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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 — $\Theta h \times \Theta v = 120^{o} \times 60^{o}$
- 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:

$$Rc = (h - 1)\ tg\ \Theta/2 = 6.2\ m \qquad /4/$$

Hall height h =12 m, listening plane height from floor 1 = 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:

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

listening plane height from the floor 1 = 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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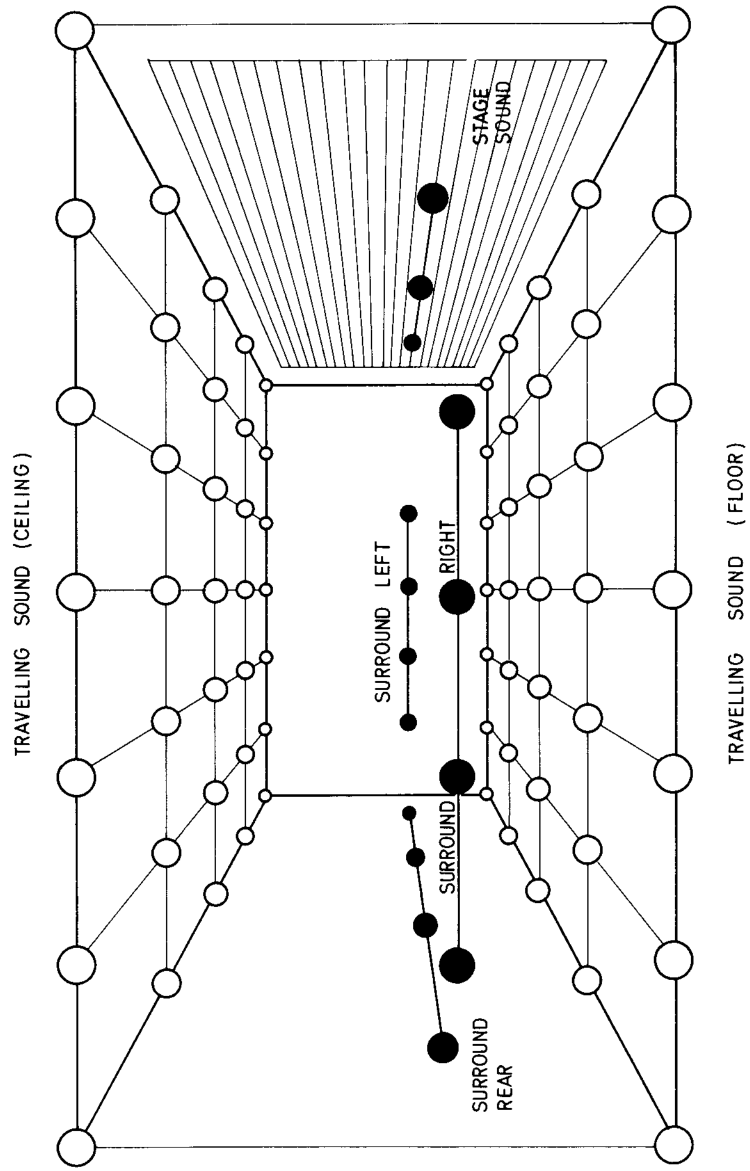

fig. 1 SPACE NETWORK IN A TSS HALL

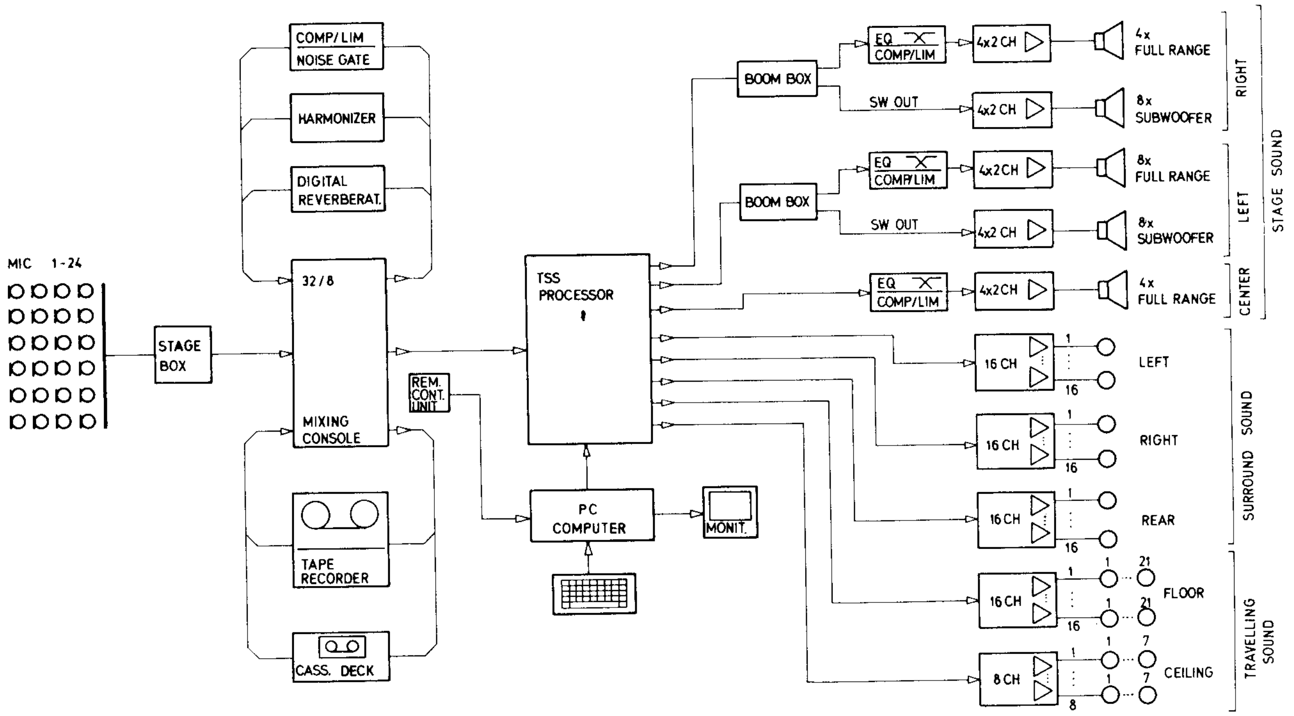

fig 2 BLOCK DIAGRAM of THE TSS SOUND REINFORCEMENT SYSTEM for LIVE PERFORMANCE

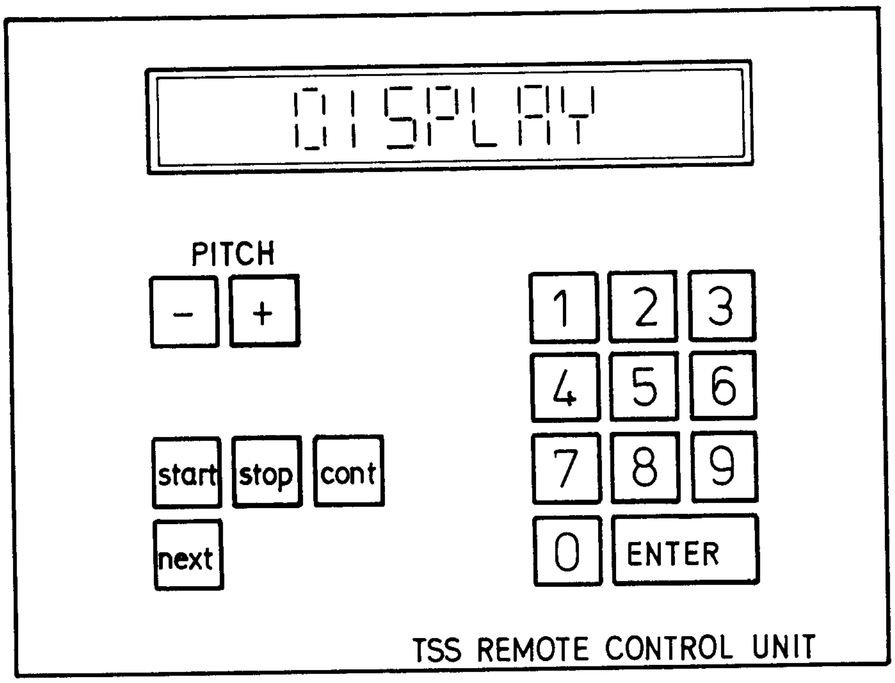

fig.3 VIEW of THE TSS REMOTE CONTROL UNIT

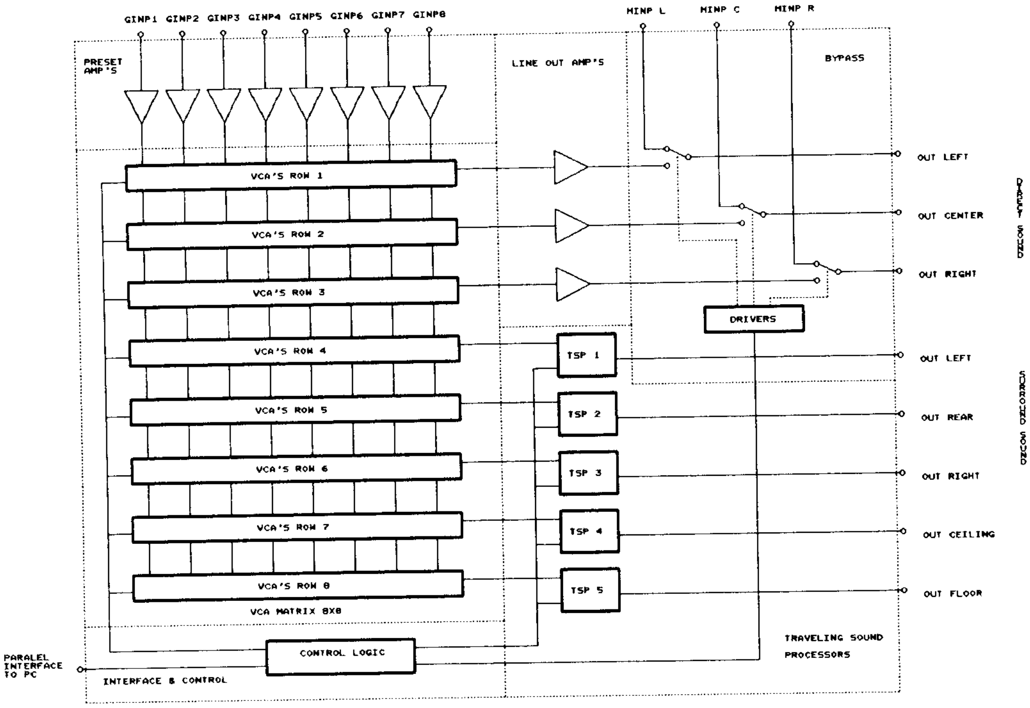

Fig. 4. TSS PROCESSOR STRUCTURE

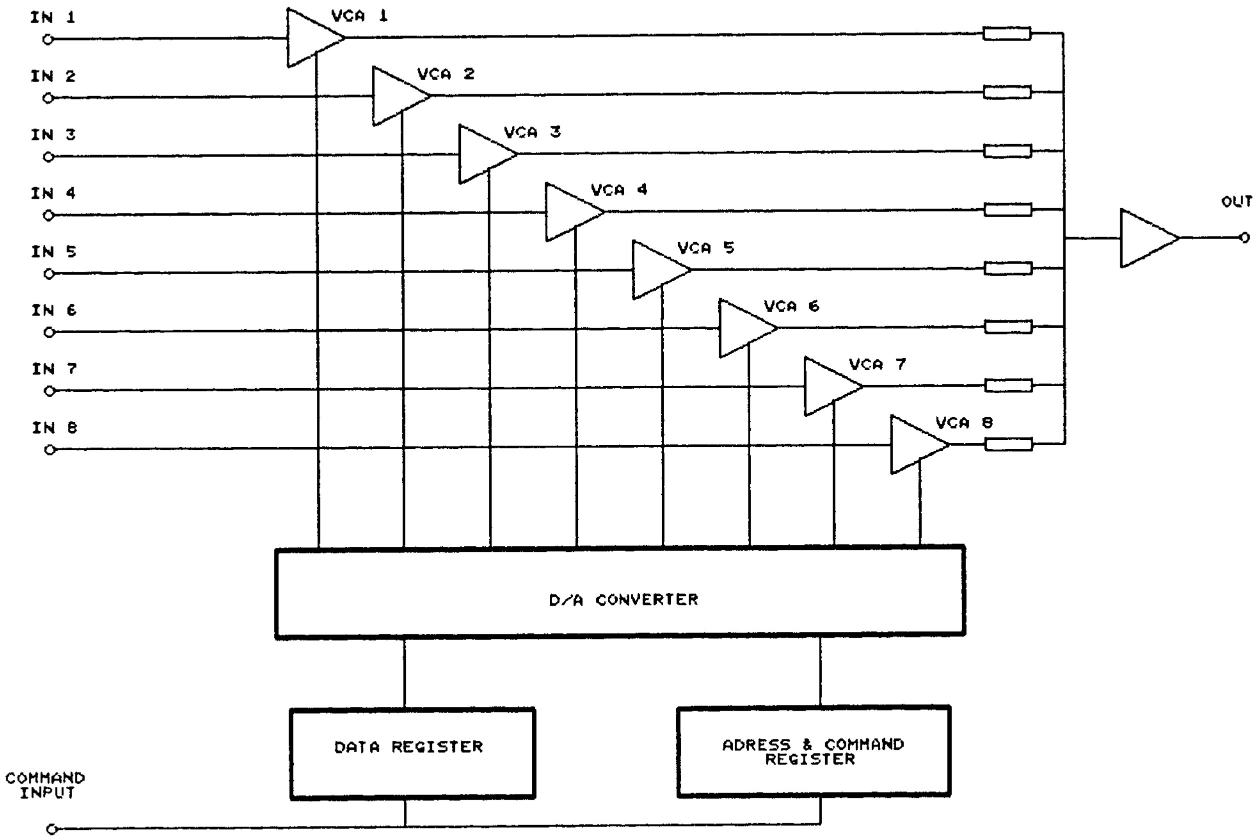

Fig. 5. PART OF VCA MATRIX - VCA'S ROW

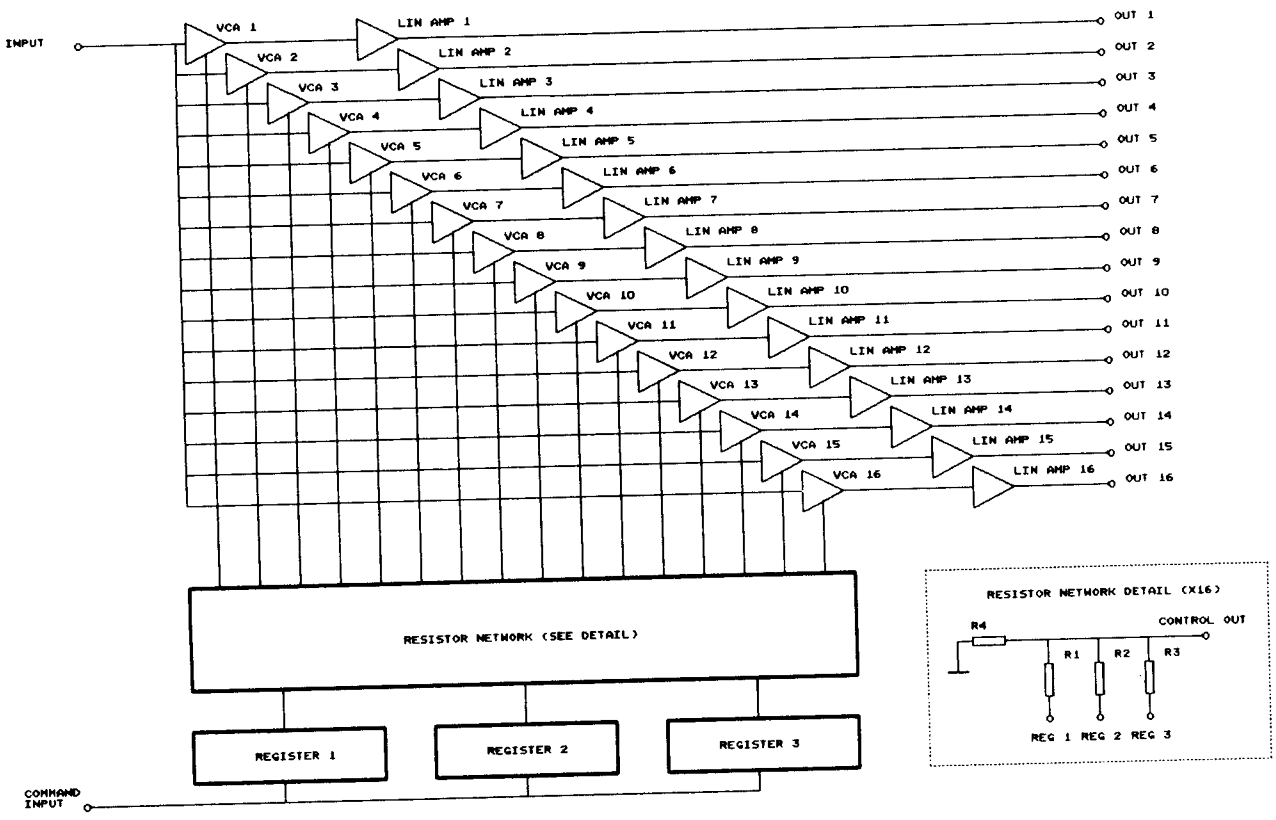

Fig.6. TRAVELLING SOUND PROCESSOR

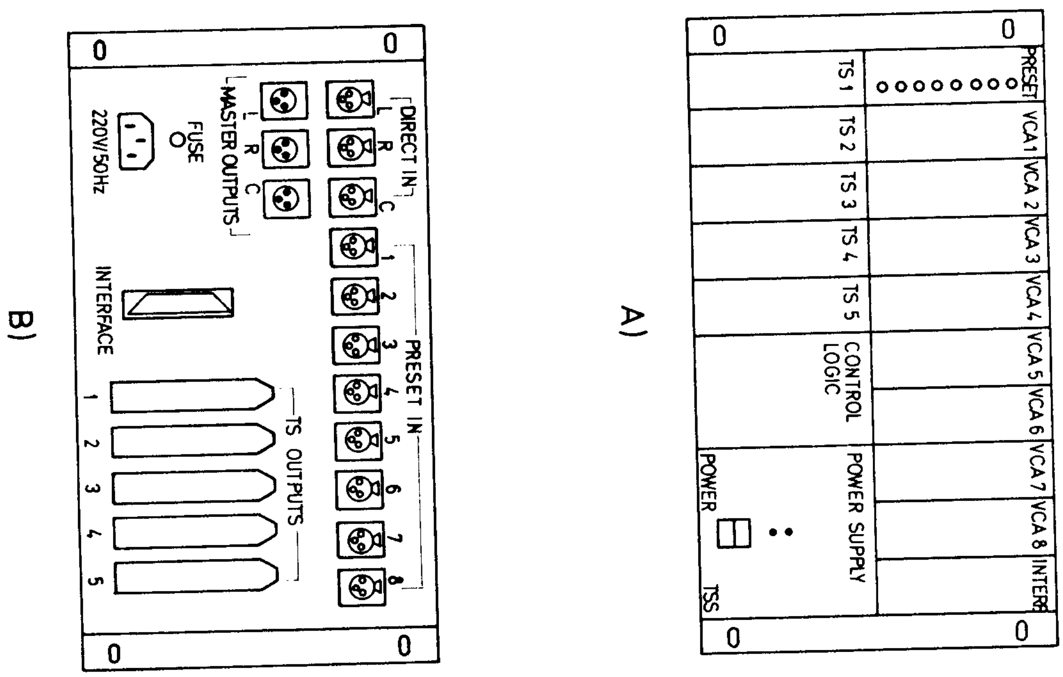

fig.7 VIEW of THE TSS PROCESSOR
A) FRONT, B) REAR

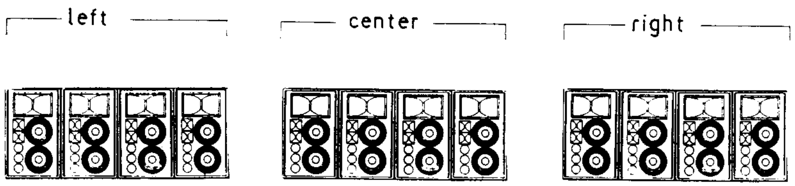

fig.8 VIEW of THE FLYING SPEAKER ARRAYS for STAGE SOUND

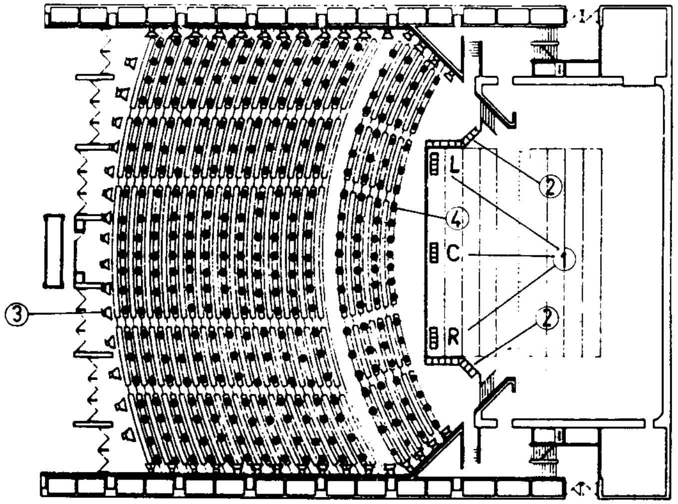

fig.9 LOCATION of THE LOUDSPEAKER SYSTEMS in THE BIG HALL of „SAVA CENTER"(plan): ① FLYING SPEAKER ARRAYS, ② SUBWOOFER SECTION, ③ SURROUND SPEAKERS, ④ FLOOR TRAVELLING SOUND LOUDSPEAKERS

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.

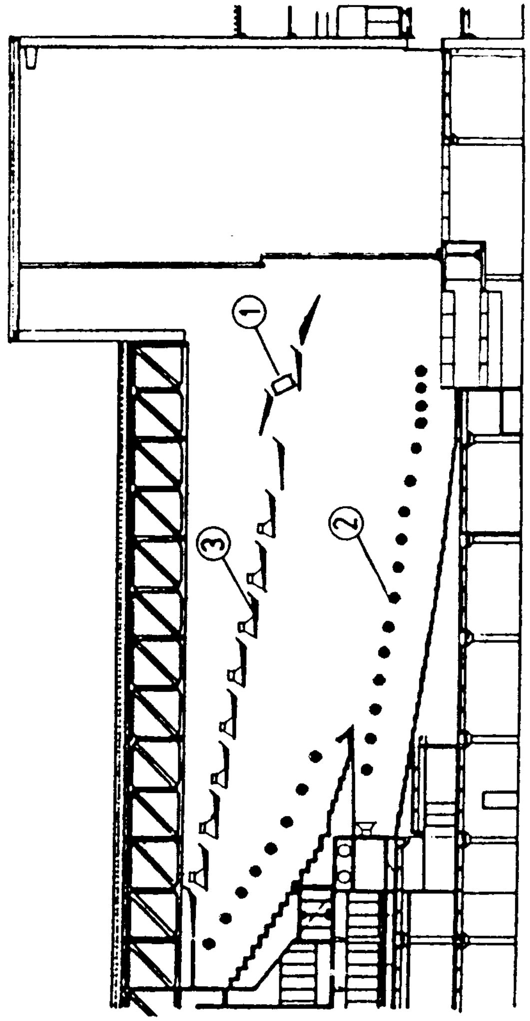

fig 10 LOCATION of THE LOUDSPEAKER SYSTEMS in THE BIG HALL of „SAVA CENTER" (cross section): ① FLYING SPEAKER ARRAYS, ② SURROUND LOUDSPEAKERS, ③ CEILING TRAVELLING SOUND LOUDSPEAKERS