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TSS System and Live Performance Sound

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ABSTRACT

The Total Surround Sound System (TSS System)*, as one of possible technological audio standards, which uses a special digital processor with 8 sound channels, beside applications in HDTV projection systems, theatre and multifunctionhalls and such like, could also be used in creating a Live Performance Sound. The TSS concept could be most interesting in sound reinforcing rock concerts and rock operas, musicals, various multimedia stage shows and such like.

*TSS System - pat. pending

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FUNDAMENTALS OF THE LIVE PERFORMANCE SOUND

In its first line association, the term of "Live Performance Sound" (LPS), is somehow irresistibly connected with a sociological and cultural phenomenon otherwise known as ROCK.

All technological advances and enhancements regarding LPS are in conjunction with the initial milestone from early fifties, when the world of entertainment produced the original Rock and Roll.

The next two decades saw a veritable "Big Bang", and in technological terms, it culminated in 1974. It was in that year that the anthological group "The Grateful Dead" presented in the Hollywood Bowl its equally anthological sound system [8] which by its concept, originality and performance surpassed everything seen and heard until that time as far as the sound stage is concerned.

The eighties brought a continuation of the "Big Bang", primarily dominated by advances made in digital technology.

Accordingly, the Live Performance Sound cannot be analyzed without defining exactly its roots in Rock. As is generally recognized, two basic concepts are used today.

THE SPLIT STACK CONCEPT

This is the simplest method of sound reinforcement and was used during the initial stages of LPS development. This concept uses two distinct and separated stacks, one on the left and the other on the right side of the stage. Today, this sound reinforcement method is mostly used for on-the-road applications, as its process of installation, or stacking up of loudspeaker boxes, is simple and expedient. This method has the following faults:

- unpleasant sound pressure levels to which the spectators are subjected if they are near the stacks;
- uneven reverberant sound contents for spectators located in the first and last few rows in the audience;
- poor sound coverage of the spectator hall rear end;
- time delayed sound between the stacks;
- unfavorable interference effects resulting from phase cancellation between the stacks, and
- a reduction of the resulting directivity coefficient (Q) by 50% on spectator hall sides, this in turn unfavorably reflecting upon the critical distance (D_c) and intelligibility ($\%AL_{cons}$ as a criteria).

THE FLYING SPEAKER ARRAY CONCEPT

In contrast with the previous system of split stacks, this method uses a configuration consisting of two basic sound clusters which are hung up above the stage using special support constructions.

In cases when greater sophistication is sought, the system may also have a third, central sound clusters, which clearly defines the sound stage center and eliminates the virtual source effect.

The Flying Speaker Array system began to be used intensively during the eighties, both in and outdoors. Its main characteristics and virtues are:

- it eliminates the most unpleasant proximity sound field for spectators near the stage itself;
- the entire spectator area is well covered;
- the performers on the stage have fully correlated visual and sound axes; and
- the multiple sound source interference effects are eliminated (using CD horns).

Common to both systems is the possibility of using component or boxed type sound sources.

It is well known that component loudspeaker systems use separate and independent bass, midrange and treble loudspeakers, located in separate units, connected to the power amplifiers via crossover networks. The boxed systems, on the other hand, consist of compact units with integrated bass, mid and high loudspeakers.

The component systems afford better coverage of the audience area, they have a higher level of system headroom and they produce lower levels of distortion.

On the other hand, the boxed systems are highly flexible, since they utilize the modular approach. A well designed vented box system attains higher sound pressure levels at lower frequencies ($f < 60$ Hz) than a horn loaded system in its component version.

In order to attain a greater level of intelligibility (using $\%AL_{cons}$ as a criteria), a more adequate ratio between reverberant and direct sound, as well as the SPL required for the entire audience area, special attention should be paid to component selection, especially in case of mid and high range drivers; this is defined by corresponding directivity coefficients (Q). Because of this, the CD horn systems ($Q > 10$) offer great advantages throughout the very significant midrange (2.5 to 5 kHz).

THE TOTAL SURROUND SOUND SYSTEM (TSS System)

This is the next, higher step in Live Performance Sound.

The fundamental virtue of the TSS System is its eliminating the monoaxial sound information and transferring it throughout the audiency. It was this monoaxial sound which was the hallmark of the preceding two concepts of LPS.

Thus, the TSS System provides for full spacial or multiaxial 3-D impression.

The effects, and the entire philosophy behind the TSS System may best be seen on Figure [1], which illustrates the special TSS network in its symbolic form. It will show that the audiency is fully surrounded by sound.

Using the digital TSS processor, which handles the 8 channel sound information, it is possible to create most varied sound effects. Evidently, the basic sound information (the direct sound) is provided by three clusters mounted above the sound stage.

Calculating the coverage and other relevant acoustic parameters in connection with direct sound generation (e.g. calculating the critical distance or required electrical power from the amplifiers needed for attaining the desired sound pressure levels at a given distance from the sound source) can be done using existing software and PC or compatible computers [2].

Beside the three channels mentioned above (needed for the direct sound), the TSS concept uses another five channels, and via cor-

responding sound sources, they generate additional sound support for the entire LPS (effects, noises, additional vocals processing, music scores, and such like).

It is evident that this provides for a new symbiosis in communication from the stage to the audiency, while the traditional stage sound becomes a new, imaginative Live Performance Sound. All that becomes a new, great challenge for future creative, composer and producer approaches to live concert sound.

Due to its complexity and sophistication, in its optimal implementation, the TSS system will require the construction of special or multifunction halls. Proper designing and furnishing of such halls are of paramount importance [1][2].

New measuring techniques for room acoustics, which use MLSS Analyzers, greatly ease and shorten the measuring procedures, since all required subprocessing of data obtained in the field (the Impulse Response has been recorded on a diskette) can subsequently be done in the lab using a PC [10].

Using a MLSS analyzer, one can arrive at an integrated audio system data for a room, parts of which is most important data, such as: the frequency response, reverberation time, speech intelligibility, ratio of reverberant-to-direct sound, energy time curve, speech transmission index, etc.

Beside the above essential acoustic room measurements, while sound reinforcing live programs and resolving certain speaker/room anomalies in frequency responses, one will also need to dynamically equalize the electro-acoustic system in real time. To do so, one can use a dual FFT computerized analyzer [5].

All this brings us to a new futuristic concept of a so-called Self-adjusting Concert Sound System. The TSS System strives towards just this concept in the near future.

Of course, the TSS System also allows making use of the DSS concept [11] in cases when a convincing combined sound and visual stage information is required.

In practice, implementing the TSS System may bring about several problems.

The first of such problems may appear due to highly specific location of sound sources in the room floor and may have to do with both the construction and acoustic problems (radiation and coverage conditions). This in fact requires a separate and specific study.

Additional problems may be in conjunction with the practical conditions of wiring, or connecting the loudspeakers with the amplifiers. In view of the large number of amplifiers and loudspeakers at hand, making use of card amps represents a logical solution for ceiling and floor channels. It is also possible to use active monitors, which will depend on the size of the room and the required TSS System resolution in conjunction with the Travelling Sound Effect.

One should also note that in case of a stage flying system, long wire lengths may cause stability problems at higher frequencies (20 kHz). Thus, a large damping factor at high frequencies could be a good indicator of a well designed and stable audio amplifier.

The TSS System could possess, quite independently from the TSS processor, a specially developed self-analyzing system which could monitor and report all significant data on the status of the amplifiers in the system, on dynamic equalization, sound pressure levels in individual room parts, etc.

Of course, it is possible to incorporate other hardware and software ideas which will doubtlessly turn up in the process of gathering experience in working with the TSS System, which could be related to psychoacoustics.

THE BASIC CONCEPT OF THE TSS PROCESSING UNIT

The block diagram of the entire system is illustrated on Figure (2), and it will illustrate a number sources whose signals are processed in the mixer in form of eight group outputs and three master outputs (left, center and right). All these signals are routed to the TSS processor, which provides for creation of various sound images, effects, etc. The TSS processor is controlled by the central PC-class computer with a color monitor, keyboard and a hard disk for its mass memory. This mass memory contains a library of effects and previously created instruction sequences, which in turn contain all necessary commands for controlling the TSS processor.

According to the beforehand determined show procedures, the operator can use a special editor, in conjunction with the keyboard and monitor, to produce a program which consists of numerous separate effects defined by type, sequence and basic duration. Execution of this program can be controlled with the remote control unit, which may be situated with the sound engineer (Figure (3)) and connected with the central computer using an RS 232C interface and cable.

The sound engineer can also view and control the current effect on an LC display. He selects the time of execution of an effect by pressing the START key. The STOP key will immediately terminate any effect, while using the CONTINUE key will have the effect go on from the place it was halted. Successive START commands will cause an effect to be repeated from the beginning. The PITCH control will change the duration of an effect in relation to its nominal duration as defined by the program. The NEXT command will cause the currently active effect to be replaced by the next one on the program list.

It is also possible to randomly sequence the effects listed in the library quite outside the predetermined sequence. It is sufficient, at any moment, to simply type the number of an effect and to confirm the choice by pressing ENTER; this will cause the effect to become immediately active and ready for use.

Execution of above commands is performed by transferring the instructions from the PC to the TSS processor via a parallel interface. The TSS processor block diagram is illustrated in Figure (4), and it may be seen that there are several individual parts, of which more will be said later. In principle, it may be said that this is an analog processor with digital control. The fundamental TSS concept provides for processing of digitalized audio signals as well, but in this application, we will not deal with that aspect. The reason for this is in the nature of implementing the Live Performance Sound, which at this stage of technology development uses analog technology only.

The TSS processor passes through the basic direct sound information just as it was created at the mixing desk by the sound engineer on his master channels (left, center and right). Only if expressly required for effects, these channels may be routed to the TSS processor outputs and this will provide for having three sound groups to be used as basic information for continuous production of effects.

The TSS processor subsystems and their characteristics will now be described in short.

a) <u>The preset amplifiers.</u>
These are to be used to adjust the initial levels for all channels.

b) <u>The VCA Matrix</u>
The matrix of voltage controlled amplifiers serves as a digitally controlled mixer which enables continuous mixing of all group channels with one or more outputs. The matrix consists of eight independent groups of VCA'S ROW, which are illustrated in Figure (5).
The computer control system, via the parallel interface and the control logic, sets the registers of DATA REG. with data on the desired levels for each channel and of ADR & COM REG. with data on the desired channel (i.e. its address) and with required DAC

control signals. By scanning, the sequence of which is no greater than 10 milliseconds, the computer maintains for each output separately the ratio of input signals required in any one moment for implementing the desired effect. The VCA circuits are the ones who manage the 10-bit DAC, thus providing for 1024 discontinuous intensity levels during signal mixing.

c) <u>The Lineout Amplifiers</u>
These provide for separating the outputs leading to the three main sound groups (Direct Sound) if the BYPASS circuit allows it.

d) <u>The Bypass</u>
Acting on computer commands, this circuit passes signals from the master channel or the TSS processor on towards the main loudspeaker stacks.

e) <u>Travelling Sound Processors</u>
These provide for additional processing of audio signals to be routed to loudspeaker enclosures needed for the Surround Sound, Ceiling and Floor sounds. The block diagram of this subsystem is illustrated in Figure (6), which will show that this subsystem enables dynamic distribution of the input signal to sixteen output lines. Each output line may have an attenuation factor of infinite, 3 and 0 dB. Data on this is stored in registers REG1, REG2 and REG3 respectively in form of digital information and a defined resistor network turns this data into control signals for the VCA circuits.

f) <u>The Control Logic</u>
It accepts signals from the central computer via the parallel interface and turns them into command signals for each of the above subsystems, and into data required for TSS processor operation as a whole.

A possible outlay of a module 19"/6HE rack for the TSS processor is illustrated in Fugure (7). Optionally, it may be possible to have the TSS processor also manage an external subharmonic generator for the subwoofer sections of the Direct Sound left and right channels.

AN EXAMPLE OF THE CALCULATION OF A LIVE PERFORMANCE SOUND SYSTEM IN A MULTIFUNCTION HALL

This example will use the calculations made for a Live Performance Sound System for the large hall in the SAVA CENTAR in Belgrade; the procedures used have been defined in [1], [2] and [3].

The SAVA CENTAR large hall was not designed with TSS in mind, but rather as a multifunction hall with variable acoustics for various shows, ranging from classic music concerts, congresses, movie and theatre shows to pop and rock concerts. In order to cater for all these requirements, during construction works (in 1978), corresponding fixed and removable electroacoustic systems were included for sound reinforcement, reproduction of effects and ambiophony. These systems represent classic system solutions for sound reinforcement and do not fit into the TSS live performance sound concept, and will thus not be included here.

The authors decided to calculate the requirements for the live performance sound system in the large hall of SAVA CENTAR for several reasons, the most notable of which are:

- the authors are well acquainted with the required acoustic parameters of this hall;
- certain calculations for this hall have been produced already, notably in [3], and
- because of the potential implementation of the first TSS live performance sound system in just this very hall.

With respect to the configuration outlined on Figure (9), the basic parameters of the large hall in SAVA CENTAR are as follows:

Volume	$V = 20{,}000\ m^3$
Reverberation time	RT = 1.5 sec
Room constant	$R = 3{,}250\ m^2$
Hall dimensions (HxDxW)	12 x 35 x 42 m

The relationship between the height, depth and width of the hall is 1 : 2.9 : 3.5, which does not fully satisfy the requirements for a TSS hall [2], which should optimally be 1: 2.5 : 3. Evidently, the depth and width of the hall are greater than optimal, and side surround effects, no matter what the calculations say, will be well defined in the lower part of the hall, while the rows nearer to hall walls will be in a somewhat worse condition.

On basis of [1], [2] and [3], calculations are needed for:

1. Front sound sources (Flying Arrays);
2. Side Surround Sound sources;
3. Ceiling Travelling Sound sources; and
4. Floor Travelling Sound sources.

The front left-center-right direct sound sources are illustrated in Figure (8) and are located above the stage; calculations need to be done for the following parameters:

- medium SPL in the audience — Lt = 105 dB SPL
- effective radiation angle — $\Theta h \times \Theta v = 120^0 \times 60^0$
- sensitivity (1W/1m) — Ls = 107 dB SPL

The electrical power per Flying Array, with a 10 dB system headroom margin, required for all values as defined above, is provided in equations defined in [1] and [3]:

$$Pe = 10^{1/10\,[(Lt + 10) + (\Delta D - \Delta D') - Ls]} = 2512\ W \qquad /1/$$

$\Delta D - \Delta D' = 20 \log Dc = 26$ dB $(D > Dc)$

$Dc = 0.14\ \sqrt{(QxR)} = 21$ m

$Q = 180/\{\arcsin[(\sin \Theta h/2)(\sin \Theta v/2)]\} = 7$

On basis of calculations for each "Flying Array", there are to be four loudspeaker system units each rated at 700W, and the whole array being rated at a total of 2,800W. Each loudspeaker array is to be accompanied by a corresponding power amplifier rating with an electronic crossover network, with an equilizer and a compressor/limiter (see block diagram in Figure (2)).

In order to reproduce the low and extremely low frequencies in conjunction with the Flying Arrays, there are also to be subwoofer loudspeaker systems located on the floor in front of the stage. Relevant subwoofer calculations are to be done using known parameters of loudspeakers to be used [3]. With the calculations referring to eight mutually coupled subwoofer loudspeaker systems on each side (left and right), maximum sound pressure levels of 124 dB can be achieved at frequencies below 100 Hz, this further reinforcing the impression of the live sound.

As for consonant intelligibility, with simultaneous use of three "Flying Arrays", and according to Peutz equation [1][3], following losses are incurred:

$$AL\ cons = (200 \times D^2 \times RT^2 \times n) / (V \times Q) = 11.81\ \% \qquad /2/$$

Room length D = 35 m, number of arrays n = 3(1),
which is less than the allowed 15%.

Reproduction of side surround effects is to be achieved using loudspeakers built into the hall walls, the resolution and sound levels of which can be calculated using adopted loudspeaker and hall parameters:

*	effective radiation angle	$\Theta = 60^{0}$
*	sensitivity (1W/1m)	90 dB SPL
*	nominal power/impedance	60W/8 Ohm
*	first listener-loudspeaker distance	d = 3 m

(1) In case the vocals are reproduced only from the central array (n = 1), consonant losses will be AL cons = 3,93 %.

Since the radiation angle of each loudspeaker is

$$Rw = d \tan \Theta/2 = 1.7\ m \qquad /3/$$

the optimal distance between the loudspeakers is Dw = 2 Rw = 3.4 m, so that for a wall 35 m long the required number of loudspeakers is N > 10; we adopted 16, of which each loudspeaker is connected to a separate amplifier channel of the travelling sound processor. The same number of loudspeakers was adopted for the rear wall as well, since its length is 42 m (see Figures (9) and (10)).

As for the "cricket" loudspeakers [1][2] located in the hall ceiling, the calculations apply to loudspeakers of the same characteristics as those in the side walls, while the radiation angle of individual speakers is:

$$Rc = (h - l)\ tg\ \Theta/2 = 6.2\ m \qquad /4/$$

Hall height h =12 m, listening plane height from floor l = 1.2 m.

The optimal distance of the ceiling loudspeakers would amount to Dc = 2Rc = 12.4 m; thus, for a hall 35 m long, this would require a minimum of three rows of ceiling cricket loudspeakers. Since the SAVA CENTAR large hall also has a gallery, we opted for eight rows of ceiling cricket loudspeakers, each row containing 7 speakers (see Figure (10)).

As for the floor cricket loudspeakers, whose basic characteristics are the same as for the ceiling speakers with the exception of their effective radiation angle $\Theta = 80^{0}$, their individual radiation angle is defined by the following expression:

$$Rf = l \times tg\ \Theta/2 = 1\ m \qquad /5/$$

listening plane height from the floor l = 1.2 m where the optimal distance between individual speakers in the floor is $D_f = 2\ R_f = 2$ m. On basis of this calculation, the large hall of SAVA CENTAR will require sixteen rows of loudspeakers (35 m : 2 = 16) with 21 loudspeakers in each row (42 m : 2 = 21) (see Figure (9)).

Using above data, we arrived at the required number of channels on the travelling sound processor, as well as at the number and power output of the amplifiers.

The per unit power output of modular amplifiers for the surround sound processor's 16 channels and for the loudspeakers in the walls should be 50W/8 Ohms; at full power, and in the middle of the audiency (22.5 m away from the speakers), this would enable sound pressure levels of some 80 dB with only one speaker working.

The per unit power output of amplifiers driving each of the 8 ceiling travelling sound processor channels should amount to 400W/4 Ohms. Each amplifier would drive 7 loudspeakers in a row connected in series/parallel with a resulting final impedance of 3.2 Ohms. At full power, all seven speakers would provide a sound pressure level on their axis of better than 90 dB.

The per unit power output of amplifiers driving each of the 16 floor travelling sound processor channels should amount to 400W/4 Ohms. Each amplifier would drive 21 loudspeakers in a row connected in series/parallel with a resulting final impedance of 4 Ohms. At full power, all seven speakers would provide a sound pressure level on their axis of better than 103 dB.

These calculations can be made using software modified existing "Computer Aided Acoustic Designer" programs for PCs (NEXO-CAAD, JBL, etc.), which allow 3-D representations of audiency coverage using specified spacial acoustics (reverberation time, early reflections, impulse response, etc.) [2]. This will be described in greater detail in one of our future studies.

CONCLUSION

The new and imaginative Live Performance Sound, described in this paper, supported by high technology (video walls, HDTV projectors, lighting and laser equipment) will force all participants (both the audience and the performers) to experience a completely new artistic and creative event using this exceptionally powerful media of mass communication.

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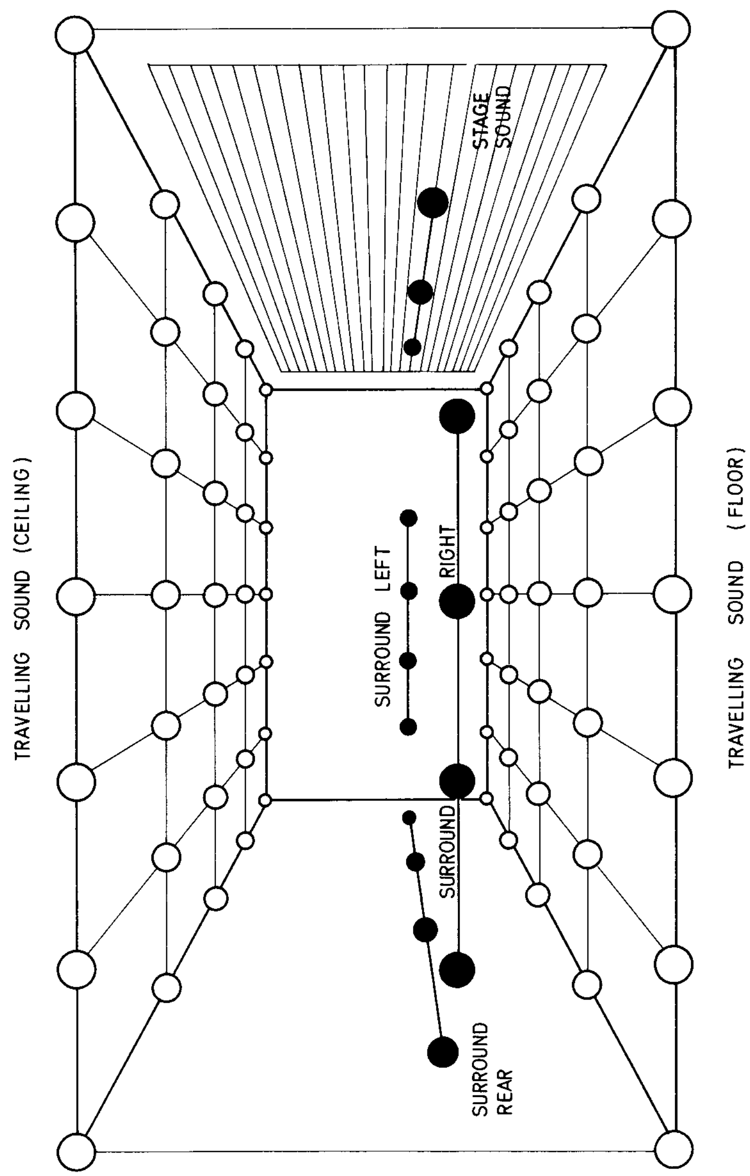

fig. 1 SPACE NETWORK IN A TSS HALL

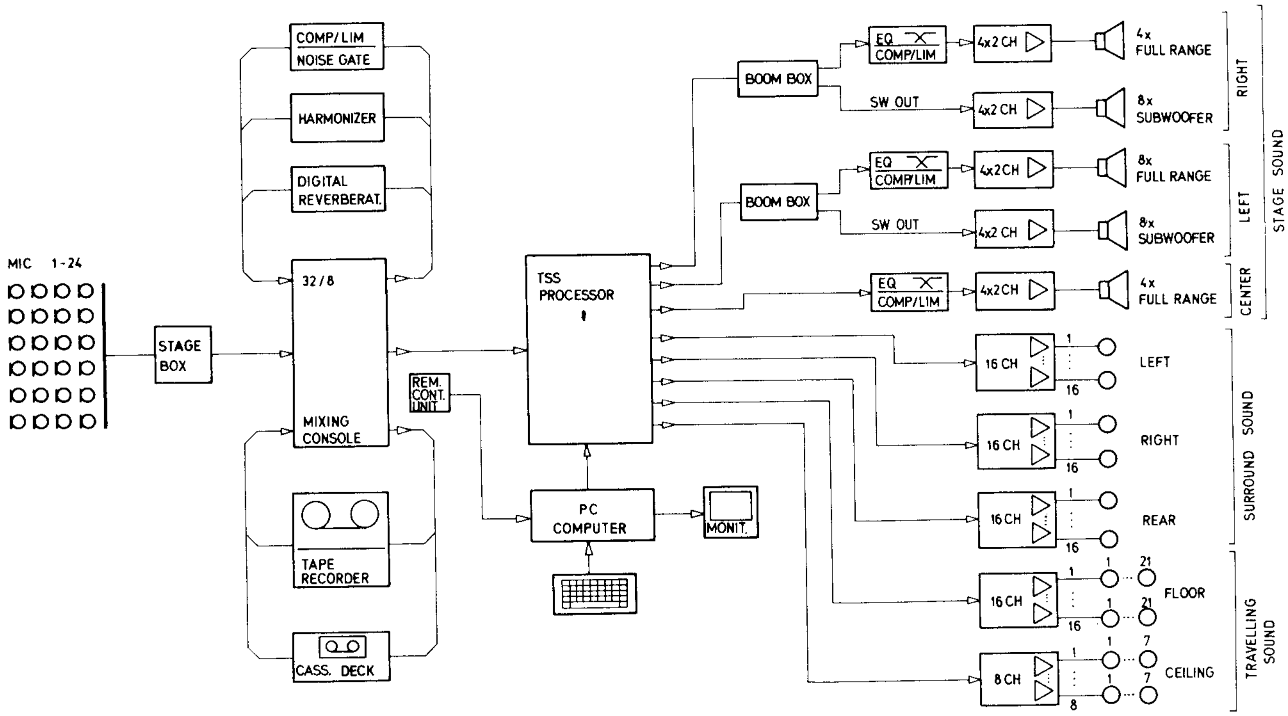

fig 2 BLOCK DIAGRAM of THE TSS SOUND REINFORCEMENT SYSTEM for LIVE PERFORMANCE

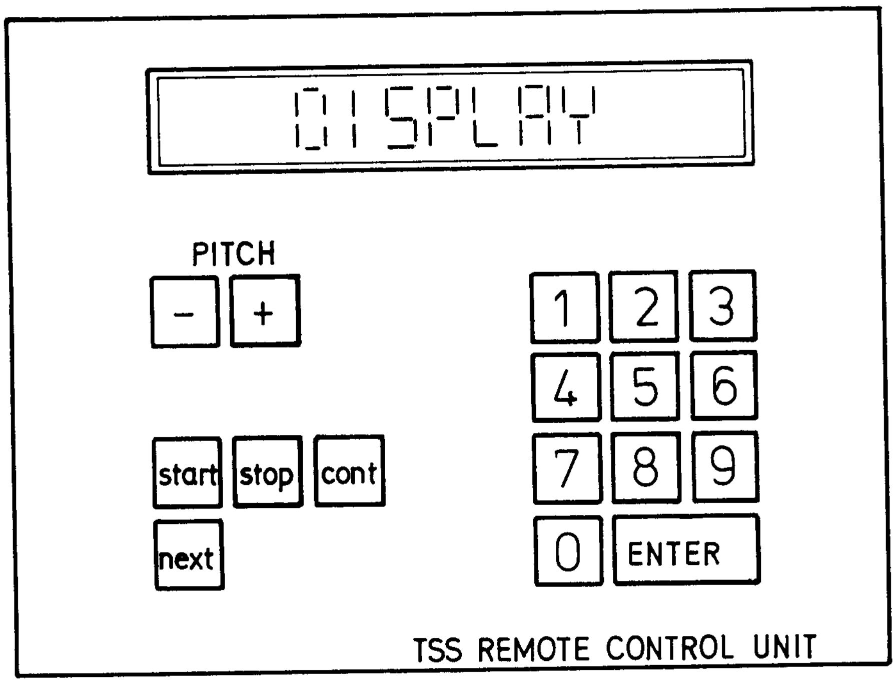

fig.3 VIEW of THE TSS REMOTE CONTROL UNIT

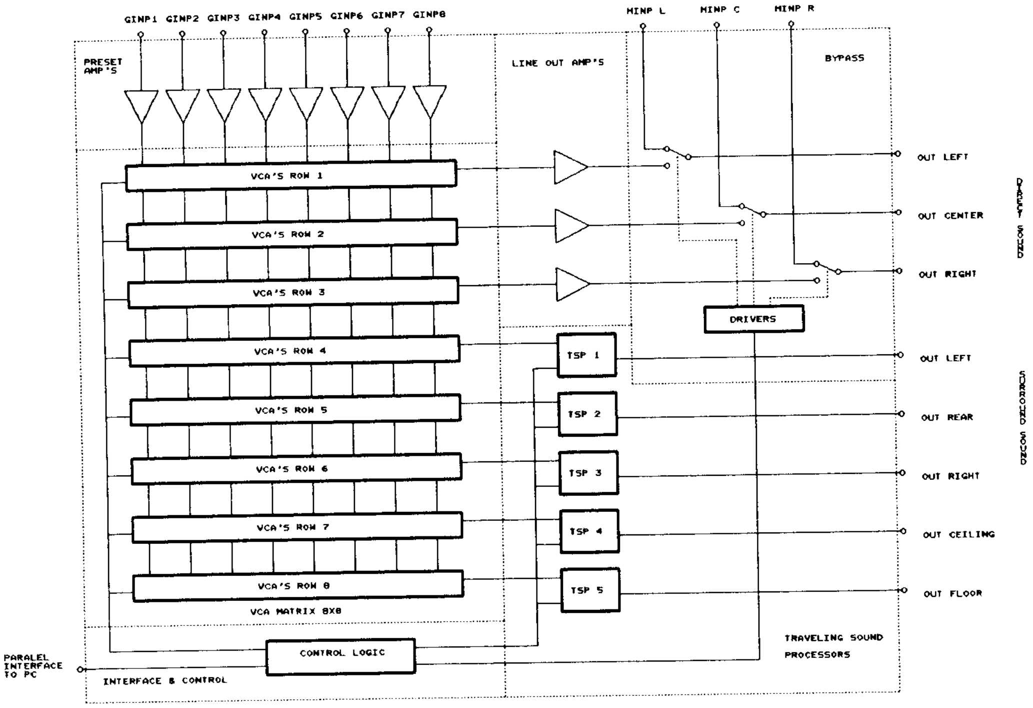

Fig. 4. TSS PROCESSOR STRUCTURE

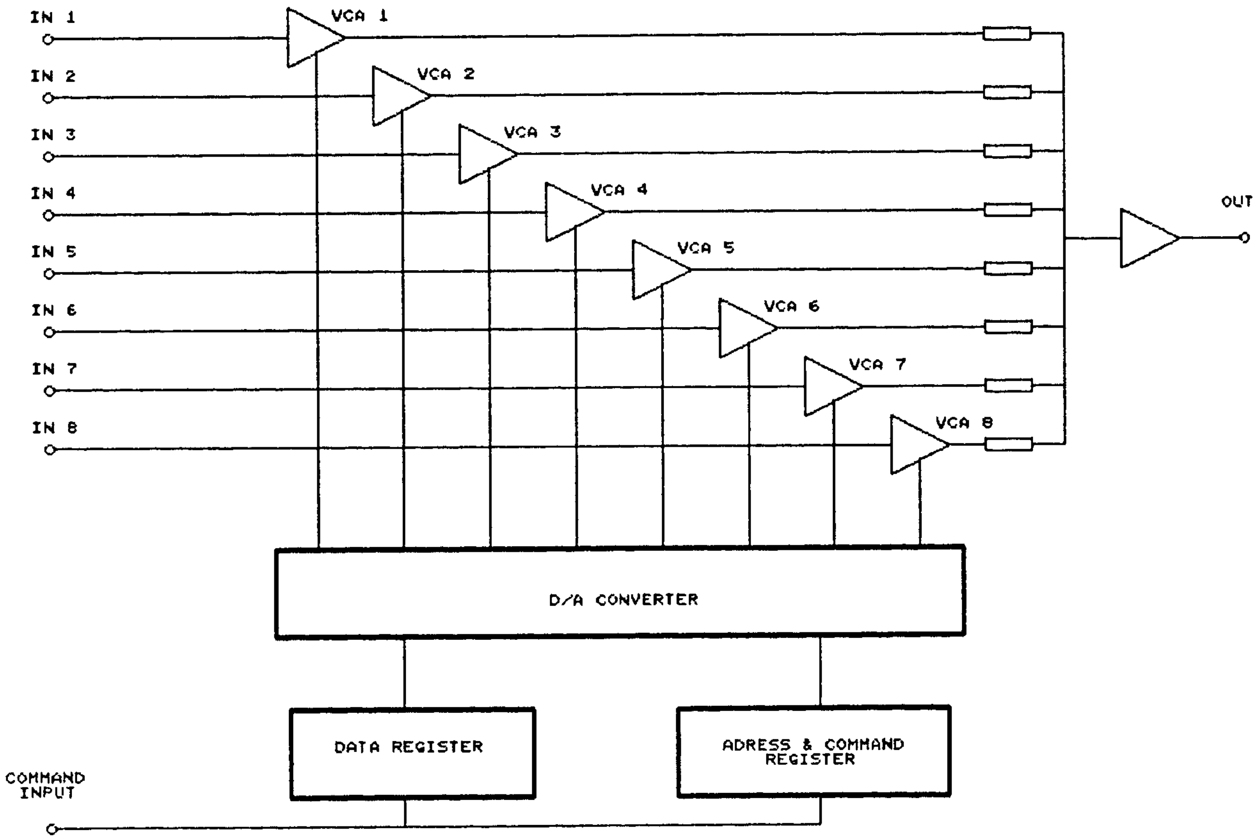

Fig. 5. PART OF VCA MATRIX - VCA'S ROW

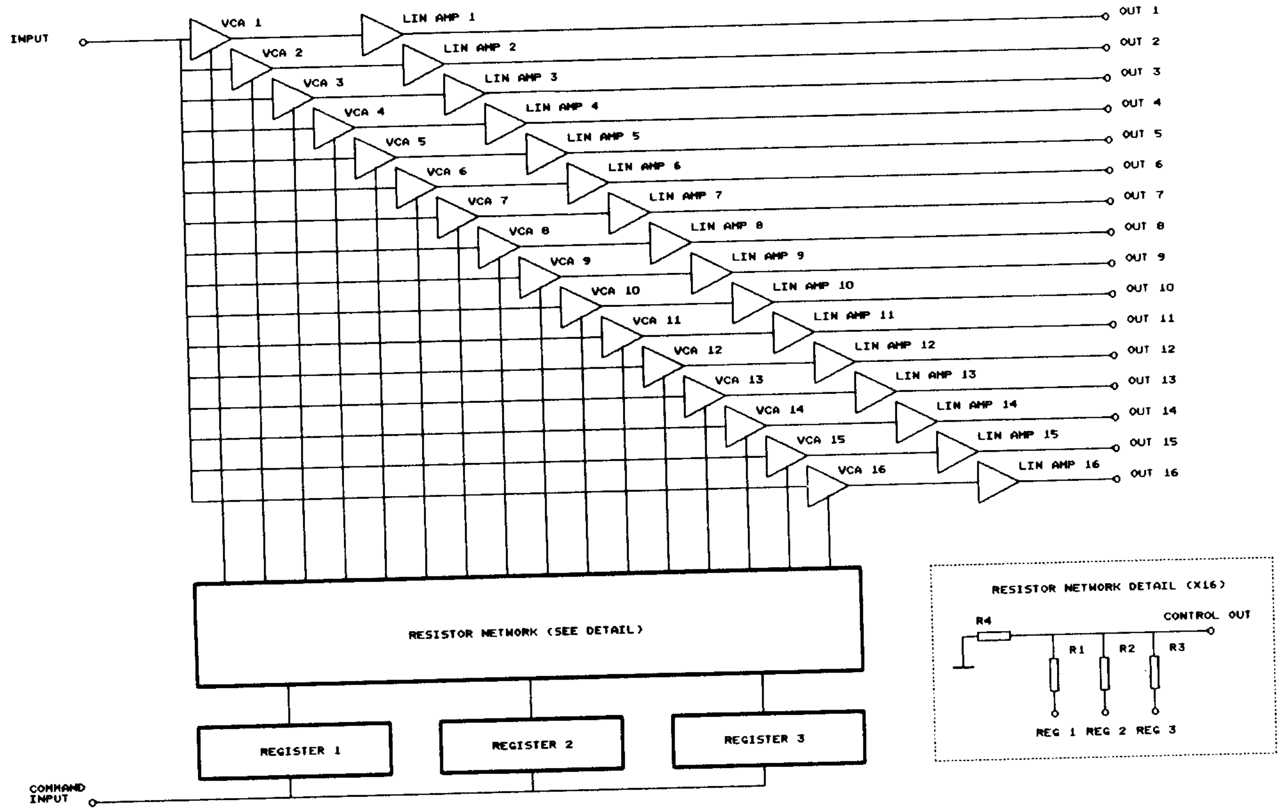

Fig.6. TRAVELLING SOUND PROCESSOR

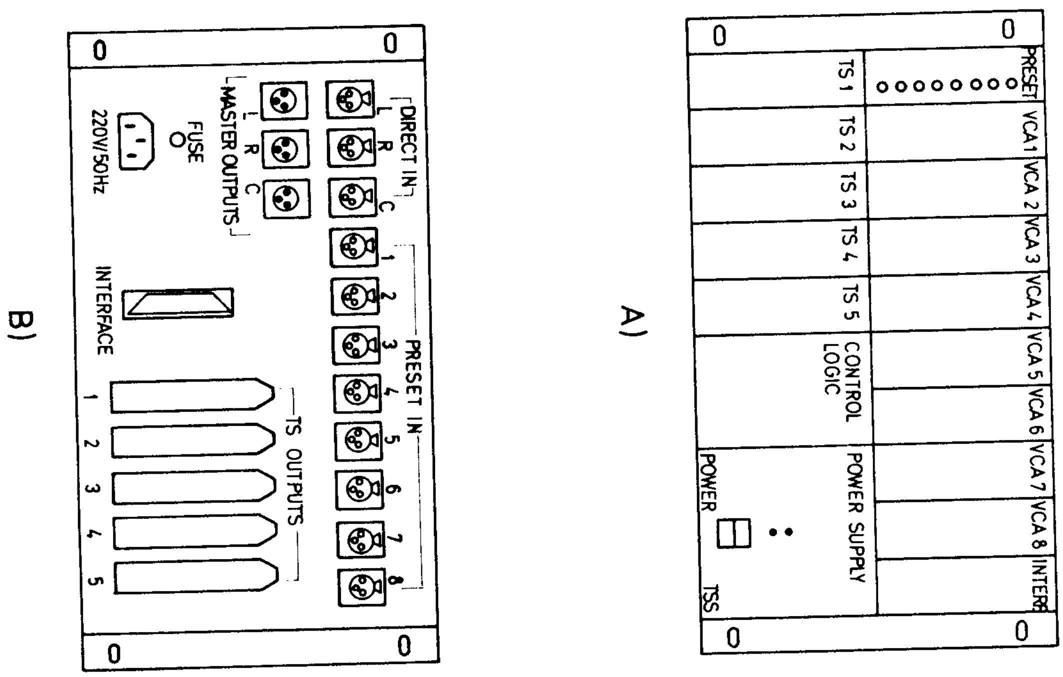

fig.7 VIEW of THE TSS PROCESSOR
A) FRONT, B) REAR

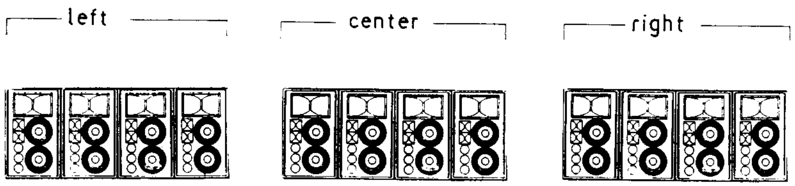

fig.8 VIEW of THE FLYING SPEAKER ARRAYS for STAGE SOUND

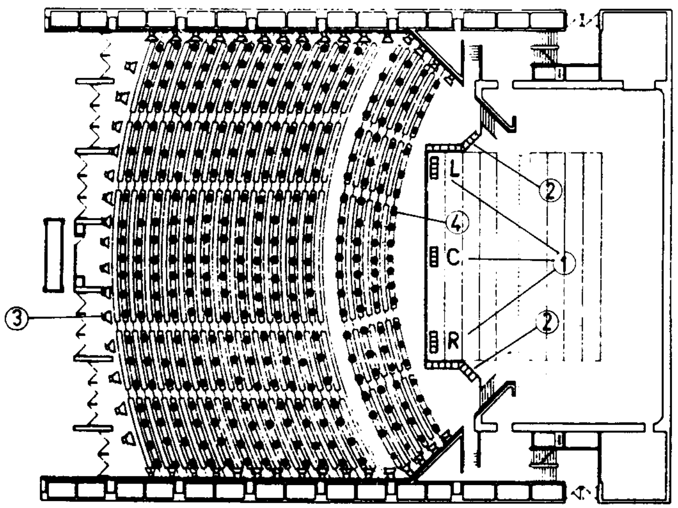

fig.9 LOCATION of THE LOUDSPEAKER SYSTEMS in THE BIG HALL of „SAVA CENTER"(plan): ① FLYING SPEAKER ARRAYS, ② SUBWOOFER SECTION, ③ SURROUND SPEAKERS, ④ FLOOR TRAVELLING SOUND LOUDSPEAKERS

REMARK:

THE SHOWN NUMBER OF FLOOR SPEAKERS ON THIS FIGURE IS A RESULT OF CALCULATION (SEE PAGE 13) FOR CONVENTIONAL SPEAKERS ($\Theta = 80^{\circ}$ AT 4 KHZ). THIS NUMBER COULD BE SIGNIFICANTLY LOWERED THROUGH APPLICATION OF SPECIALLY CONSTRUCTED FLOOR SPEAKERS WITH WIDER RADIATION CHARACTERISTICS ($\Theta = 90^{\circ}$ AT 4 KHZ).
THE TECHNOLLOGICAL CONCEPT OF THESE NEW TYPE OF SPEAKERS WILL BE PRESENTED IN ONE OF THE FUTURE PAPERS OF THE AUTHOR ON THE SUBJECT OF TSS SYSTEMS.

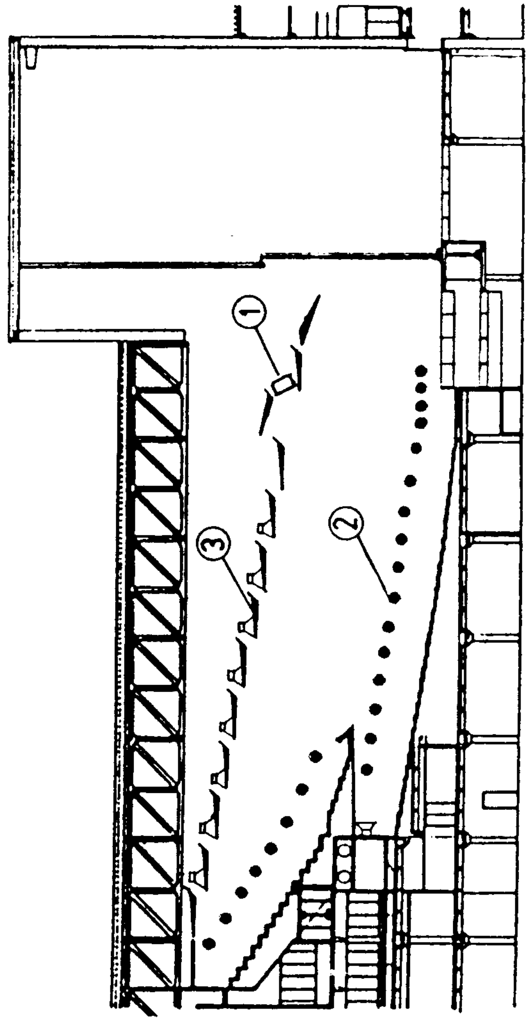

fig 10 LOCATION of THE LOUDSPEAKER SYSTEMS in THE BIG HALL of „SAVA CENTER" (cross section): ① FLYING SPEAKER ARRAYS, ② SURROUND LOUDSPEAKERS, ③ CEILING TRAVELLING SOUND LOUDSPEAKERS

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SOME TECHNICAL POSSIBILITIES OF USING THE TOTAL SURROUND
SOUND CONCEPT IN THE MOTION PICTURE TECHNOLOGY

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Some Technical Possibilities of Using the Total Surround Sound Concept in the Motion Picture Technology

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Abstract

This paper deals with a new concept of cinema theater which can be expected in the near future. It explains a possible concept of a 3-D Surround Sound based on the so-called Total Surround Sound system (TSS system).

The TSS system makes use of a specific disposition of sound sources (in form of a spatial network), and the 3-D sound itself is generated by means of a special processor, whose inputs are fed with six channel audio signals from the digital decoder within the projection system.

The TSS system could be used in all halls with projection systems and equipment for reproducing multichannel digital soundtracks (CDS and Dolby SR-D).

Some Technical Possibilities of Using the Total Surround Sound Concept in the Motion Picture Technology

Introduction

The Total Surround Sound System (TSS System - pats. pending) is a specific sound concept for generating 3-D sound in special acoustic ambiences.

The basic TSS configuration requires eight independent sound channels, whose signals are brought to corresponding inputs of a special audio processor (the TSS processor).

Signals processed by the TSS processor are then used via power amplifiers to drive sound groups, themselves placed so as to completely encircle the auditorium or audience. This creates a highly convincing sound impression with precise location of sound sources and possibilities of simulating movement of sound in space.

The TSS concept may be used for live performances in real time [3], in theater and multifunction halls, and in HDTV and movie projection halls and elsewhere.

Main Characteristics of the TSS System

Figure 1 illustrates the symbolic sound configuration within a specific acoustic ambience, most notably the eight independent loudspeaker groups.

The fundamental role of the TSS processor is to process the 8-channel sound. Management and distribution of sounds, as well as activating the so-called Travelling Sound Coprocessor, an addition to the main processor which produces the Travelling Sound effects (simulations of sound movements), is affected by means of control impulses, which are in our case recorded on a special control channel and as such are taken to the TSS processor.

This allows a multitude of 3-D effects to be generated within the sound field, with special processing of phases and amplitude of individual sound signals, this in turn enabling much creative research in the domain of psycho-acoustics to take place.

In itself, the TSS concept initially has numerous advantages over classic methods of sound generation insofar that the sound sources in its case are located along several sound planes. This eliminates the basic fault of monoplane sound systems, namely their lack of ambience response within an acoustic space.

Using sound sources in the floor and on the ceiling, we can define the height and depth coordinates within a specific ambience.

More sophisticated versions of the TSS processor are capable of driving individual arrays, or only parts of a loudspeaker group on the ceiling or in the floor. These matrix configurations, primarily intended for reproducing Travelling Sound effects (i.e. speed effects), can if required be used for stationary sound as well.

The three sound source groups, marked as Left, Rear and Right Surround on Figure 1, which encircle the audience along the horizontal plane, are primarily intended for reproducing stationary sound effects. If required, these sources could generate speed sound effects (Travelling Sound effects), which is to say that they too could use the matrix configuration.

References for more details on this are provided here.

TSS Concept Compatibility With Existing Standards

As noted above, the TSS concept requires using eight independent audio channels which after being processed by the TSS processor are used to create a 3-D sound impression within an acoustic ambience.

Latest standards assume six discrete channels in digital technology for left, center and right screen loudspeakers, for separate left and right Surround arrays, for the subwoofers and a special control track.

The first three channels are used mostly for dialogue and music, while the two Surround channels are used for special effects.

The idea we are proposing here is in conjunction with the two Surround channels, which could become carriers of sound contents to be used for generating 3-D sound in a hall. This is shown on Figure 2, itself modified in comparison with Figure 1.

By having two independent audio channels, we have the capability to distribute their contents not only as has been done so far, to side walls of halls, but also to floor and ceiling matrix loudspeakers. Other combinations are also possible, like using Mono Surround in certain parts of the movie, while simultaneously using the ceiling or floor configuration for reproducing a separate effect, or we could simultaneously use the left Surround and the ceiling matrix for an integral effect, while the right Surround and the floor configuration could produce a separate effect, and so forth.

In certain situations, we could also use the left or right Surround arrays and one (floor or ceiling) array to enable the so-called spatial dialogues (such as, say, a dialogue between an actor on the screen and an "invisible" partner located "somewhere" in the hall among the audience). By analogy, we could also produce highly suggestive music and other effects in this manner.

Sound sources illustrated as Floor and Ceiling groups are in fact matrix members driven by corresponding modular (card) amplifiers. Depending on acoustic properties, design goals and calculations, these matrix members could in fact consist of several loudspeakers. Each matrix amplifier possesses an address code controlled by the TSS processor, and the number of matrix amplifiers depends on the required resolution of the Travelling Sound effects and the software used.

The scanning speed of matrix arrays, i.e. of the total Travelling Sound effects, is controlled using driver impulses. The scanning itself could go by matrix lines or by matrix columns (Figure 3). Therefore, it becomes possible to generate a longitudinal rotation effect by using the main longitudinal axis of the hall to rotate the sound around it using floor arrays, right Surround arrays, ceiling loudspeakers and the left Surround array. In this context, the term "rotation" assumes successive scanning of above speaker arrays.

Similarly, we could produce a swinging effect by successively repeating the above, but without the ceiling loudspeakers. In that case, only one sound information channel is needed. The other channel could simultaneously be used for generating another sound effect using ceiling loudspeakers, and this other effect could be completely independent of the first effect.

Using phase control, and by phase synchronization of two adjacent sound configurations, we could obtain virtual sound sources which could then be moved to certain spatial coordinates.

Analogous effects could be obtained by changing signal amplitudes in two adjacent sound sources. This could enable simulations of various turbulent sound movements in space.

Actual placement of matrix sound sources on the ceiling should present no particular technical problems. As sources, one could use classic distributed loudspeakers for ceiling mounting, or the recently more used spheric loudspeakers [11].

Placement of floor loudspeakers could present certain problems. However, in this case, also depending on the configuration, acoustic properties and hall size, various solutions are feasible.

Ceiling and floor speakers as noted above are connected to modular (card) type power amplifiers, grouped in a rack. These amplifiers are of course controlled by the TSS processor. In designing future movie projection halls, all this would have to be taken into account and optimal solutions for practical introduction of floor speakers would be found.

Figure 4 illustrates the method of connecting existing digital multichannel reader (depending on the soundtrack format used) with the TSS processor so as to generate 3-D Surround Sound effects in a given hall.

Therefore, the TSS concept could be used in any cinema hall in which above adaptation works have been completed and which already contain any one of modern six channel cinema soundtrack systems. Of course, movies with 3-D Surround sound could also be used on purely "classic" equipment, but logically without the 3-D impression due to the lack of the TSS decoder.

The sound of rain, waves, thunder and storm, human steps and many more natural or artificial sounds reproduced on such a 3-D system benefit much in terms of realism as it exists only in the natural, real world.

The TSS Processor Concept

The structure of the TSS processor, in its cinema version, is given as a block diagram in Figure 4.

As is shown, decoder signals I/01, I/02 and I/03 are directly passed on to the equalization circuits (EQ1-EQ3) and from there to the amplifiers for three loudspeaker groups generating the direct (Screen) sound. The last input signal, I/06, is passed on to the synthesizer, and from there on to the subwoofer section.

The remaining two input signals, I/04 and I/05, are taken to the mixing sections, TS1-TS4, themselves driven by signals from the control circuit. This circuit, as shown in Figure 5, receives all information from the decoder control channel. In VCA-based mixing circuits, these two signals are balanced and then distributed. By passing these signals to equalization circuits and then on to the modular amps, the Travelling Sound effects are generated in a given acoustic ambience using channels driving floor, ceiling and side Surround loudspeaker groups.

Therefore, we are dealing with the basic effect of moving two audio signals through four spatially distributed channels.

The next equally interesting effect can be obtained by changing the microlocation of the sound source. This is achieved by virtual movement of the sound source. Auxiliary channel are equipped with a matrix of sound sources (ceiling, floor and sides of the hall). Each sound source (or matrix member) has its own modular amp preceded by a VCA circuit. This enables each independent source to have its own signal level regulation capabilities, affected by control logic signals.

The control logic extracts the required information from control channels. It first separates control signals from the equalization circuits. Since this is done using 16-bit signal which are turned into control signals using logarithmic steps, it may be assumed that the mixing "purity" is around 0.5%. Complete regulation of all four channels requires four 16-bit words (8 bytes).

Level regulation on modular amps need not have such fine regulation steps. It is sufficient to determine four levels only: OFF, -12 Db, -6 Db and 0 Db in terms of nominal levels. This means that for every modular amp (separate sound source in the matrix), one needs two bits of information in the control channel. Accordingly, the total amount of information consequently depends on matrix size.

Figure 6 illustrates the front view of the TSS processor.

Some Notes on Designing TSS Cinema Halls

In order to enable optimal sound coverage of the audience in a movie projection hall, and to satisfy optimal location of sound sources on basis of performed analysis, we have produced the required form of cinema halls, which need to have a gentle slope (of some 5%) so as to provide for the necessary visibility of the screen. Because of the ceiling loudspeakers, in this case it is necessary for the ceiling to follow the slope of the hall itself (this provides for constant ceiling-floor distance, required for optimal ceiling loudspeaker network design calculations).

If a cinema hall has galleries, then matrix networks of loudspeakers with different spacing in relation to ceiling speakers need to be installed, both in terms of the audience below the gallery and the audience on the gallery.

To obtain the optimal matrix speaker to audience distance, the relationship between hall height, length and width needs to be 1:2.5:3; side walls may gently widen by 5-10% from front to back of the hall.

Since the TSS system is basically an electroacoustic system for reproducing multichannel digital sound, the hall reverberation time should be as short as possible in order to minimize ambience influence, since a longer reverberation time will unfavorably influence speech intelligibility, music transparency and will prolong the duration of original spatial effects.

In medium sized TSS cinema halls (2,700 m^3 or 100,000 ft^3), it is recommended that the reverberation time for midrange frequencies (500 Hz) be from 0.45 to 0.7 sec. Frequency dependant reverberation time should be as linear as possible, with permissible increase of up to 30% at 100 Hz and a gentle rolloff above 8 Khz at 6 Db/Oct.

To eliminate harmful reflections, the back wall of the hall (as well as gallery walls) should use absorption materials.

Electro-acoustic calculations for three sound groups generating direct sound (Screen sound) and for the Subwoofer section may be performed using procedures defined in [1] and [5], while matrix groups on the ceiling and in the floor may be calculated using equations for distributed sound systems [6]. The disposition of the Surround arrays on side walls have also been defined in [7].

Direct sound sources should use powerful and efficient loudspeaker systems with constant directivity, also using corresponding Triple Chamber subwoofer units for extremely low frequency reproduction.

The walls and ceiling of the hall should use efficient sound sources with a directivity characteristic of $60^o \times 60^o$, while the floor units should be less efficient with a directivity characteristic of $80^o \times 80^o$. Floor speakers could also be located as illustrated in Figure 7.

Calculations of sound source placement, with corresponding 3-D display of auditorium coverage, could use various available software packages often noted in professional literature. These programs enable the required calculations of halls using the TSS concept to be made, with reverberation time analysis, early reflections, impulse responses, etc, to be taken into account.

Figure 8 illustrates a detail of a TSS hall.

Some Notes on Multichannel Cinema Sound

In the above, we have taken a look at a practical technical implementation of a system for reproducing 3-D Surround Sound in a cinema hall.

Practical use of the TSS concept in multichannel cinema sound assumes a considerably greater use of sound contents intended for two channel Surround in view of the fact that these 2 channels are being enlarged to 4 loudspeaker groups. In this new concept, the Surround Sound channels would not as until now play the role of a passive supplier of stationary sound effects, activated in corresponding movie scenes. Quite to the contrary, they would be almost equal to the direct sound reproduction channels, since a large number of ambiophonic sounds could be implemented using 3-D Surround. This would need to take into account optimal distribution as well, or in other words, a proper balance of integrated 3-D sound.

Such a concept enables considerably more precise defining of the sound ambience and a much higher fidelity of the reproduced sound; in effect, this concept implements the principle that ***the sound is where we see it or where we assume its source is.***

Actual implementation of final multichannel Soundtrack makes mandatory use of Digital Audio Workstations, since they enable excellent precision and work efficiency.

Recording of control impulses which organize and define the entire 3-D setup will rely on a special software package, which must also contain a graphic presentation of loudspeaker group status from Figure 2.

The sound studios where the final processing of movie soundtrack is done would have to be adapted in the sense of introducing control audio monitors, which alongside side Surround arrays should also have matrix monitors as well [4].

Finally, it should be stated that the TSS concept demands a certain change of current artistic standards and order prescribed by directors and executive producers in modern movie industry, since implementing of this concept would also mean abandoning a fairly long lasting traditionalist cinema sound, which had thus far more or less simply kept company and indirectly illustrated the picture itself.

Conclusion

New technical solutions used in multichannel cinema sound herald significant changes in the technological sophistication of motion picture art. The 3-D Surround concept illustrated here could become another contribution to that common goal.

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Captions:

Figure 1: A symbolic spatial TSS network

Figure 2: Modified TSS network in the Motion Picture version

Figure 3: Scanning directions in the loudspeakers matrix configurations

Figure 4: The functional block diagram of TSS playback system

Figure 5: A detail of function diagram of TSS processor

Figure 6: The frontal view of the TSS processor

Figure 7: A version of floor loudspeaker's position

Figure 8: An artistic view of a TSS Motion Picture hall

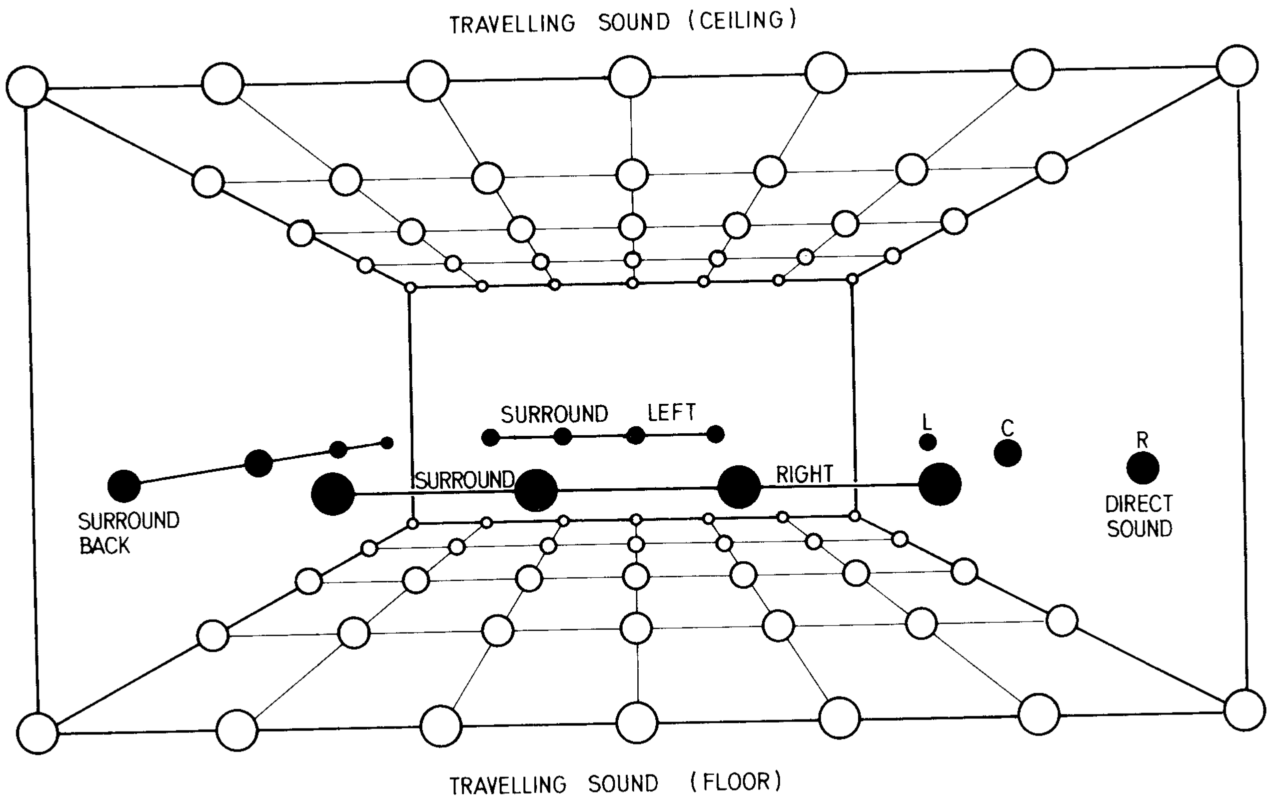

fig. 1

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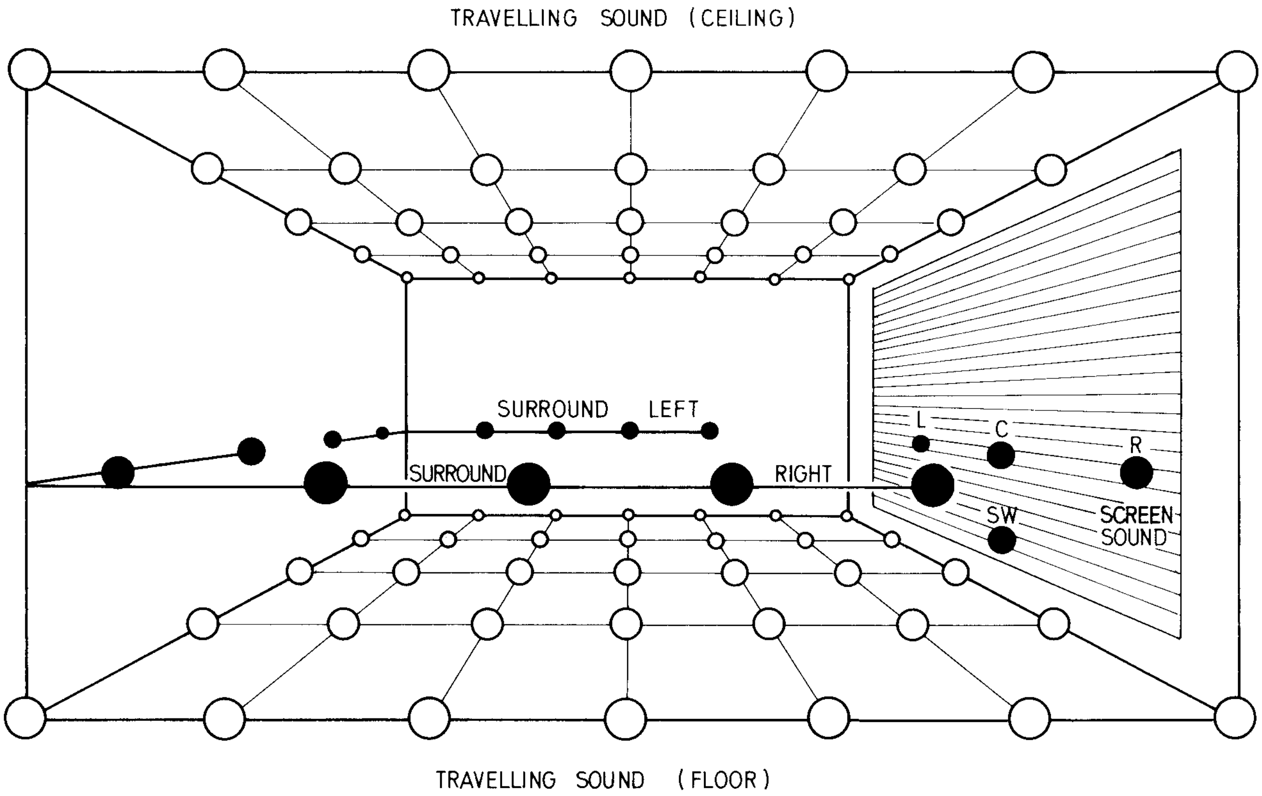

fig. 2

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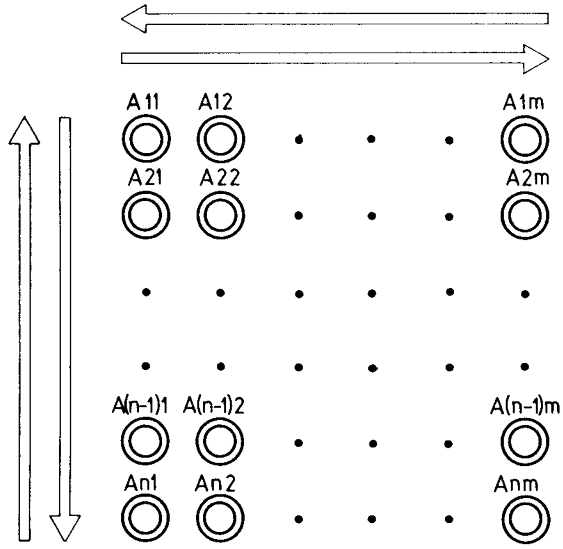

fig. 3

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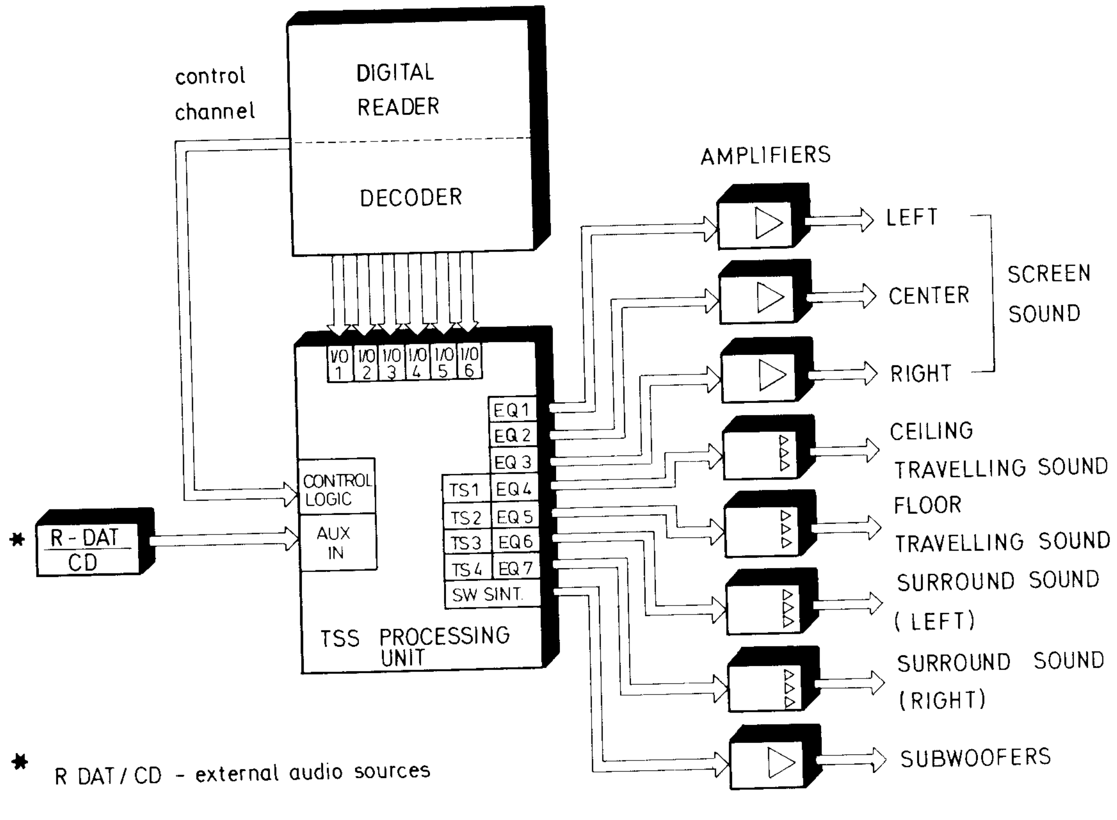

fig. 4

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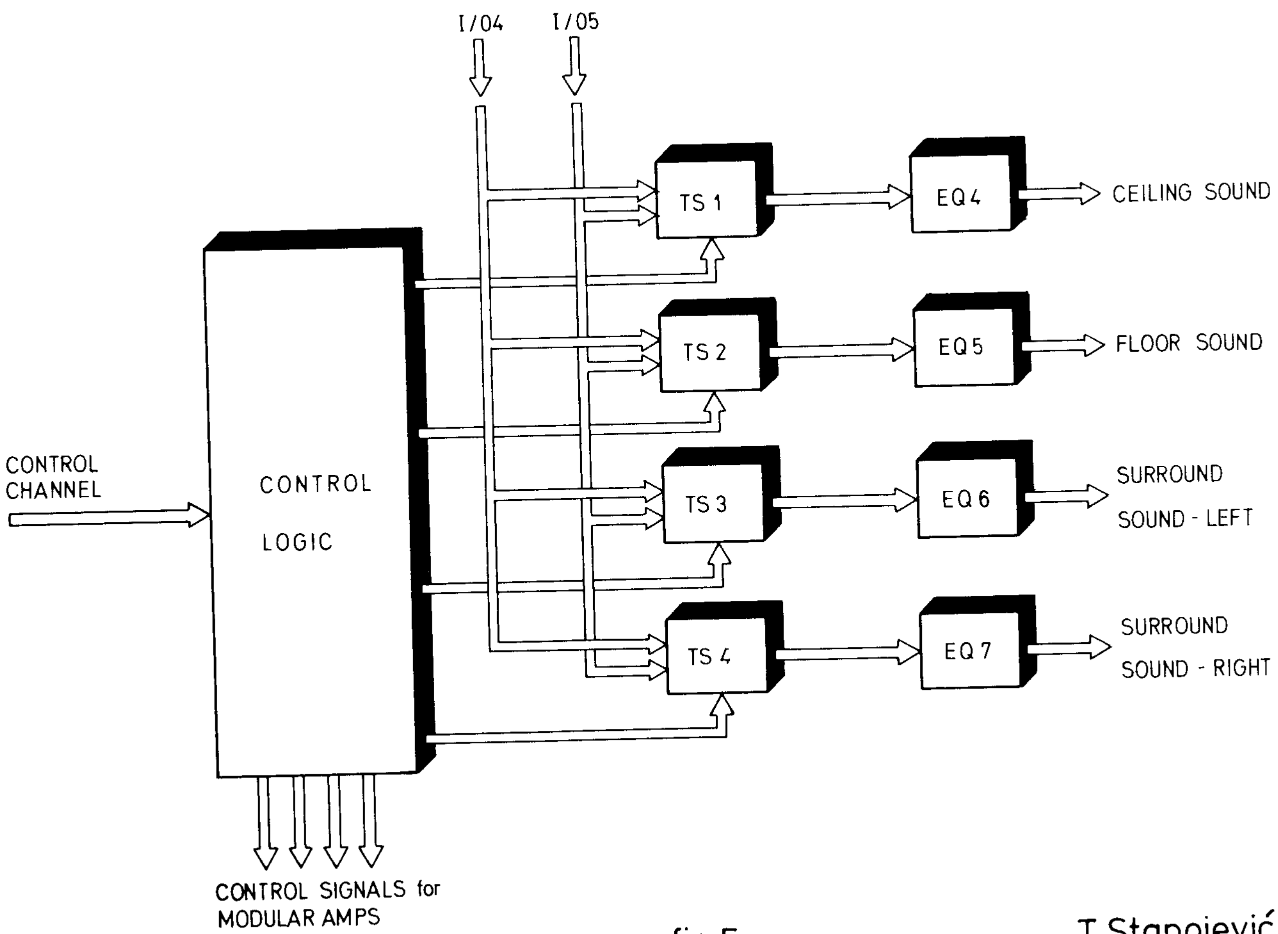

fig. 5

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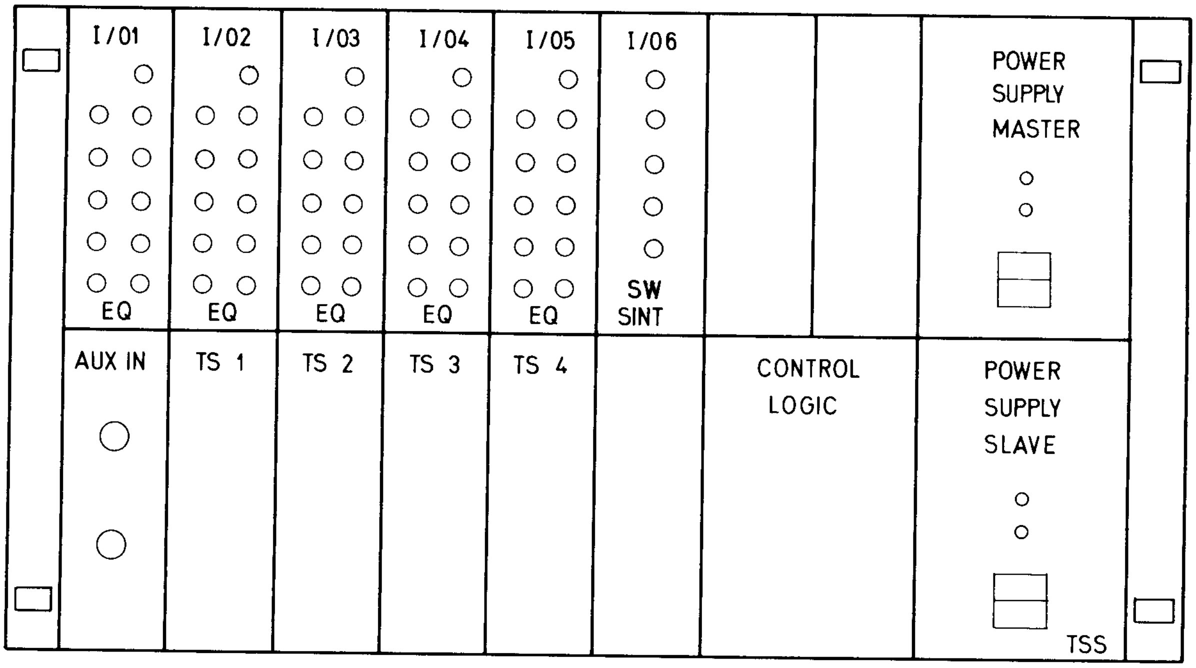

fig. 6

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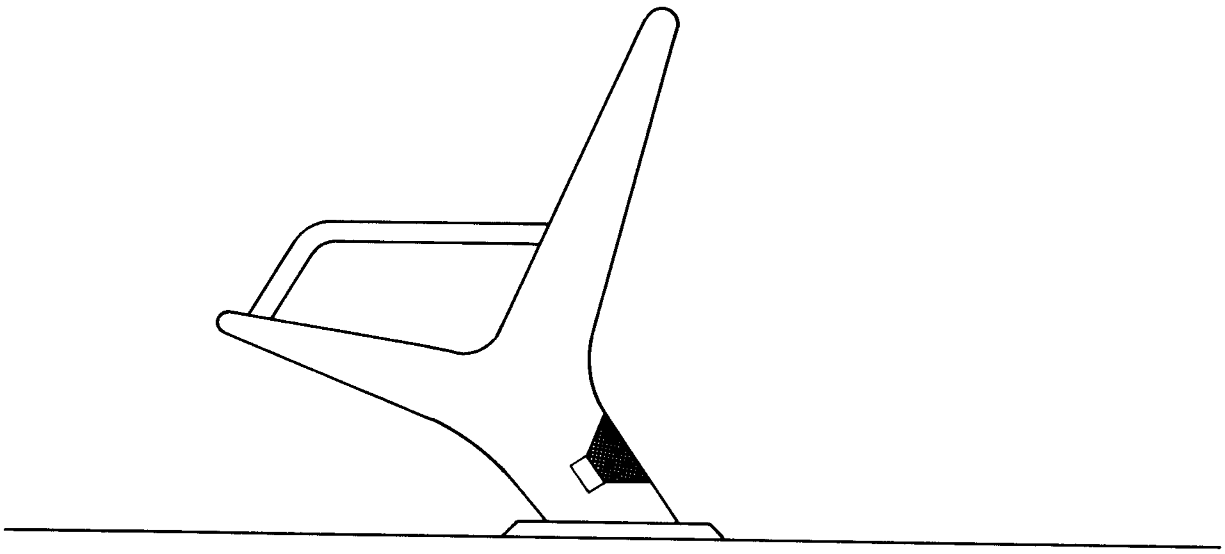

fig. 7

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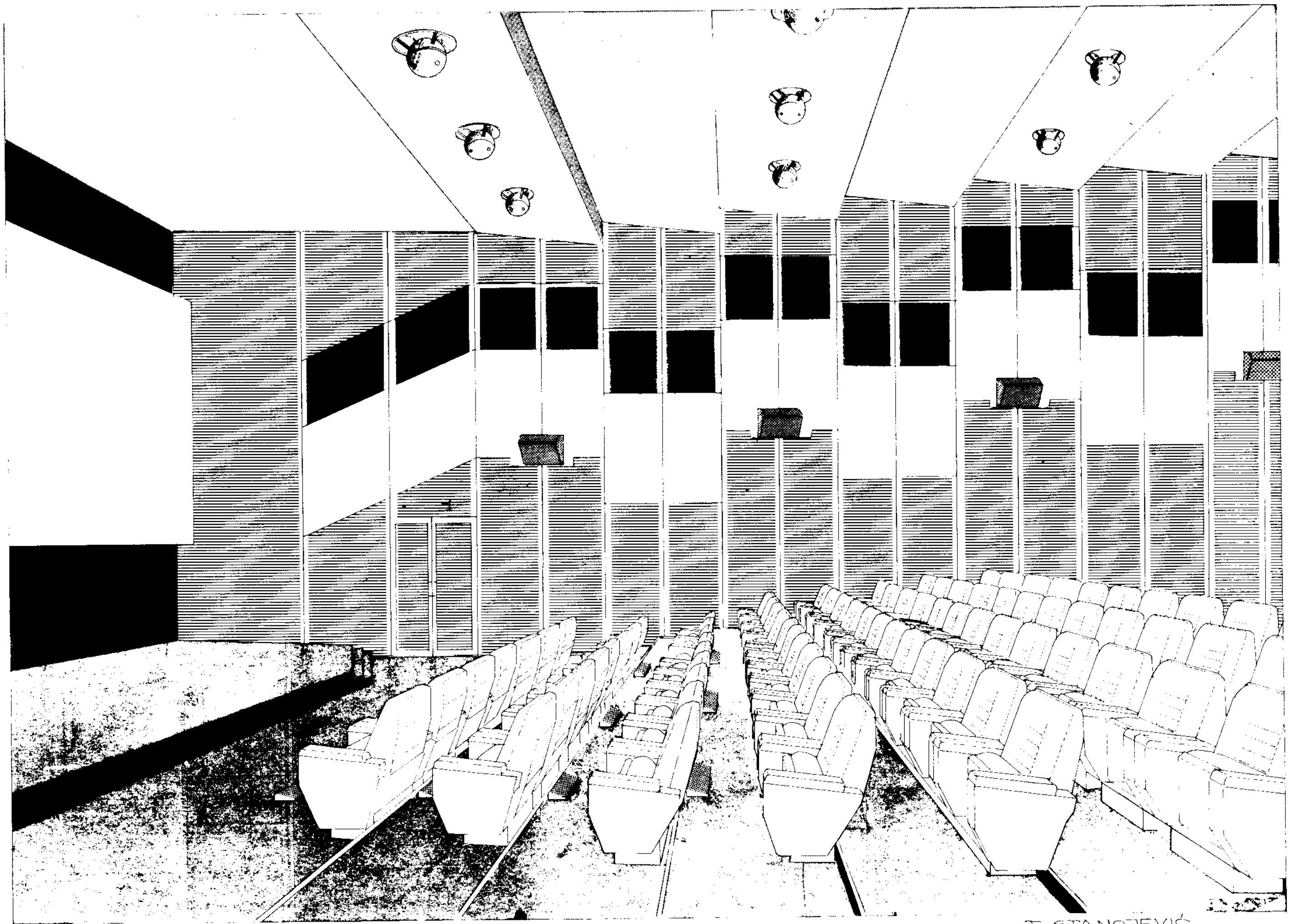

FIG. 8

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