System is always *capable* of describing multidimensional sound but depends on the nature of the signal that is intended to be reproduced.

The way to reproduce the stereo signal in its L+R form and L-R form also approaches how the brain works in discriminating and localizing sound sources and makes the inter-channel differences to directly communicate with the listener senses in a more natural way. This makes the listener experience more from the Single Bipolar Loudspeaker System than from any conventional stereo system. It is interesting to note that the L+R information represents the depth (y - axis) of the stereo image and that the L-R information represents the lateral plane (x - axis). The Single Bipolar Loudspeaker System stands above the conventional stereo system when used as a reference monitor speaker system. When a mix is monitored and tailored for the Single Bipolar Loudspeaker System that relies on an equation (Eq.7) that only include the angle of the phantom source, the mix will be truthfully reproduced in any conventional stereo system that also rely on the speaker separation angle (Eq.1), but will of course be altered by the actual speaker configurations own features and flaws.

20. Applicability

The Single Bipolar Loudspeaker System is originally targeted for the professional use where it surpasses any previous known reference speaker system due to its uncluttered time domain and there through gained precision of the stereo image. The system is optimal to implement in all applications where reproduction of multidimensional sound is desired. Obvious applications include where conventional stereo previously has been impossible to implement due to the form factor of the products but that now is made possible with the Single Bipolar Loudspeaker System. It must however be understood that the advantages goes far beyond this as also the sound quality is improved and the imaging is maintained even if the quality of the components are poor.

The system can be implemented in anything from mobile phones, PDAs', Lap tops, TV sets, automotive audio etc. to high end hifi and studio monitoring speakers for critical reference monitoring.

21. Future work

Further investigations will be pursued on the subject of the resolved phase difference of humans in the region 1,5 - 10kHz. Other fields that will be covered include stereo microphone configurations and their function in the Single Bipolar Loudspeaker System and applications in combination with downmixes from multi channel sound formats. There will also be further investigations of alternative driver configurations concluded by the author.

22. Acknowledgements

The author would like to thank: PhD Håkan Hådeby for all criticism that has helped to get the consistency of this paper and Fredrik Solenberg for his critical eyes and bright mind.

23. References

- Ref. 1 Streicher, R & Alton E "The New Stereo Soundbook, 2nd ed." Audio Engineering Associates ISBN 0-9665162-0-6, 1998.
- Ref. 2 Atal B.S., Hill M., Schroeder M.R, "Apparent Sound Source Translator." United States Patent Office, No. 3,236,949, February 22, 1966.
- Ref. 3 Kirkeby, Nelson, Hamada. "The "Stereo Dipole": Binaural Sound Reproduction using Two Closely Spaced Loudspeakers." AES Preprint # 4463 (16). 1999
- Ref. 4 Bock T. M. and Keele D. B. "The Effects of Interaural Crosstalk on Stereo Reproduction and Minimizing Interaural Crosstalk in Nearfield Monitoring by The Use of A Physical Barrier" AES Preprint #2420A and #2420B 1986.
- Ref. 5 Blumlein A. D., "Improvements in and relating to sound transmission, sound-recording and sound reproducing systems", UK Patent 394325, 1931.
- Ref. 6 Clark, Dutton, Vanderlyn, JAES April 1958, volume 6, number 2
- Ref. 7 Sandel et. al. "Localization of Sound from Single and Paired Sources" JASA Vol. 27 #5 Sept. 1955 p842 - p852.
- Ref. 8 Bauer B. B. "Phasor Analysis of Some Stereophonic Phenomena" IRE Transactions on Audio 1962 Jan-Feb p.18-21.
- Ref. 9 Weinzierl S. "Kommunikationstechnik II Aufnahmetechnik" Technische Universität Berlin Institut Für Kommunikationswissenschaft.
- Ref.10 Bernfeld. "Attempts for Better Understanding of the Directional Stereophonic Listening Mechanism." Presented at the 44th AES conv. 1973
- Ref.11 L. Rayleigh, "On our perception of sound direction," Phil. Mag. vol. 13, pp. 214-233, 1907.
- Ref.12 Jeffress, L. A. A place theory of sound localization." J. Comp. Physiol. Psychol. 41, 35-39 1948.
- Ref.13 Hirsh, I. J. "The influence of interaural phase on interaural summation and inhibition." J. Acoust. Soc. Am. 20: 536-544, 1948.
- Ref. 14 Licklider, J.C.R. "The influence of interaural phase relations upon the masking of speech by white noise." J. Acoust. Soc. Am. 20: 150-159, 1948.
- Ref.15 Koehnke et al : "Binaural interaction experiments" J. Acoust. Soc. Am., Vol. 79, No. 5, May 1986
- Ref.16 D. C. Fitzpatrick et. al. "Neural Sensitivity to Interaural Time Differences: Beyond the Jeffress Model" The Journal of Neuroscience, February 15, 2000, 20 (4):1605-1615
- Ref.17 McAlpine D. "Creating a sense of auditory space" J. Physiol. 2005;566; 21-28;
- Ref 18 R. Woodbury: "MS- Speaker System". Speaker Builder 1/89
- Ref 19 Jon S. Fixler "Acoustic projection stereophonic system" US Patent # 4418243