5

ELECTROACOUSTIC FIELDS IN LARGE OUTDOOR SPACES. REQUIREMENTS FOR MUSICAL REPRODUCTION FOR LARGE AUDIENCE

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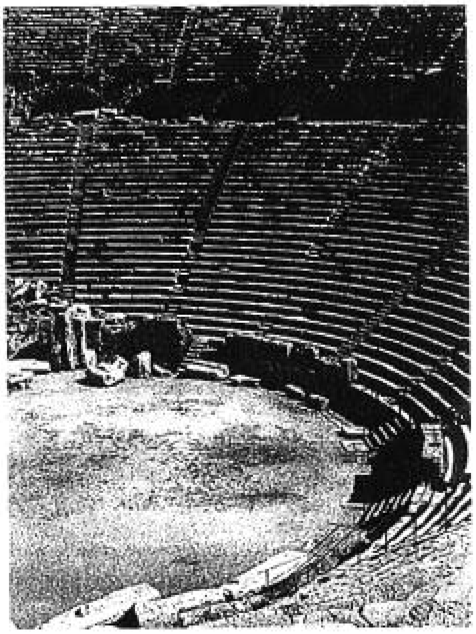
1. Introduction

First of all, we would like to thank the Foundation Ramón Areces and the Spanish Socie of Acoustics for the opportunity they give us to share this podium with such distinguishe international personalities in the field of acoustics, as these lecturing here in the occasio of this anniversary.

When Prof. Lara proposed to us to talk about this topic we thought it was a goo idea since it is a very current phenomena that has grown in interest during the last 30 years.

Let us remember that recently the 2nd Woodstock Festival was held for an audience of over 200.000 people outdoors and at Los Angeles at Dodger Stadium, this summer, three fantastic tenors and a very well known conductor got around them 60.000 people in a very large stadium with a final audience of 800 millions people through television.

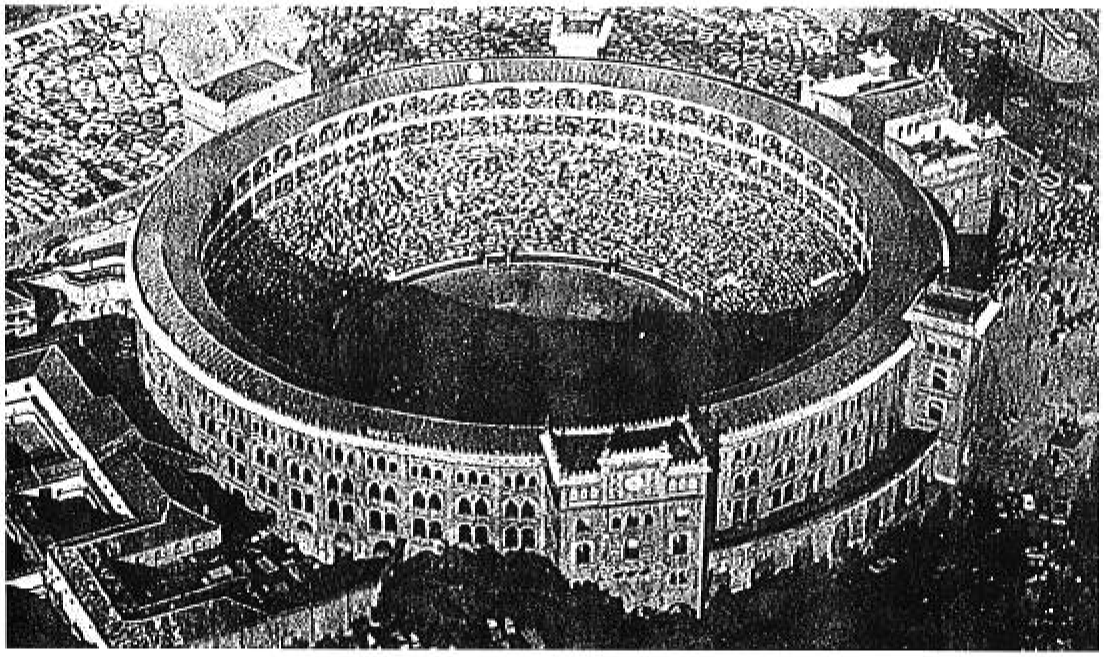
In Spain, these outdoors events, happen all over our country. We don't have large spaces similar to the American bowls, therefore we use more Mediterranean spaces, whose origin goes back to the entertainment arenas of Greek roman civilization (circus-theatre). We are referring to the bullfighting rings and their adaptation for other "fights" such as modern music concert (folk, pop and rock music). Let us remember that in the 65's the Beatles in the European tour, played at Las Ventas bull-fighting in Madrid, with a capacity of 20.000 people.


Aspendos, a agreek-roman arena

In the preceding papers it has been presented very rigorously the relationship among the objective parameters that may characterize a concert hall, the type of music played, and the subjective impressions of the listeners, precisely by the scientists that have given us a greatest data along this research line and whose works are obligatory reference.

The optimum design of a concert hall must be done without any electroacoustic help what implies limited dimension and volume, designed to conveniently use all the energy emitted by the orchestra or the singers. Therefore, the musical manifestation should limit the audience to some hundred individuals, who although actively listening to the concert, participate in a passive way in it. The images that they see are the orchestra and those brought on by music, and in the case of operas, the action that take place in it. In that sense, Wagner can be considered a precursor since he has aimed to revive the ideal Greek drama. A drama that should be a ritual with the participation of everybody through the "Gesämmtkunstwerk" (integral art) unifying the dramatic poem, music, staging and dancing in a show that was thought to be the real justification of Gluck theories based, in turn, on the Camerata Fiorentina and lastly in the ideas of Esquiles and the Greeks of the V century b.c.

In outdoor concerts of modern music, practically everything is different. The public which participate actively in a show designed to be a real symbiosis of the visual and auditory images where performers as well as spectators need to be immersed in sound and vibrations. Simultaneously, visual information is presented in large videoscreens trying to produce a communicational catharsis among spectators and playing musicians.



Madrid bullring "Las Ventas", 20,000 people

Obviously, the idea is to emit with enough energy a sound that should directly arrive to an audience of thousand of spectators, avoiding in general any sort of lateral or rear reflections.

The impression of stereophony, so basic to the listeners of concert halls, in our case, it is handled by the sound engineers that rule the electroacoustic system from a previously fixed point in the place.

In the musical aspect, not even Berlioz, who loved orchestral experimentation, perhaps the precursor of quadrophony with his work "La grande messe des morts" (he distributed the orchestra into 4 sections) could have not thought of the constructive (or destructive) effects that electroacoustic can bring about, such as we see in the concerts of Pink Floyd, who use a quadraphonic reproduction system.

The range of frequencies of modern music extends a decade below that of classical music spectra. Also, if in classical music the average level ranges between 80 and 90 dB, in the case of modern music, they exceed by far the 100 dB, with peaks that may reach up to 120 dB.

2. Requirements for powerful outdoors sound systems

The most large systems of sound for outdoors have gone from being large clusters on the stage, to a radiating wall at the back of the stage as used by the "Grateful Dead" group in the 70's with excellent acoustic results.

Currently the most usual lay out of the transducer system is two large groups of sound sources in an array form located on both sides of the stage, in a typical lay out of stereophonic reproduction. Sometimes, satellite clusters on the right and left of the above mentioned devices are used for reinforcement, but oriented in the direction of the audience which is located in the arena or in tribunes. In areas of difficult sound reinforcement, like amphitheatres it is possible resort to satellite clusters that are time delayed and directed to the public.

These transducers are the basic elements of the complex electroacoustic chain, therefore one should focus primarily on the parameters and the basic characteristics required by them. To understand this in the best possible way we will go from the most simple, which is the acoustic box, (and even each one of the loudspeakers), to the most complex element, the alignments or arrays.

Every unit of the emiter system requires:

- capacity to emit a great power, which relies on a high electroacoustic efficiency,
- a broad range of frequency response, normally from 30Hz to 16 kHz,
- a directivity which should be adequate to cover the audience area.

3. Frequency response

In previous paragraphs we have seen that for live reproduction of rock music, an due to the type of instruments used, bass-guitar, synthesizers etc., the frequency spectrum should go from 20 Hz to 16 kHz. The final goal of the electroacoustic reinforcement is its ability to reproduce, in the most natural possible way, the frequency range that we have mentioned, and specially the region from 400 Hz to 7 kHz, which is closely linked to attributes of clarity, intelligibility and articulation.

Currently, the use of synthesized musical instruments extend the conventional frequency range, demanding the use of adequate reproduction systems. For this reason, it is customary to resort to the use of 4 ways: sub-buffers, buffers, medium and high frequency. The transportability of the systems force them to be much smaller, but maintaining and even increasing the acoustic output power with good electroacoustic sensitivities (>98 dB re 1 Vat/1m). Some critiques on new technologies say that these new devices "scream out" messages instead of trying to get a more intimate reproduction.

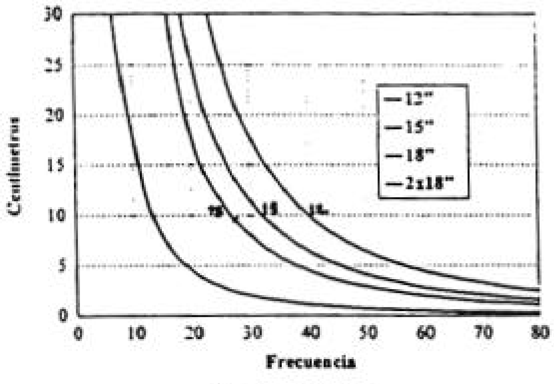
Most of these units are autonomous, and they include the audioamplifier (that can be controlled by a main computer), the active filters and the necessary time delayers.

For the bass or woofers it is in general necessary to use more than one loudspeaker. If the desired SPL in function of the frequency is known, it is possible, through elementary mathematics to get the excursion of volume necessary for the sub-woofer.

The next graph shows the theoretical peak excursion which is necessary to obtain a level of 110 dB for different loudspeakers, treated as piston of 12, 15 and 18 inches of diameter. For an 18 inch loudspeaker emitting a frequency of 20Hz, a displacement of +10 mm is needed to produce 110 dB in its axis.

This very large excursion can be reduced using loudspeakers of a greater diameter, increasing their number or using the low frequency boost, due to the reflections on the ground.

Displacemen



Frequency Hz

As for the subjective impression of the harmonic distortion in the frequencies range on a sub-woofer (10-100 Hz), we can say that it has been scarcely investigated. Scott and Axon, in 1957, studied the modulation amplitude of the signal reproduction above 50 Hz, and they concluded that the auditory threshold for harmonic distortion was 8% or higher. Fielder in 1988 studied the increase of the auditory threshold, by masking, due to signals located in the first critical band of the auditory system, finding that the distortion perception at the lower frequencies is more sensitive at relatively low level than in the higher ones as it is summed up in this table, based on the data of this author.

	2°	3°	4°	5°	THD
110 dB					
20 Hz	4,9	1,6	0,7	0,4	5,3
50 Hz	2,8	1,1	0,8	0,5	3,2
100 Hz	2,5	1,4	1,1	0,9	3,2
100 dB					
20 Hz	4	1,2	0,5	0,3	4,2
50 Hz	2	0,9	0,6	0,4	2,3
100 Hz	3,2	1,8	1,2	0,9	3,9
80 dB					
20 Hz	6	1,6	0,7	0,4	6,5
50 Hz	1,4	0,5	0,2	0,2	1,5
100 Hz	1,6	0,7	0,4	0,2	1,8

Loudspeakers are very inefficient devices. The maximum efficiency of the cone for direct radiation is as low as 5% and only horn loaded systems reach between 10 and 40%.

Consequently in the case of woofer and sub-woofers, 95 % of the power driven to the loudspeakers becomes caloric energy, mainly in the coil, and it must be dissipated by the system.

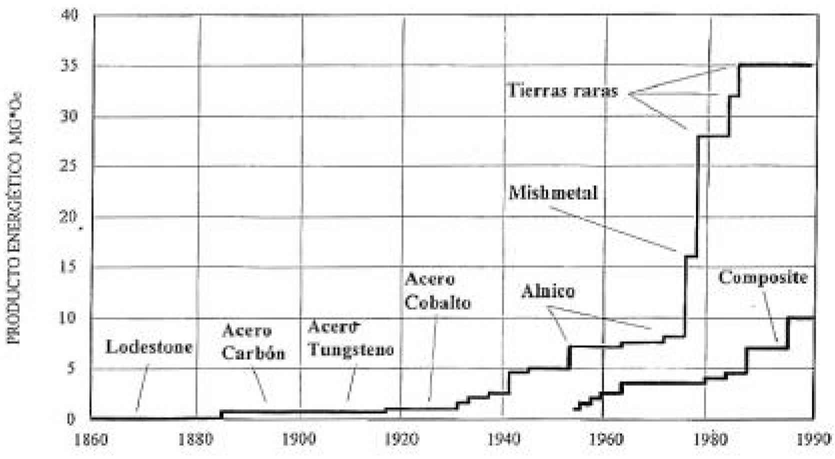
The mechanisms that linked thermal parameters and nominal power handling capacity of loudspeakers, are well studied, including equivalent thermal circuit models with lumped elements by Henricksen (1987) or other more simplified, in function of the resistance and capacitance of the coil and the structure of the magnet.

From the graph of the heating curves, it is seen that the most sensitive element of the circuit is the coil; its thermal capacitance is low, therefore it warms up very quickly. It is possible to verify that the coil temperature of a woofer in a steady stay can reach of 150 °C for an electrical power from 150 to 200W. The maximum safe operating temperature is 300°C in order to calculate the maximum admissible power. To reduce the influence of heating on the transducer (that is to say on the coil and the magnetic circuit) every designer has his own philosophy, although there are commonly accepted aspects such as :

- Larger coils are cooled down before the smaller ones, (which increases the high and the diameter of the coils
- Thicker top plates in the magnetic circuit implies a better conduction of the coil.
- Aluminium coils, that heat up more slowly than copper coils, give better response to transitory signals.
- In the vented boxes, the air movement in the pipe is used to cool down the coil motor set.

This combination of effect is the so called power compression that can bring about an attenuation of up to 6 dB in the radiated acoustic pressure level.

Evolution of permanent magnets



On the other hand, it may be interesting to see the evolution of these magnetic structures that form the permanent magnets of the loudspeakers, since the strength applied to their membranes or diaphragms will linearly depend on them.

So since the twenties, when there were energy products of 1MG*Oe, we have move to the actual 4/10 MG*Oe updated with stroncion ferrites or alnic devices.

Other alloys like the Samario-Cobalt, give energy levels of 30 MG*Oe and although this material is very expensive, it is essential when magnets should be small like microphones, velocity sensors, etc.

Recently, new compound materials are used, based on neodimio-iron-boron (NdFeB) that give "energy products" up to 35 MG*Oe.

The possibilities of applications of these new magnets, that are made of "rare earth", are immense in composites, and they could bring about the polymeric extrusion of magnets as motors for the drivers. Of course, the composites will reduce their energy product to 6MG*Oe.

With bigger magnetic fields, the dimensions of the magnet can be reduced for the same density of flow. The reduction in weight and mainly in volume, allow to integrate in the high frequency exponential horn, two or more motors increasing the SPL. This the number of components in the unit and avoid the possible interferences due to creation of acoustic dipoles or cuadripoles

The weight of these units of NdFeB is reduced by a factor of 3 (as compared to the ferrites) which is very important when you consider the metal structure underneath the stage.

4 Unitary equalization of sources

The equalization of every acoustic box, has as a goal for the whole assembly to give a flat pressure response in the desired frequency renge, as well as a phase response without any sudden steps. The acoustic centres of all loudspeakers should be well aligned, otherwise there will be a comb filter effect that can give rise to very important level losses in that range of frequencies .The sound engineers call this "microalignement".

These effects should be taken into account when designing the cross-over filters in a complex box including several loudspeakers. An acoustic box made up by two or more transducers will have better quality as to the frequency response and phase if it is biamplified or multiamplified with the introduction of active filters, instead of using only passive cross-over filters connected to a single power amplifier. The reason is that when the power amplifier receives an overload clipping in the range of low frequencies, series of harmonics are generated due to the crests cut down, which are easily eliminated by the band pass filters of the multiamplifying system since it has a higher filtering slope.

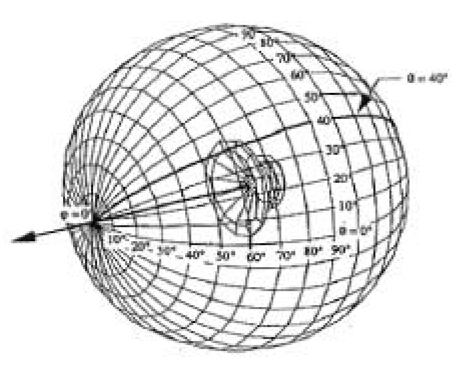
The cross-over filter of each unitsystem should be designed carefully so to achieve a "minimum phase" system, which can be described as the one that the pole and zeros of their transfer function are located in the left semispace on the complex plane representations of the phase-amplitude. Another objective of these filters design is to accommodate the delay time of each one of the loudspeakers that make up the acoustic area so as to align the respective acoustic centres.

If for each one of these systems or for all of them, it is desired a given response, flat in frequency and phase, adaptative filters can be used, based on 18 bits analog-digital converters and vice-versa, controlled by a microprocessor.

Today, there is a great profusion of these intelligent baffles, programmed to act in a pre-established dynamic frequency range that avoids breakages, decreases, clipping distortion and holding steady the responses in amplitude and phase.

5 Arrays

In the sound reinforcement of large spaces, it is normally necessary great "penetration" and also a wide distribution. This requires several high powered columns together with numerous units of subwoofers with frequencies below 50Hz. This required a special transport and handling of the different units (greater flexibility) having in mind the directivity, which has to be highly studied. The mid-woofer normally have an angle of 70° and the highs unit 30° angle in the horizontal plane.



When putting together a certain number of loudspeakers, in order to increment the emitted acoustic power or sending this energy in preferential directions there are other problems which are consequence of modifying the directivity of the group of radiating sources.

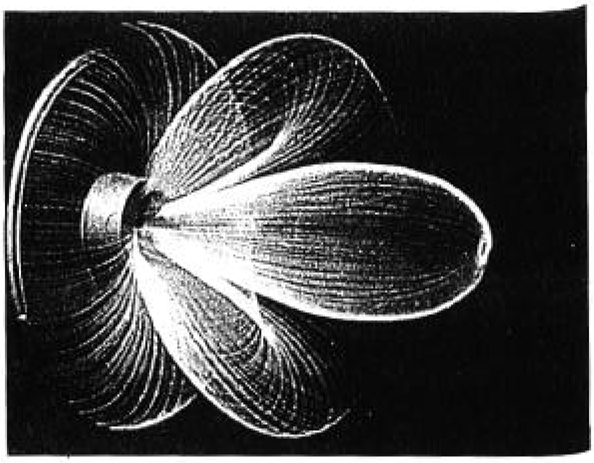
The directivity of the acoustic source can be reponsible of a low level speech intelligibility or of uneven musical perceived frequency-response, if echoes or very delayed reflections are present.

The techniques of foreseeing the directivity in the far field of arrays start from the sum of contributions of the individual sources, the amplitudes and phases of the electrical signals applied to each source, as well as the distance between these, are the determinative factors of the directivity.

In general, a predictive calculation starts with a consideration of individual sources with spherical radiation, as in the figure when radiates a frequency whose wavelength is several orders of magnitude greater than its effective diameter as a piston.

As the frequency emitted by the hypothetic loudspeaker increases, the emitter becomes more directive, concentrating its spatial energy in lobes, as see in figure.

If all the sources in the array are identical, this problem for calculation can be easily upset by applying the principle of diagram multiplication; that is to say, considering the array as composed of individual sources and multiplying the directivity function of the unitary source.



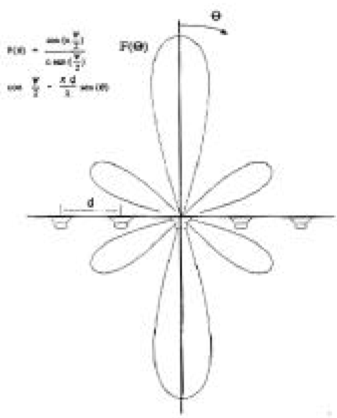
The directive behaviour of the arrays, is well known from decades in the electroacoustic laboratories. Nevertheless "sound engineers" have normally worked empirically, and, late at the beginning of the nineties, they brought up the "old classical works" with techniques based on optics antenna technologies and sonar systems.

$$F(\vartheta)=\frac{\sin n\frac{\psi}{2}}{n\sin\frac{\psi}{2}}$$

with $\frac{\psi}{2}=\frac{\pi d}{\lambda}\sin\vartheta$

Minima (zeroes)

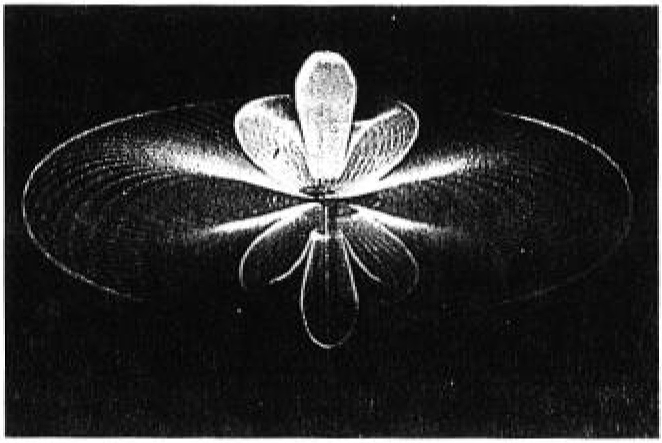
$$\sin n\frac{\psi}{2}=0\rightarrow v\pi\ldots\ v=1,2,\ldots,n-1$$



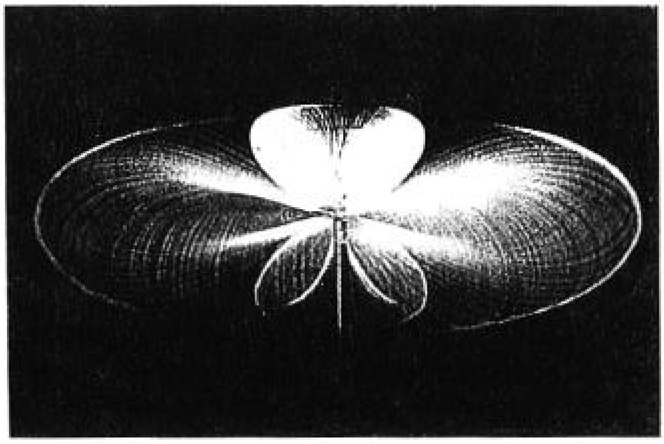
These figures show some procedures on directivity control of different type of arrays which were carried out, at our Institute of Acoustics, some years ago.

First of all, we present a simulation of a spatial emission of two arrays, first without lobes, situated at 90 ° of the principal radiation axis, and the 2nd one should correspond to an array called "uniform" of five elements. In the first photo, the relation of d/λ is greater than 1, and "d" the separation between sources (supposed to be identical for all elements) and in the second one is less than unity.

In both cases, it can seen that the radiation solid has a circular symmetry on the array axis, and preferential directions of radiation, with the maximum radiation lobe at 90° coinciding with the revolution symmetry of each loudspeakers. If the individual sources were directive, with loss of radiation energy in the back, the revolution symmetry

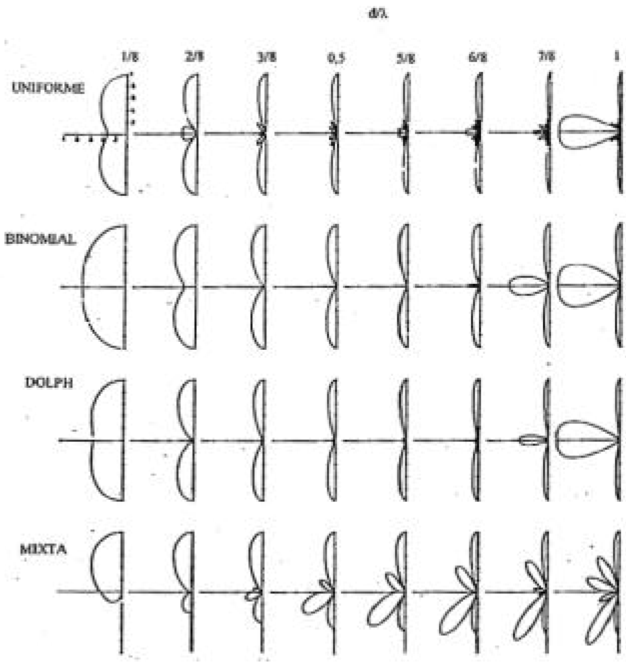


according to the axis that join them would be lost, with a notable loss of the energy radiated in the rear semispace.



To find a properly relation between amplitudes and phases and between each one of the sources in an array, we the electroacousticians have come to certain propriety of functions or known series. From simple uniform arrays, with classical equations, to arrays with amplitude distribution following binomial series (their fundamental propriety being the annulling of the secondary lobes for values of $d/\lambda<1$), or the polynomes of Tchebitcheff used by Dolph to control the secondary lobes amplitude.

This figure is an example to show the evolution of the directivity patterns depending of d/λ. In all them and because of the transcendental function that integrates the algorithm of the directivities, for multiple integer of d/λ.there is a lobe at 90 whose amplitude is the same that the one of the main lobe. As d/λ increases, the secondary lobe is displaced toward the direction of the principal one.

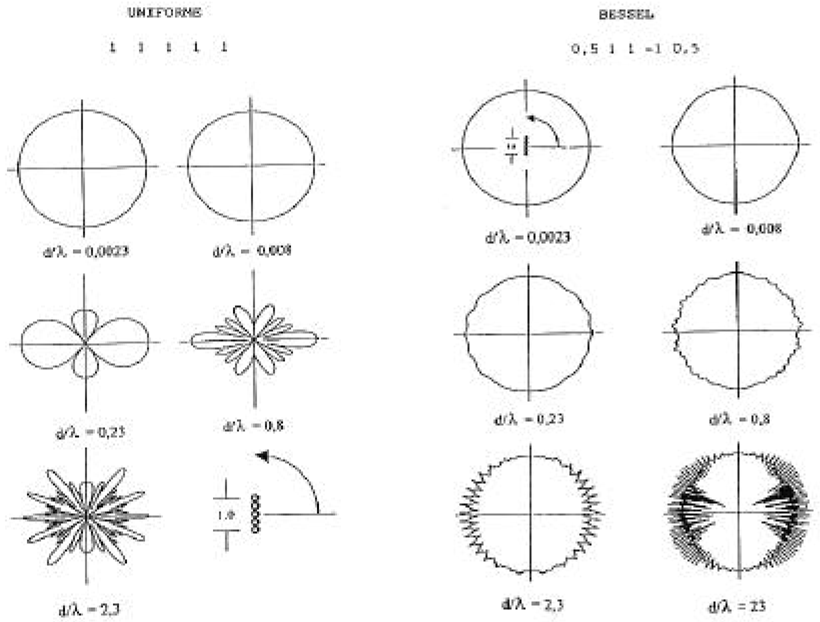


Therefore we can see that when we try to sound reinforce an arena, with only one array, we can choose between different types of arrays in order to radiate in preferent directions.

Up to now we are not realice that in all these array there is a delay of 180° between the adjacent radiation lobes. For certain areas, we have two lobes with an opposing phase coming from each one of the arrays situated at the two extremes of the stage, there would be destructive interferences in the frequencies corresponding to the directivity patterns, and music would have uneven sound.

Recently, in the last 3 or 4 years, the electro-acousticians, to offset this problem of desphasing by linear array, that basically consist in to pile up acoustic boxes with identical electric signal, they are using the so-called "Bessel Arrays" which were introduced by Franssen in 1983 when building these acoustic towers. As the name indicates, these are based in the Bessel functions properties of first class and "n" order. Combining the properties of these functions with the equations which give the magnitude of the acoustic pressure in the reception point the total directivity of the groupal sources is approximately the same to the individual transducers of the configuration, extending then margin of operativity of the array in frequency to regions where the length of the array is a high number of wavelength.

This implies a cost, because the emitted energy is lower than that an equivalent uniform array, since in the serie of corresponding numbers to the distribution amplitude, one or several takes a negative coefficient and what means that one (in case of a 5 loudspeakers array) or several (for higher order arrays) would radiate in counterphase.



This figure shows the comparison between a uniform array and the Bessel equivalent for different relations d/λ ,in the case of 5 aligned sources.

We can therefore create some directivity patterns as directivity balloons, like the so popular "rugby" very appreciated by technicians, especially in the case of two dimensional arrays. We should add that, currently, there are computer programs for calculating these arrays.

We believe that, with a careful relation of d/λ, the binomial arrays would give better results not only because of the absence of the lateral lobes, but because of its greater directivity index, permitting a more uniform sound distribution in the stadium or bullring. (It can easily be obtained a relationship of 6 dB between the angle of 0° and 30° depending on the vertical plane of the main lobe). These facts together with the loudspeakers radiating in phase allows to increase the system efficiency with a greater range.

Today it would not be difficult to use a digital controller and delay lines, aimed to set a desired distribution of amplitudes on the loudspeakers array. This distribution of amplitudes could be function of the frequency (with the aid of 1/3 octave band or parametric filters). It can also vary the group delay with the frequency and also vary the gain of the signal applied to each element of the group and to get sweeping effect, similar to the sonar, over the audience. In the case of stereo, this is quite easily with the pan-pot.

An other aspect which is important in the piling up of acoustic boxes, is the over all frequency response. With a flat response in the interesting frequency range, we can see that when the number of loudspeakers increases, there is loose of highs, what is equivalent an increase of the pressure level in the frequency range of the woofers, as shown in the study of Grander and Eargle (1990). This phenomena is for the following reasons, first of all, as you increase the number of units, the resonance frequency of the system is moved towards lower frequencies because there is a mutual coupling between loudspeakers that increases the radiation reactance. Likewise, there is an increase of the resistance on the complex radiation impedance both in the tube and in the diaphragm, which increases the efficiency of the array at low frequencies. This effect is easy predictable from Thiele-Small parameters, will reinforce the woofer and should give a boomy response. The second effect is the increase of the directivity in the system as compared with an isolated radiator, on the principal axis of the array.

All this corresponds to a reception point situated in the Fraunhoffer zone, this is normally consider as being several times the length of the radiated wave and several times the maximum dimension of the array.The sound pressure produced would be n times that of each single loudspeakers and the acoustic power radiated would be at n^2. In this area applies the propagation law of $1/r$.

On the other hand when the reception distance is the same or less than theλ radiated and the dimensions of the array, we are in the near field or Fresnel zone, characterized by extremely abundant maximum and minimum of complicated evaluation, and here the propagation law is $1/\sqrt{r}$, characteristic of bidimensional sources, that is to say cylindrical waves.

6. Acoustic field in the stage

As we have seen, we can now have an idea of the acoustic field generated for the audience, but in the stage the situation is quite different. The normal placement with two arrays or boxes at the sides of the stage, to radiate a great acoustic power, for large distances and covering the arena as uniformly as possible, has the problem that, on the stage the acoustic field created is very irregular.

First of all this acoustic field is situated in the near field of the arrays, in which there are a great amount of interferences between the acoustic wave from each array and between arrays happens. These interferences are function of the frequencies and of the listening position.

Secondly, the greater directivity of the high frequency units, directed toward the audience, make that in the stage, the sound field has an excess of bass and mediums, which difficult the harmonization of the musicians.

To avoid this, a monitor is placed in front of each musician so that they can listen the music they produce together. This process is more complicated because each musician only requires certain signals. For example, the percussionist just need the sounds of the bass-guitars and the singer use the ensemble signal and, the return emitted to the audience, etc. For this reason, usually a technician controlling a second auxiliary mixer should assure the correct reproduction of the music created in the proscenia.

The number of monitors to cover the stage, depends on the groups, for example Serrat used 7.400 Watts on scene whereas, to the exterior, 16.000 Watts per channel were needed for an audience of 3000 persons. Sometimes the monitors are substituted by earphones for certain musicians.

7. Putting all together

Up to now we have dealed with the main part of the system, the emitter transducers, both as independent units, and the problem of grouping these them together in complex units.

We should also look at other electroacoustic elements which are used and the integration and handling will characterize the resulting sound (which in turn take part and symbolize the musical group).

First of all, we would have to bear in mind the captor transducers, which can be classified into two major groups: microphones and vibration pick.ups.

Really it is not an easy task for a sound engineer to find the most proper transducer for each instrument. It should not only amplify the sound produced by the artist, voice string or winter, but also the electroacoustic extension of the modifications which the artist wishes to include as part of its musical expression, therefore the transducers are connected in a differential way to the mixer and to the rest of the electronic complex.

As to the microphone, there are many types in use, but basically condenser, dynamic type and the electret (very small) for low diameter percussions, are used.Normally, the common denominator defining them are:

- Flat frequency response and phase (20 to 16.000 Hz)
- A cardioide directivity response in order to eliminate feed back in the system (front to back >20 dB)
- Adaptive sensibility for each type of instrument: the one we introduce within the percussion boxes are normally hard (low sensitivity, about -90dB re 1v/1Pa)
- Great dynamic range, with high relation signal-noise (>70 dB), adapted to each type of instrument.

In the case of dynamic microphones of mobile coil used for voice, there is a special technology to avoid the impact effects, incorporating MYLAR diaphragms with tangential compliance. They also include pop screen filters to avoid whistling coming from mouth, as well as variable gain and filters. In the case of the microphones for singers we should not overlap the importance of the look.

For musical instruments, normally it is avoided any airial acoustic pick-up, using magnetic pick-ups, or capacitative transducers with polarized plaque for the strings of guitars and pianos. In the case of wind instruments, small microphones are installed (usually electrete) in the mouth pieces. Many instruments are connected directly to the electronic centre, such as keyboards and synthesizers, controlled by MIDI (musical interphase digital).

Due to the high SPL existing on the scene and despite the careful directive characteristics of the transducers in use, there are sometimes acoustic feed-back. To avoid it the distances between the microphones and the acoustic sources are reduced to the maximum, and it is necessary to use electronic to avoid the Larsen effect. Some decades ago, this effect was minimized by moving the frequency band between 3 and 4Hz in order to avoid the Nyquist frequencies for wich the system gain was > 1 with 180° of phase shift. Actualy, use is made of adaptive digital filters with a reject band with a width below 1/10 octave, that acts in an automatical way.

The electrical signal of the different instruments are sent to the sound enginier mixer to be properly equalized before being send out to the power channel they , by application of high, low pass, 1/3 octave or parametric filters, in order to equalize acoustically the sound to be radiated to the audience.

Also digital electronic gates are used to move the dynamic of the system just a couple of dB above the ambient noise, in order to radiate at high acoustic level with the maximum relation signal-noise. This operation is automatic and the system is adjusted any moments. The noise gates work as expansors at the lowest level of its dynamic.

Among the added effects the most commonly used is the artificial reverberation. In this effect, here there has been a great evolution in the systems since the fifties, where time delays were obtained with one or several springs, with 2 transducers at the ends (this is a retarder which could be called linear, with limited response and very coloured frequency), in the sixties, and seventies they used reverberative plates excited by electrodynamic transducers, the flexion waves produced in the plate, were reflected at the edges, and it was time controlled (control between 0 and 5 s)with a close porous sheet (bidimensional model with high colouring).

Actually, the reverberators are of digital technology based in the statistic model proposed by Schröder (3 dimensions) in 1960, with comb filters and all pass filters which have practically no colouring.

8 System equalization

After having settle the stage, which is not an easy task, because the large weight of the components (normally calculated for more than 1000 kg/m2), and the bearing structures for the loudspeakers columns or walls, the arrays have to be aligned and the system equalized so the different angles be covered by the arrays.

In our opinion, instead of looking for the most adequate directivities, different types of configuration used for these columns should be intended to obtain the minimum of destructive interference, and a uniform distribution of the SPL.

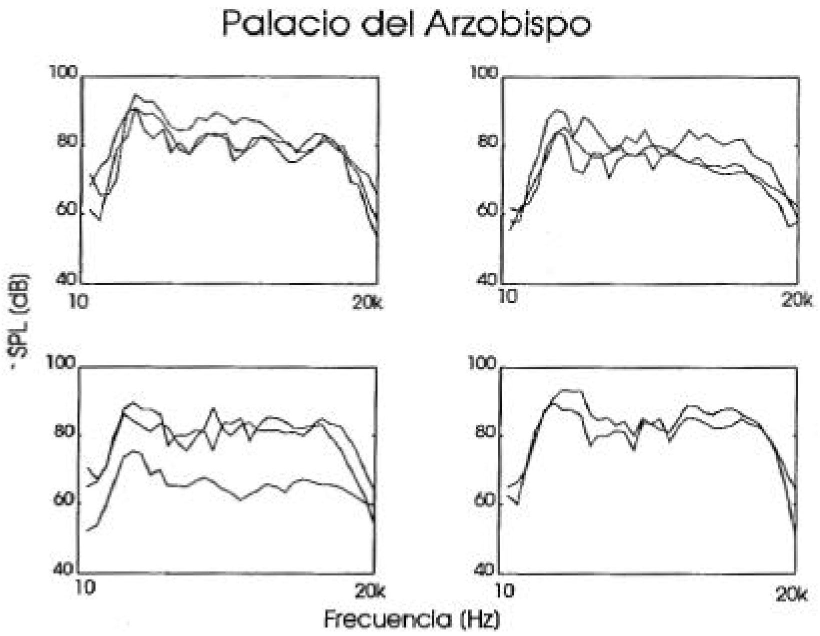
On a other hand, and because normally there are not any masking effect due to multiple reflections, the defects of the arrays in the acoustic signals become audible and sometimes the results are devastating. To avoid such situations signal delayers are used.

Another criteria in sound reinforcement in open air, for the different angles of coverage are the gradients of temperatures on the audience. Depending on the path of the acoustic waves, the positive temperature gradients (ground warmer than air) do not produce negative effects whereas the contrary would produce the same effect as the inverse of a comb filter, with a loss of frequency, in zones curving the sound rays upward. The concerts in the evening and night help since the temperature of the ground is higher than that of the air. We should also bear in mind the wind effect, also the absorption introduced by the public and the attenuation in the air at high frequency for distances over 40 m.

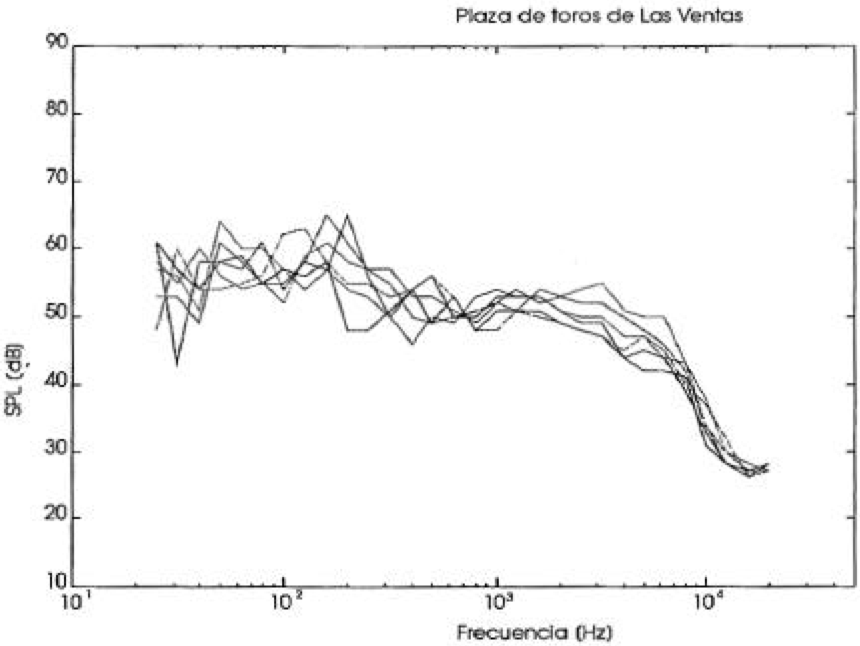
There are also possible reflections from the surrounding buildings or protecting walls which would oblige in these cases to vary the orientation of the arrays, or change their locations.

After this precaution and after calculating the required electric power, follows the lineal equalization of the system, with insertion of parametric filters. Usually, this operation is realized sending pink noise to each sound unit and registrating the spectra in the different listening zones. The ideal register would be that of flat frequency response and phase with a minimum of deviations. The instrumentation to bring about these types of adjustment are the analyzers of 1/3 octave, backed by the FFT analyzers, or TEF and SIM, more and more used.

We are now going to give two examples of performances where we have being present during the lineal equalization process. The first case is a concert for 2500 people, in the Archbishop Palace in Alcalá de Henares (Madrid). The place had two big difficulties because there



were two buildings at the back and side of the scenary. In the figure are reproduced the pink noise for three positions along the axis of the array and along the symmetrical axis between them (at 3, 18 and 31 m). The distribution of the levels on the audience is fairly uniform, but there is a dead zone for the first row of seats at the central axis of the scenery. Equalization was made at very high level and in our opinion the subjective impression was very deficient.



The second case is the Ventas Bullring, It was equalized with the SIM at low level pink noise, and some echoes were perfectly detectable in certain areas: In both cases the presence of public improved the attenuation of the echoes in medium and high frequencies. The figure presents the levels and spectrum for 6 reception points situated at the axis of the array, at the left of the stage. (Nearest point 6 m and farest 70 m). The obtained type of frequency response give a high subjective quality for pop music. In both cases, and this is normal, the distribution of the stereo signals was very closed (practically identical signals on both channel), so that the public who is far away from the symmetric axis between the arrays, do not loose part of the musical information.

9 Summary and conclusions

In large places, the following factors should bear in mind :

- The area to be sound reinforced should go: with the relieve of plane and cross section.
- The environmental noise (dBA o NC) normally considered greater than 60 dB.

- Maximum distances in which the sound is to be projected.
- Angles which cover the potential listening areas.
- Maximum acoustic level require and the minimum admissible.
- The presence of possible destructive interferences.
- Directivity factor of the emitters to be used.

Among all these factors, the most important which require more attention is the last one, since with the electroacoustic characteristics of sound sources it can be calculated the electrical power necessary to produce the desired acoustic pressure levels.

In the open air (using the l/r law) the directivity of the radiating systems have a greater importance, rather as an additional way for increasing the sensitivity rather than improving the intelligibility of system. When about intelligibility, it seems that music and speech are interpreted at different brain hemispheres, and therefore for the type of rock-music related with this kind of happenings, some attribute which evokes the articulation seems not to have much importance for the "perception pleasure" of the audience. For this reason it is possible to pile up loudspeakers and use high electric power quite freely.

The common denominator of the sound system is:

- Very powerful reproduction in the range of low frequencies
- Acoustic pressure levels very high > 100 dB
- Very closed micing
- Powerful intimate monitor loudspeakers systems.

Therefore, depending on the musical group an size of the audience, the power for the system can be between 30.000 Watts (for audiences of 2 to 3000 people) and 200.000 Watts, as is the case of the Jackson, (for audiences of 40 to 50.000 people). This can give an idea of the size of the arrays needed.

All these systems working at very high levels, do not permit the listeners to distinguish small nuances, at such high level the hearing mechanism produce distortions. For loudspeakers manufactures, it is a real challenge to design systems capable of working at such high levels and to produce a clean sound, and also the better responses to possible transients, which in slang is called "punch".

Modern music and the system used have very little relationship with the carefull design of electroacoustic reinforcement for speech and music. This means that there are factors out of the bibliography, which can only be acquired through experience.

Besides that judgements are different for different groups, the rock sound is product of sound images, illumination, visual images, representation, symbolisms, ambient and a lot of very costly extra, factors that must be born in mind.

If we ask the neighbours about their opinion, they will be very different. The figure shows the variation of SPL at a nearby house a "night of concert".

We deeply believe that the phenomena related with modern music, and the paraphernalia involved requires a deep sociological study, that should help to understand, on acoustic basis, why mixes which are so distorsioned and with incoherent phases have such great level of acceptation among the majority of listeners

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