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LOCALISATION OF SOUND WITH A MULTI SPEAKER REINFORCEMENT SYSTEM

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1. INTRODUCTION

The growing popularity of close microphone techniques using radio microphones in theatre sound has resulted in poor aural imaging. Before the use of amplified sound in theatre what the audience heard was the direct sound of the actors voice and not the amplified signal from a loudspeaker in a fixed and different location. This shortcoming has been commented on a number times in the press in recent reviews of stage plays as well as in professional audio industry publications, which has heightened both public and professional awareness of the problem.

Fortunately, the psycho-acoustic phenomenon known as Precedence or the Haas Effect, correctly applied can help the sound designer to achieve the seemingly contradictory objectives of a uniform loudness across the listening area while at the same time allowing the perceived direction of the origin of the sound to be controlled.

2. BACKGROUND

Today stereophonic and multi-channel sound systems in the home are common place, they have the advantage over monaural reproduction systems of giving the listener additional panoramic or spatial information which enhances the listening experience.

In concert halls and theatres where sound reinforcement is used, it is equally important to convey directional information about the localisation or source of a sound but this is at odds with the requirement to deliver an even sound pressure level with good intelligibility over a wide listening area.

The technique of level panning between left and right loudspeakers used in stereo reproduction is of limited effect in auditoria as only those people seated centrally between the loudspeaker positions will benefit fully from the effect. Just as with home stereo reproduction all the panoramic information is lost if a listening position directly in front of one of the loudspeakers is chosen, in theatre situations those listening positions nearest to the stage but away from the central block of seats hear the sound coming from the wrong place. It is arguably more important to convey spatial information in theatre sound, in order to reinforce the visual cues from the stage, rather than to contradict them which has the effect of undermining the audiences' willing suspension of disbelief and therefore their enjoyment of the show.

The principle of Precedence was first researched in 1840 and was later developed by Helmut Haas some 100 years later. Haas demonstrated that a listener seated in front of (and centrally between) two loudspeakers both set to the same level will hear a centre image. If the sound to one of the loudspeakers is delayed by small amounts of between 0.5 to 1mS, then the image will shift towards the undelayed loudspeaker.

When the time delay to one loudspeaker is increased to more than 1mS but less than 30mS, the delayed loudspeaker is not heard at all, even though its energy is equal to the undelayed loudspeaker. Time delay differences of more than around 40mS result in the delayed loudspeaker being recognised as an additional sound source which will adversely effect the intelligibility of the sound and cause stress and disturbance to the listener.

With a time delay of 10 - 20mS between the two loudspeakers, increasing the level to the delayed loudspeaker by 10dB will return the image to the centre and the delayed speaker will become recognisable as a sound source.

In summary the listener tends to lock on to the first arrival of sound and ignore subsequent arrivals within a 30mS time window even if they are louder. Secondary sounds arriving within the 30mS window tend to fuse or integrate with the original or preceding arrival and add to the overall level.

The Kutruff Effect further demonstrates that the upper limit of delay can be extended from Haas's maximum of 30mS by adding more loudspeakers with additional delays, so for example in a three loudspeaker system provided both the delay from speaker 1 to speaker 2 and the delay from speaker 2 to speaker 3 are within the Haas range, the image can be pulled from speaker 3 to speaker 2 to speaker 1, even though the total delay from speaker 1 to 3 exceeds the Haas maximum.

In practical terms the 30mS allowable error in arrival of the preceding and subsequent sounds allows the sound designer to pan sounds around a room by altering the relative time delays to the loudspeakers. In addition to this amazing phenomenon, in a properly set up system the effect can work over a large listening area.

3. PRACTICAL ISSUES

The ear is more sensitive to the direction of sound from in front and to the side than it is to sounds that come from behind or overhead. In a theatre sound reinforcement situation this is good because sound designers are normally concerned with pulling the sound from under balcony loudspeakers which are normally overhead, towards the stage which they are facing.

The timbre of the sound is also an important factor in the localisation of sound. The ear is most sensitive to the direction of sound at the same frequencies as it is most sensitive to level. Proper equalisation of loudspeakers can be used to extend the Haas limit of relative levels from two sound sources, but equally poorly equalised loudspeakers can destroy the effect.

Visual cues have a marked influence on the perceived direction of sound. However if the direction of a sound and a visual cue clearly contradict one another, this can lead to the listener developing a heightened awareness of the existence of an amplified sound system with a resulting loss of appreciation of the performance.

Room reverberation and echoes can both help and hinder the effectiveness of localisation of sound. In a room with very little natural reverberation, any comb filtering effects resultant in the arrival of the same signal two sources with a small delay and equal level will be clearly audible. In a room with a lot of reverberation the listeners perception of direction of sound can be confused.

Intelligibility of speech can be adversely effected if the listener hears multiple arrivals of the same signal with different delays, especially if the second and subsequent arrivals are outside the Haas effect limit of 30 - 40mS.

4. TECHNOLOGY AND A SOLUTION

TiMax is a new DSP based fully automated audio matrix that offers static and dynamic control of level and time delay at every matrix cross point. TiMax is among the first audio matrix systems ever that makes time delay panning and therefore the use of the Haas effect possible in any distributed or surround sound system.

TiMax offers a maximum capability to route up to 32 inputs to 32 outputs, with each of the 1024 crosspoints having a unique level and time delay setting. The control software is cue driven allowing an infinite number of audio events both static and dynamic (hard disk space allowing) to be saved to memory and recalled by a variety of means, including manual recall, remote control or time code triggered.

TiMax is controlled from a PC running highly graphically oriented control software on a Windows 95 platform. The PC communicates with the DSP via a high speed parallel link, and offers real time intuitive control. The PC is supplied less monitor in 19" rack mount format, complete with SMPTE / MIDI interface, 2 CD ROM drives, a removable 1 Gigabyte hard drive, a 1.2 Megabyte 3.5" floppy drive, an internal modem for remote control and fault diagnosis, serial and parallel ports and SVGA card.

The matrix core and audio input and output cards is housed in an 8 unit high 19" rack mount case and is based around the Analogue Devices SHARC DSP with internal 40 bit architecture. Up to 4 devices operate in parallel offering facility to accommodate up to 32 inputs and 32 outputs. External world communications and boot up is controlled via a smaller "Supervisor" DSP.

Inputs and outputs to TiMax may be analogue or digital. Audio Inputs and Outputs are arranged in groups of 8 on plug in cards with two versions available offering either analogue or digital AES/EBU inputs and outputs. Each SHARC administers 8 inputs and 8 outputs. Analogue and digital input and output cards can be mixed within the system so a system could have for example 24 analogue inputs and 8 digital inputs with all 32 outputs analogue.

Analogue inputs and outputs are arranged in groups of 8 each in plug in format, with each card containing 8 audio connectors electronically balanced on XLR with ground lift switches isolating pin 1 on the XLR. Analogue inputs use 20 bit stereo A to D converter running at 48KHz sample rate, while analogue outputs use a 20 bit stereo D to A.

Digital inputs and outputs have 4 stereo AES/EBU digital audio connectors on XLR per card. Digital input receivers feed to a sample rate converter to cater for asynchronous incoming data at any standard sample rate. Digital outputs are 48khz only.

Another 8 "bypass" analogue inputs are provided on an optional input card that allow audio signals to be summed into DIP switch selected outputs. These bypass inputs may be permanently assigned or automatically activated by a relay in the event of a system failure.

5. THE SOFTWARE

Venue Set-up

Several venues can be programmed into the TiMax computer, with a separate record card for each venue with notes and basic information such as address, phone number, contact names etc. The active venue can be selected from a browse list.

Details of which loudspeakers are in use, their names, types, location, multicore line, amplifier type etc. are entered as part of the venue set-up. Additional notes space is provided for each output and loudspeaker and other output types can be set up to appear in a pick list to speed entry of details. All the data entered can be printed in report format.

The physical layout of the venue can be imported from any Windows drawing program that allows files to be saved in bit map (*.Bmp) format. Here positions of loudspeakers and any other features of the venue can be noted.

Time delays to correct for loudspeaker position can be programmed as part of the venue set-up. These delays would typically be for under balcony speakers or for speakers at other locations that are not be used for primary sources for sound effects, i.e. where the delay settings are not going to change from cue to cue.

Show Set-up

TiMax will store details of several shows, each show has its own record card and associated details. Loading a new show will not change any of the venue set-up, and loading a new venue does not alter the show set-up.

Show details can include information on which inputs are in use, their names and type, the actors name, the input channel on the console etc. A right click on the mouse brings up an additional comments box. All the data entered can be printed in report format.

Specific show details such as MIDI device names and channel numbers as well as audio Image Definition names are entered as part of the show set-up.

Image definitions and sound effects

An Image Definition is a level / time delay relationship between a group of loudspeakers. The loudspeaker nearest to the desired sound source would have a short time delay, or no time delay at all, loudspeakers further away from the source have progressively larger time delays so that a coherent sound emanates from the desired location with even level over a larger listening area than would be possible if the Image Definition consisted of only one loudspeaker.

To include a loudspeaker in an Image Definition, double click the icon representing the speaker to move it from the parking area to the control zone. Dragging the icon up the screen will increase the level and dragging the icon across the screen will increase the time delay. Precise level and time delay parameters may be entered into the values box. As parameters are changed, control data is sent to the DSP matrix in real time so the effect can be heard and trimmed as necessary

Once a set of Image Definitions have been programmed then the effects screen can be used to control and program dynamic sound effect movements. This is done by bringing the Image Definitions for the effect into the control zone by double clicking on them to activate them, and then selecting the input signal with the source for the effect in the same way. Then by clicking on the source and dragging it about the control zone the sound movement is programmed and may be saved or updated as required.

Several dynamic sound effects can be run at the same time, the system allows dynamic effects to the programmed in layers, so that programming is not over complicated as well as to make straight forward editing and updating possible.

Event Control

The TiMax control software offers sophisticated show control facilities including MIDI events management, Serial port events on RS232 / 422, time code events with MTC and SMPTE compatibility, relay closure events and CD play events from two internal CD drives. This facility enables TiMax to take care of all the show control requirements to be actioned in background each time the next cue is recalled.

TiMax allows up to 16 MIDI events to be programmed within each cue. Program changes can be used to control external MIDI compatible equipment such as reverberation or equalisation units by recalling settings from the units internal memory. MIDI note on commands have an associated duration setting to they can be used to trigger sound effects stored on a sampler or to play a note on a synthesiser, TiMax then automatically sends the note off command at the end of the duration. MIDI controllers can be used in many different ways including; levels of sound or lights, pitch of a synthesiser note, equaliser or reverb parameter ...

Serial port events can be programmed either in ASCII or Hexadecimal to allow simple interface to equipment that offers serial port control, for example some sound effects playback equipment offers this type of control.

An internal time code generator can be started and stopped at precise time code values for purposes of external synchronisation of tape playback machines connected via a chase synchroniser, or for timed follow on cues set to automatically trigger after a programmed interval from the last cue.

CD tracks can be set to automatically start to play from either or both of the two CD ROM drives built into the controlling PC.

A relay interface built into the controlling PC can be programmed to close the relay contacts for a user defined duration. This feature can be used to start tape cartridge or reel to reel machines, to trigger lighting or pyrotechnic effects, to ring telephones .....

Play List

Cues are recalled from the play list which is a scrolling display that moves forward on pressing the + key. Cues may be automatically or remotely triggered and may be defined as Time Code triggered cues, MIDI triggered cues, follow on cues or manually triggered cues.

The Play List shows the previous 4 cues, the current cue and the next 7 cues in the running sequence of the show. Associated with the current cue is a large notes area for operator prompts and reminders. TiMax allows decimal point cues to 3 decimal places to be inserted into the running sequence, thus allowing new cues to be added without having to renumber the script. The play list also prominently displays the current time code value and any CD tracks that are playing during that cue.

Matrix

Programming set-ups can be achieved using the Matrix screen. An input - output crosspoint or a range of crosspoints can be selected and the level and time delay may be set using the control faders. Positioning the mouse over a cross point and clicking will select that cross point for editing and highlight its input and output names. To select a range of crosspoints click and drag if the range is continuous, or click and Ctrl click if the range is discontinuous. To set all inputs to one output, click on the output name and in the same way to select one input to all outputs, click on the input name.

Level changes can be made in relative or absolute mode by clicking on the desired mode button. In relative mode if a range of crosspoints have been selected, the level and delay fader will jump to the level of the highest crosspoint in the range. Increasing the level by 3dB will increase the level of all of the cross points in the range by 3dB. In absolute mode the same action will set all crosspoints to the same level i.e. to 3dB above the level of the highest in the range prior to adjustment.

Once a set-up is programmed it can assigned a cross fade rate and be saved to a Cue number and name and later recalled.

Faders

Crosspoint levels can be set on a fader screen as shown. If an input fader is selected the output faders will show the levels that that input feeds to the outputs, or if an output fader is selected the input faders will show at what levels the inputs feed to that output.

Bargraphs

Audio levels can be monitored on the bargraphs screen. The characteristic of the bargraphs can be altered to suit the operators preference between VU and PPM and Peak Hold responses, with a range of scales available from -30dB to +6dB, -60dB to +20dB and -10dB to +10dB.

6. APPLICATIONS

In sound reinforcement applications where a few inputs need to be distributed to several outputs each with a unique or variable time delay and level, TiMax is the ideal control system. Film sound recording is another typical application for TiMax. Spatial sound effects from a number of loudspeakers to an audience seated in a large listening area can only effectively be achieved using time delay panning (Haas Effect) as well as level panning.

In fixed installation sound systems where the only criterion is to have a number of inputs feeding to outputs with a delay, TiMax offers a cost effective solution compared to the alternative of a hand full of professional delay units.

TiMax is also suited to applications where a number of inputs need to be time aligned and mixed together, for example in classical recording where microphones are placed both close to instruments and some distance away for stereo perspective.