The Basic Principles of I.S.S.R. (Individual Sound Source Reinforcement) 3413 (K-7)

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THE BASIC PRINCIPLES OF I.S.S.R.
(Individual Sound Source Reinforcement)

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ABSTRACT : The spacial radiation characteristics of a sound source and its interaction with a particular acoustic environment are important components in the perception of a real sound source "object". This paper describes the main aims of the research programme "I.S.S.R.", together with the application of its basic principles to three important fields of audio activity: live concert recording and sound reinforcement, studio recording of electroacoustic sources, and the recording and reproduction of MIDI controlled systems.

INTRODUCTION

Over the last thirty years we have seen a gradual evolution of microphone techniques applied to music sound recording ; multimicrophone and near-field microphone techniques have come to dominate the scene both in studio recordings and in live concerts with sound reinforcement. Inevitably we have seen at the same time an evolution in the range, complexity and sophistication of the techniques used in signal processing, to compensate for frequency response and timbre problems created by these close-miking techniques. The sound mixing desk and its associated auxiliary equipment are capable of a high degree of signal processing in the anological domain, and even more so in the digital domain, with the advent of digital signal processing in this type of equipment.

Nowadays, the sound recording engineer has some very powerful tools at his disposal for creating the desired frequency response from almost any type of sound source, on condition that parts of the sound spectrum are not missing altogether. However there is still a considerable amount of work to be done in collaboration with research workers in the field of musical instrument acoustics. In using "close-miking" techniques one needs an intimate knowledge of the close-range radiation characteristics of specific instruments to be able to obtain rapidly the desired tonal balance. Much of the present knowledge of this subject is confined to the realms of "experience" or "accepted practice".

However, although we have become very powerful in the manipulation of timbre, we have almost lost sight of any spacial representation of the source. In fact we have to return to very "primitive" (some would say "pure") microphone techniques to be able to reproduce natural acoustic sound source size and its associated spacial environment. It would seem that we are not yet able to manipulate or generate artificially these two fundamental characteristics of good sound recording given the present state of our knowledge of Digital Signal Processing.

This lack of acoustic dimension and real acoustic environment are particularly evident in three important fields of audio activity:

- live concert sound reinforcement and recording,
- studio recording of electroacoustic sources,
- recording and reproduction of MIDI controlled systems.

In the case of a live concert, sound reinforcement is generally designed to give a good overall sound distribution for the maximum number of spectators at quite often very high power levels, without any intention of creating the impression of a spacially structured acoustic source.

The only noteable advance in this direction, in recent years, is the research and development of the system called "Delta Stereophony" by Professor Ahnert (Ref. 1). Here the impression of localisation of individual sound sources is considerably improved for the great majority of spectators in a hall, using individual sound source reinforcement with standard loudspeaker enclosures as near as possible to the original sound source positions, with the necessary delayed Intensity and Time Difference signals being fed to the power reinforcement system.

It is possible that the ideas concerning the "Musical Instrument" Loudspeaker Enclosure developed later in this present paper could improve the "character" of the "on stage" sound sources used in Delta Stereophony, however interaction with the acoustic environment still needs a lot more consideration.

The recording of a live concert with sound reinforcement and the studio recording of electroacoustical sound sources is generally achieved by direct feeds to the mixing desk. Therefore the only means of achieving a localised sound image is by Intensity Difference "pan potted" signals. The information corresponding to the acoustic structure of each individual sound source is inexistent and the acoustic environment can only be produced by artificial reverberation. The unique nature of early reflections produced by individual sound sources is impossible to generate given the present state of the art in artificial reverberation units.

The recording and the reproduction of MIDI controlled systems suffer from the same difficulties concerning the acoustic sound source dimension and the sound source environment.

BASIC PRINCIPLES OF INDIVIDUAL SOUND SOURCE REINFORCEMENT

For each individual primary sound source, a real electroacoustic reinforcement or secondary source must be created having the same spacial characteristics as the primary sound source and radiating the same timbre distribution. In other words we must be able to create sound sources having the same radiation characteristics as the original sound sources.

Sound reinforcement is perhaps the most difficult context in which to apply this technique. The position of the secondary sound sources must to some extent reflect the position of the musicians on the stage, with all the inherent problems of acoustic feedback through microphones on the acoustic sources and through some pick-up devices on other instruments. However new microphone transducer techniques are gradually becoming available, often within the instrument structure itself, that can help improve both feedback thresholds and frequency balance of the signal. Electronic protection against feedback is also becoming much more efficient.

In the recording of a live concert using normal sound reinforcement techniques, the secondary sources must be located in another acoustic environment isolated from the main concert hall. In this case the recording procedure is simply a matter of using the secondary sources to create a new sound stage and choosing one's prefered stereophonic sound recording microphone system (Ref. 2) to produce the desired results. The flexibilty and quality obtained by this procedure surpasses any present recording context. All signal processing either for frequency correction or special effects has to be carried out within each individual sound source circuit.

The studio context is a simplification of the previous situation, the sound engineer having complete freedom to choose his acoustic environment and then set up the electroacoustic sources to satisfy his needs.

The specification of each secondary electroacoustic source is obviously based on a sound knowledge of the radiation characteristics of each individual musical instrument. There still remains a considerable amount of research to be done in the acoustics of musical instruments before we have an overall view of the radiation characteristics of the majority of commonly encountered musical instruments.

THE SECONDARY ELECTROACOUSTICAL SOURCE

There are a number of possible approaches to creating these electroacoustic sources :

1) Construction of a radiating enclosure using methods very similar to those used in the construction of an actual musical instrument, the enclosure surfaces themselves contributing to the overall radiation characteristic.

2) Division of the frequency band into subbands, each subband a specific directivity pattern, individual enclosures being created to cover each subband with the required directivity pattern.

3) Multifacet/multitransducer source with individual computer controlled drivers to create controlled directivity and frequency source over the whole audio range (Ref. 3).

A "MUSICAL INSTRUMENT" LOUDSPEAKER ENCLOSURE

For over twenty years, the manufacturers of loudspeakers and loudspeaker systems have been able to obtain considerable control over the technical characteristics of these transducers, even more so with present day computer simulated optimisation. The research into the correlation between the objective characteristics of an electroacoustic system and the subjective impresion of the listener has also enabled development of products that correspond very closely to the needs of a particular market.

Two specific aims in the research and development programmes of electroacoustic systems are very much in evidence :

1) The Hi-Fi oriented product, with a considerable amount of research into consumer taste, and the resulting development of products to satisfy this market.

2) The professional product used as the ultimate tool in controlling the quality of recorded or transmitted sound information.

We are a long way from the days when some 'enlightened' researchers considered that the loudspeaker cabinet should itself be likened to a musical instrument in order to reproduce the same. Indeed the coloration and directivity patterns of these types of enclosures are now known to be counter productive in the search for fidelity or neutral reproduction. However the world has turned full circle, as it is exactly this type of enclosure that is needed to reproduce, as a secondary source, certain types of musical instrument.

Even in the early days, research into this type of enclosure was "few and far between". In France one such étude was produced in 1977 by A.Mas of the Laboratoire d'Acoustique de l'Université de ParisVI under the direction of Professor Leipp (Ref. 4). The cabinet construction was inspired entirely by the techniques used in the construction of a violin (figs 1 and 2). It is not surprising that the reproduction of stringed instruments was considered "interesting" at the time, but as a general purpose loudspeaker enclosure, opinions were very much divided.

The same could be said for the loudspeaker enclosures developed by "Charlin" in the sixties. Some of these cabinets were in the form of vertical standing columns and reproduced advantageously the excellent organ recordings made by Charlin himself.

It is clear that the development of the "musical instrument loudspeaker enclosure" is almost the work of an musical instrument maker or "luthier/facteur".

INDIVIDUAL SUBBAND DIRECTIVTY SYSTEM

The main function of this type of secondary electroacoustic source, is to create the correct directivity pattern over the whole audio range. The timbre or frequency response characteristic is dependent on the sound pick-up technique used and can easily be controlled by electronic signal processing. The complex radiation pattern of an instrument must be divided into specific subbands, each subband having a relatively simple model pole structure (Figure 3 : monopole, dipole, quadripole etc) corresponding to its basic directivity pattern in that subband, and each subband is reproduced by a specific loudspeaker enclosure with the corresponding directivity pattern.

MULTIFACET/MULTIDRIVER COMPUTER CONTROLLED SOURCE

Certain types of musical instrument, the violin for instance, present a highly complex radiation structure. It would seem in this case, that the only solution to creating an equivalent secondary source would be to adopt the computer controlled multifacet/multidriver source. However before adopting this complex and highly sophisticated method, it is necessary to be sure that the ear is indeed receptive to all the aspects of this complex acoustic structure. It is highly probable that generalisations can be made as to radiation patterns within certain wider frequency bands due to masking effects of the more predominant lower harmonic response over the finer structure of the higher harmonics.

THE FIRST EXPERIMENTAL STEPS IN PRATICAL APPLICATION OF I.S.S.R.

The recording and reproduction of MIDI controlled systems is the environment that is the most flexible in which to try out this technique. As it will take some time to develop adequate secondary sound sources for most of our needs, standard loudspeaker enclosures can be used for the initial experimentation.

MIDI equipment must be used that can give separate output signals of each instrument. The use of a MIDI drum machine with individual instrument outputs, is an excellent illustration of this type of approach. Each output is amplified separately, and fed to its respective loudspeaker system. There are three initial source configurations that enable one to control two basic spacial characteristics.

1) The loudspeakers are grouped in a circular arc around either the listener or the stereophonic sound recording microphone system. The result obtained is only remarkable in its lack of interest! No impression of sound perspective or spaciousness is perceived.

2) Whilst keeping the same orientation, the loudspeakers are positioned at different distances from the listener or microphone system, similar to the normal position adopted by the musicians. The feeling of depth or sound perspective will become apparent. This is obviously due to different ratios of direct to reverberant sound for each sound source.

3) If the orientation of the loudspeaker cabinets is now changed to correspond with the approximate general direction of radiation of the particular instrument being reproduced by each source, a new feeling of space associated with early reflected information will become apparent.

The next step is to replace each individual source by its electroacoustic "clone". It is the intention of the I.S.S.R. research programme to gradually develop electroacoustic "clones" for each family of musical instruments.

CONCLUSION

The impression of reality of the sound image created by sound reinforcement, the recording of live rock and jazz concerts, and the studio recording of MIDI controlled equipment, can be considerably improved by the use of electroacoustic sound sources that are "clones" of the original acoustic sound sources. This enables a new sound stage to be created with all the character of the original acoustic sound.

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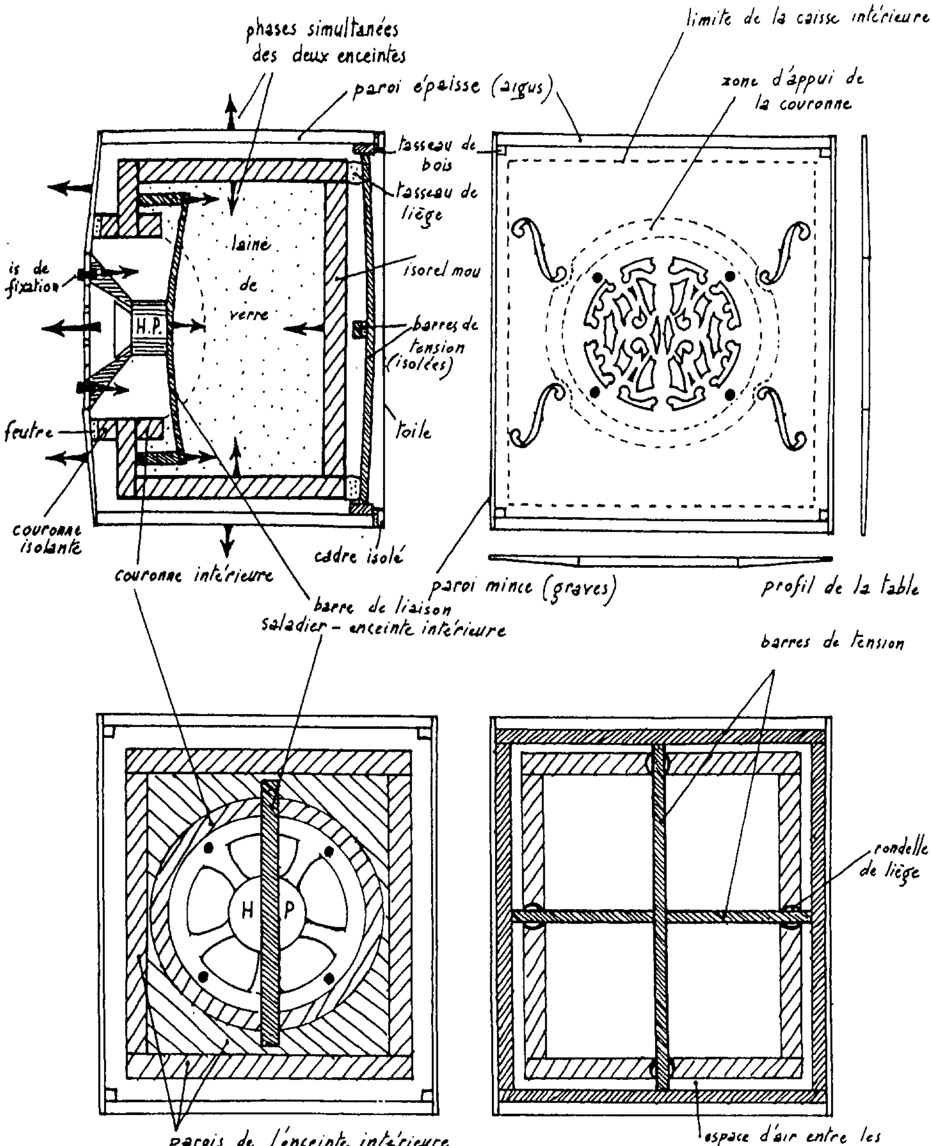

FIGURE 1 -"Enceinte Acoustique Instrumentale" (page 15 of Bulletin 89 du Groupe d'Acoustique Musicale, Université de Paris VI.)

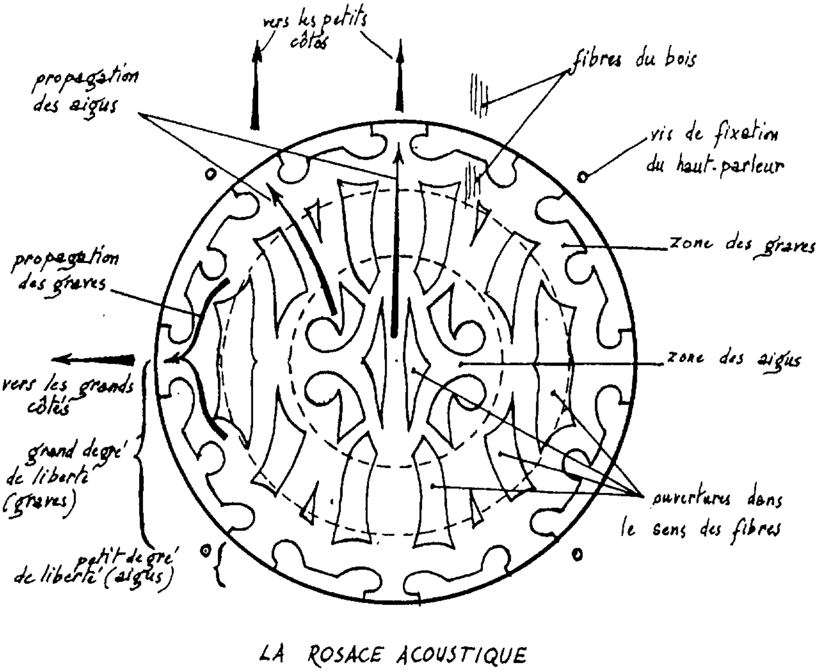

FIGURE 2 - "La Rosace Acoustique de l'Enceinte Acoustique Instrumentale" (page 16 of Bulletin 89 du Groupe d'Acoustique Musicale

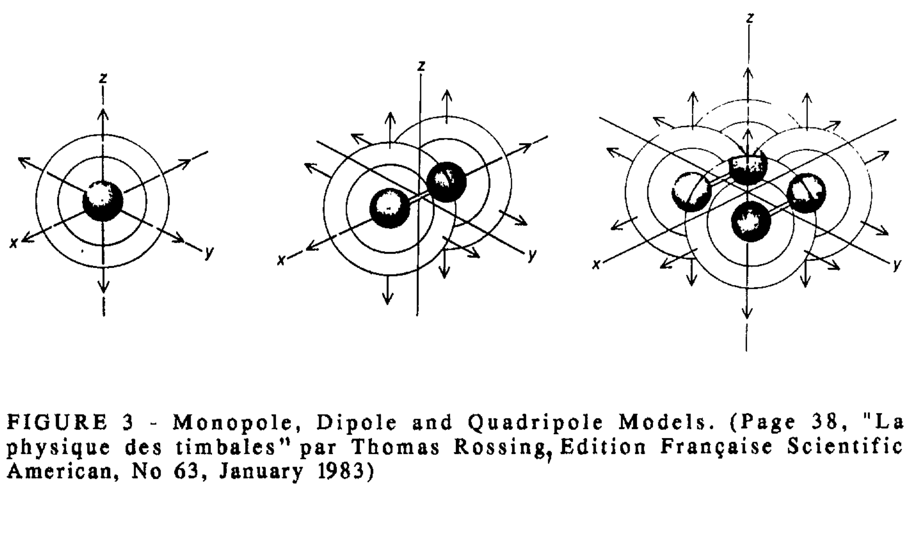

FIGURE 3 - Monopole, Dipole and Quadripole Models. (Page 38, "La physique des timbales" par Thomas Rossing, Edition Française Scientific American, No 63, January 1983)