UNIVERSIDAD POLITECNICA DE VALENCIA

ESCUELA POLITECNICA SUPERIOR DE GANDIA

I.T. TELECOMUNICACIÓN (SONIDO E IMAGEN)

UNIVERSIDAD POLITECNICA DE VALENCIA

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"Test and Assessment of Sound Equipment of the School of Media of Oulu University of Applied Sciences."

TRABAJO FINAL DE CARRERA

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GANDIA, 2012

Thanks...

...to Rubén Picó, for your life lessons and academic support.

...to Heikki Timonen, for your wit for making easier the difficult tasks.

...to Jusso Jakola and Julius Zlioba, for your valuable help with the heavy job.

...to my family, for your sincere support whatever my way.

...to all my life mates, because there is a piece from each of you in this project.

Index

1. Introduction of the project

This project consists on an assessment of sound equipment in several predefined situations or conditions.

It is conceived as a useful guidance for the tested material, more than a technical characteristics catalogue. Its purpose is to show how the equipment behaves under different test situations, specially chosen to approximate as closely as possible to real sound picking situations. Thereby, a person who is going to do a sound pick, can consult this project to decide which is the best audio equipment he can choose for his purpose, and also which is the best way to use the selected device, or perhaps, what assembly he should dispose to obtain a certain effect.

The project is composed of five parts, with the finality of obtain, present and study the expected results in an ordered way.

First, a list of the material to use is displayed. In this chapter, the reader has an introduction of all the devices used for the project where general features of each device are described to give an idea of why this device is interesting both for the project and the user. In order to complement this introduction, the technical specifications of each device are attached in the annex of the project. Not being the main results of the project, these technical features will be the starting point for explaining the behaviour of each device in the different tests.

Second, a study of three different environments that allow to simulate three real, well differenced conditions. This "real" means probable situations or emplacements that students are going to find while doing their projects. Since the project is raised as a useful guidance for sound picking, one of the main goals is presenting common situations that the user will probably have to deal with.

Third, in this chapter we find two main sections, first the selection of the recorder and second an introduction of the test procedure. The election of the recorder is faced from different points of view in order to find the one most appropriated for the tests. About the test procedure, it must allow us to gather all interesting data that we need to obtain the expected results, so it is carefully designed and it will also be well explained in the corresponding section.

Forth, presentation of results. They must be presented in a way that allows us to easily compare the results obtained for different microphones in the same location, and the results obtained for the same microphone in the distinct environments.

Fifth, a study of practical applications for the tested equipment, which will be extracted from the obtained results. The pretention is explaining some basic concepts to have in mind where we have to make a recording in any of the presented situations. A sound picking is always a combination of technical knowledge and artistic feeling, used together for achieving a wanted sound and these two aspects are discussed in this chapter under the shape of technical contributions or artistic advises.

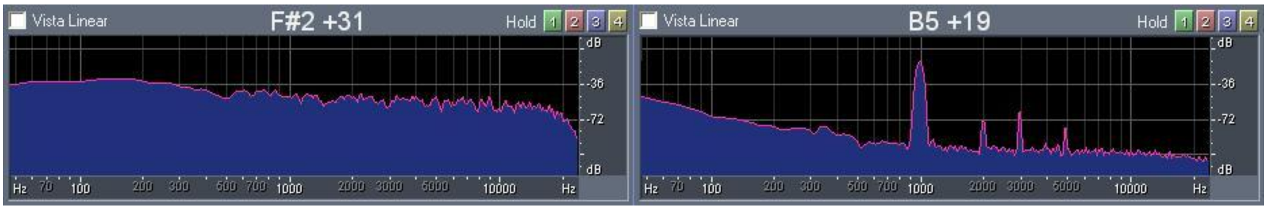
With this structure, the information is given in the logical way; from the introduction of the different devices, the discovering of their capacities, until the learning of the way of using them for exploiting their capacities.

1. Interior of the recording studio.

All tested material is professional equipment, possessing its official technical characteristics provided by the manufacturer, and they will be used in this project as a starting point. But the difference between these technical characteristics and the results we expect to obtain with this project is the following: these characteristics are always picked under ideal conditions; it means that all measures are taken in anechoic chamber in order to get the most precise data. On the contrary, we expect to pick the behaviour of this equipment in real or usual conditions and uses.

Therefore, sometimes the sound picks will not be clean, including some background noise or reverberation, but it is part of the finality of the project. The idea is using these tracks to show to the reader the common and real troubles or issues that he will find in a real sound recording. From them, the reader will be able to learn how affect these issues to the signal, how can he avoid them or on the contrary, how can he use them in order to achieve a wanted effect.

Due to the nature of the project, obtained results will be presented in two main ways. On one hand, in order to show the data in the most precise way, numeric results will be given, supported by graphics or other objective data when it’s required.



2. Example of displaying of results.

This is an example of the graphics used to extract the numerical information. The first one represents the frequency response of the microphone, and the second one represents the captured signal, when the source is emitting a pure 1kHz tone. Among the information that we can extract from these graphics is the frequency response of the microphone (supposing that recorder has a completely flat response between 10Hz and 20kHz), and the level of the 1kHz tone, which allows us to compare the different sensitivities of the microphones and the variation of the sound pressure with the distance, for each different environment.

On the other hand, we will provide a sound track of each sound pick, accompanied by a “subjective” assessment. It allows the user to asses by himself if the sound pick corresponds to what he is looking for. In this assessment, some acoustical terms will be used, and they will be explained so that the reader can learn them at the same time that he is listening the effects of that concrete parameter in the sound signal.

This has been considered as the best way to show the results because it covers all goals that the project pretends to reach. First, it provides precise data that allows making acoustical calculations of sound level, noise level or frequency response. Second, it establish a database where to hear the response of the equipment under certain condition, which, in some cases eliminates the need of doing equipment tests before the sound picking, and in other, reduces the amount of equipment to check in order to select the best one to use for sound picking.

Due to the way the project has been raised, it easily covers an educative task. It is very appropriate as a support material for sound or media courses, working as a guide for the students to understand the appropriate uses of the different devices, how the response of the equipment changes depending on the surrounding environment, and the effects that can be achieved depending on how the equipment is used.

Obviously, it is especially useful for the O.U.A.S., because tested equipment belongs to the department of media of this university, so the pupils are studying and listening the response of the same devices that they are using. But moreover, this same project can be interesting whatever is the available equipment. Thereby, the reader shall use the results of this project to learn the bases of the behaviour of the different classes of microphones. Lather, devices from different brands will have different response, due to the construction process or because they are specially made for achieving specific reactions, but anyway, behaviour of any device of the same kind will always agree with the results shown on this project.

A good example are the dynamic microphones used in this project, EV RE50 and Shure PG58. They are different models, belonging to different brands, but as we will extract from the results, they have a common behaviour compared with the rest of the microphones. This is because they are sharing the dynamic feature, while the rest are condenser ones.

Then, this project pretends to complement the technical information provided by the manufacturer, showing how each device adapts to the different situations in which is tested. So, after studying this document a reader should be able to know how to compare different microphones from their technical specifications and moreover, he should know how these differences among devices will affect the sound signal, allowing him to choose the best equipment for its concrete purpose.

2. List of equipment to test

The equipment used for this project belongs to the department of media of Oulu University of Applied Sciences. All this equipment is completely available for the students of this department to use it in their projects, therefore this project is an interesting tool for these students to decide the best equipment to chose.

The available equipment can be divided in four groups:

- **Recorders**
- **Microphones**
- **Headphones**
- **Mixers**

Since teaching the behaviour and the possibilities of use for the equipment in different situations is one of the main goals, this project shall focus on the devices most affected by environmental conditions. For this reason, microphones are going to be the main tested apparatus. We will notice how important is the emplacement and the distance to the sound source, and how can we use it to achieve a concrete wanted effect.

Obviously, if we want to assess and compare the recorded sound signals, we have to achieve the truest signal according to the response of the microphone, meaning that used recorder mustn't distort the caught signal. At least, it should only put on an acceptable level of distortion, always below a maximum limit. Then, an election of a recorder is going to be done, following some reasons described later on.

About mixers, they are not going to be tested in this project. The reason is that theoretically they are not affected by surrounding environment, unless it is extreme. Then, it is enough knowing their technical characteristics. An interesting aspects we could asses are the ease of use, or which one is more appropriate for each situation, but these are features that do not affect the audio signal itself.

Headphones are also not so depending on the environment as the microphones, so the most valuable information that allows us to know their behaviour is their technical data.

Due to the characteristics of the main target group of this project, it is possible that, at the beginning, some difficulties appear when treating to understand the technical information, but one of the aims of this project is just teaching basic acoustic features, like directivity, sensitivity or frequency response, that allow the student to interpret technical data and moreover knowing how this technical features affects the sound signal.

I. RECORDERS

Testing the recorders has the finality of choosing the one which offers the best features for recording all the sound picks. During the whole project, the same recorder will be used to record all sound takes, ensuring that it won't be any sound variation due to the recorder itself. Then, recorders will become first equipment to test.

The set of recorders available at O.U.A.S. comprises the following ones. Among them we can find from hand recorders to devices used in professional cinema productions, then a general description of each one is given in case it is useful for the reader as reference material.

o HHB MDP-500

The HHB Portadisk MDP500 is a compact portable MiniDisc recorder. Is remarkable to be the first portable MiniDisc to feature a USB interface, for transferring audio to and from Windows and Macintosh computers. It has high quality balanced XLR mic inputs with switchable Phantom +48V power. Its main inconvenience is that MiniDisc storage format is becoming obsolete in favour of the new solid state memories, which offer more storage capacity in a robust case.

Price: ± 1500$

3. HHB MDP-500

o Aaton Cantar X

Cantar X is the top of the class digital audio recorder. It has been designed for highest quality, high performances and ergonomic use, meaning that all components have been selected or designed to offer the best characteristics, but also the user interface has been set in a way that allows instant access to all recording and monitoring configurations, rejecting multi-function knobs in favour of dedicated ones. This fact, gives it clear advantages over the others, but also an important disadvantage, the price. It makes this recorder only affordable for big productions, which deviates from the goal of this project.

Price: ± 15000$

4. Aaton Cantar-X

o Maycom Handheld II

Hanheld II offers rugged and reliable portable recorder for field recording due to its small size, long duration batteries and built-in microphone. Using state-of-the-art electronics, it is able to obtain a high quality sound, but never comparable to the one achieved using a dedicated professional microphone instead of a built-in one. Although it is appropriate for both professional and consumer usage, is easy to notice the home use design, specially due to the ease of use.
The Docking Station is also available, including two microphone preamps which allow to connect external microphones increasing the capabilities of the device.

Price: ± 780$

5. Maycom Handheld II

o Sound Devices 702T

The two channel 702T is a powerful file-based digital audio recorder. Compact Flash cards and Firewire make field recording simple and fast. Microphone preamplifiers are designed specifically for high bandwidth and their fidelity allows high bit rate digital recording. Is also remarkable the great amount of processing possibilities offered in the users menu, and the ease of use for such a high quality recorder.

Price: ± 4000$

6. Sound Devices 702T

o Sound Devices 744T

This device offers four track recording. It has two microphone preamplifiers, being the third and forth channels for line level. As the 702T model, it offers wide possibilities of routing, being increased in the 744T with the two extra tracks. Being based on the SD702T, they both offer similar signal quality, but extending the 744T the use possibilities.

Price: ± 4000$

7. Sound Device 744T

II. MICROPHONES

Microphones are the main equipment which is dedicated this project, due to they are the most affected devices by the surrounding conditions.
The main target is showing clearly and in ordered way the variations in sound picking for the different situations defined in the tests.

o Neumann RS191 Stereo

The RSM 191 is a stereo microphone system consisting of the microphone and the MTX 191A matrix amplifier. It has an adjustable pick-up angle and high directivity. The microphone has two separate capsule systems, a hypercardioid element and a figure-8, both in a short shotgun. Together they catch the central and the side signals.
The signals captured by the two capsules can be combined through the MTX191A Matrix, allowing us to use this microphone for different stereo techniques.

8. Newmann RSM191.

9. Newmann RSM191 configuration device.

o Shure PG58

It is a vocal designed microphone as can be appreciated from its frequency response diagram. As a vocal one it has a cardioid polar pattern helping it to focus on the source and reject the surrounding sound to avoid feedback. Also it presents a remarkable proximity effect, which gives presence to the voice and for this it is very appreciated by singers.

10. Shure PG58

o Sennheiser MKH-416

This is a shotgun microphone that employs a mechanic-acoustic interference principle to reach its high directivity. Observing its technical characteristics, from the directivity pattern we can distinguish how the high frequencies are even more directives than the low ones. This become an important effect to consider for every sound take because varying the angle of the microphone respect to the source, we are going to obtain different EQ for the recorded sound.
This high directivity together with its remarkable rejection to feed back, makes this device very appropriated for television or cinema uses, specially in open air takes.

11. Sennheiser MKH416.

o Electro Voice RE50

It is the industry standard handheld interview microphone for television production. Very well protected against unfavourable environmental conditions, and isolated of wind-noise and p-pops via its four-stage pop filter, which prevents dust and magnetic particles from reaching the diaphragm. Mechanically induced noise is also greatly reduced.
One of the main reasons of its roughness is the construction of the internal transducer where each part is perfectly nested inside the others, becoming nearly a solid structure.
Despite a high sensitivity generally is a good feature for a microphone, in this case the low sensitivity allow this microphone avoiding part of the background noise that we find in the open air. Knowing the main application for this device, it becomes a very good feature in this case.

12. Electro Voice RE50

o DPA 4006 Studio

DPA 4006 Studio is a worldwide reference thanks to its precise and warm reproduction. It's equipped with a state-of-the-art transformer-less preamplifier that increases the sensitivity and provides an extended low frequency handling capability.
It offers something more than a standard good sound, for this reason it has become a reference for many classical recording engineers. Christoph Frommen, producer and recording engineer for the company Aeolus sentenced "Hardly any other omnidirectional microphone is able to match the clearness and the undistorted sound of a small diaphragm microphone like the 4006", after recording a complete organ work from Bach using a set of seven DPA 4006 Studio microphones.

13. DPA4006 Studio

o DPA 4011 Studio

It is renowned in recording studios all over the world as an exceptional microphone. State-of-the-art components have been carefully selected to provide optimal neutrality, accuracy and extremely low distortion.
A completely flat on-axis frequency response and excellent phase response deliver a totally faithful reproduction of the original sound. The off-axis response is similarly smooth so that any leakage is an accurate reflection of the original, but attenuated according to the true first-order cardioid pickup pattern.

14. DPA4011 Studio

o Audio Technica AT 871R

This is a wide-range condenser microphone useful in surface-mounted applications. This characteristic shape offers some improved features, comparing with standard microphones. Directionality is increased by 3 dB, allowing enhanced gain before feedback and higher suppression of ambient noise. Sensitivity is increased for improved signal to noise ratio and phase distortion due to reflected sound energy from the boundary itself, is eliminated.

15. AT871-R surface mounted microphone.

o Sennheiser K6

In fact, what we call Sennheiser K6 is composed of two elements. The preamplifier K6 that provides power to the set and the hyper-cardioid capsule ME66.

The set conforms a shotgun microphone especially suitable for film and broadcast location applications and for picking up quiet signals in noisy environments as it discriminates against sound not emanating directly from the main pick-up direction.

As a condenser microphone it needs phantom power to be able to transform pressure waves into an electrical signal, but this one offers the possibility of operating with a single standard 1.5V AA battery. This feature results quite interesting for broadcasting uses due to the autonomy that confers to the microphone, non requiring a special device for supplying the phantom power.

16. Sennheiser K6 and Hyper Cardioid ME66 capsule

3. Election of the environment

The main feature of this project is testing the equipment in different real situations, pretending to obtain as a result the real behaviour of the equipment in common sound takes. To achieve it, three different environments are going to be selected assessing two main features.

On one hand, they should be common places, probably used by students on their projects. These common places can be recording studio, video studio, open air, particular homes, etc.

In this regard, we must divide the different locations in two main categories. First, students will use conditioned rooms, as the recording studio, for their takes. In this kind of locations, they should know how to take profit of the good acoustical conditions. For instance, it is important to know how to set a microphone respect the walls of the room, or depending on the material of the surfaces surrounding this microphone.

Second, they will also find non-conditioned or even acoustically-undesirable locations, as a crowded place or, for instance, a beach. In this case, they should know how to avoid or, at least, reduce the undesirable background noise and how can they obtain the clearest signal.

On the other hand, selected places have to cover a wide range of acoustical conditions. First, because it ensures that almost all possible situations will be collected in the project. Second, because it will shows adequately the different response of each device in almost all acoustical conditions, which is great for educative task. Thereby, although open air is never a proper environment for doing measurements, it will be useful for us to study the results given by the equipment at this particular environment, learning the troubles caused by wind or undesired noises and the best way to face them.

With these premises, and based on the available facilities of the school, the following three environments have been chosen:

o Open air

This is a totally dry environment (no reverberating), which completely changes the sound behaviour respect the enclosed rooms. The only point to take care about is the possible reflexions, but they can be easily reduced until they are not influent by caring the source position. For this, we have chosen a wide street with some separated buildings and the testing set-up has been separated from any influent wall.

17. Emplacement for open air takes.

The main element that we can expect at this environment is a low frequency background noise. The low frequencies are the ones which suffer less attenuation, then in almost every place we can appreciate a level of noise formed by signals coming from different distant sources that create a continuous low frequency sound.

Added to this constant noise we expect to find some other circumstantial sounds. In some cases they mix perfectly in the situation, creating a natural ambient, but in other they can be really annoying, as the noise of the wind is.

Although all these sounds can seem too weak, they create a background ambient in the recordings that we should mind depending on the kind of recording. As we will discuss later, in some cases we are going to take profit of this ambient, because it fits in our production, but in some other situations we will need to avoid it.

All this amount of secondary sounds together with the lower level of desired signal arriving to the microphone, due to the fast dispersion, makes the sensitivity of each microphone a very relevant parameter in this environment.

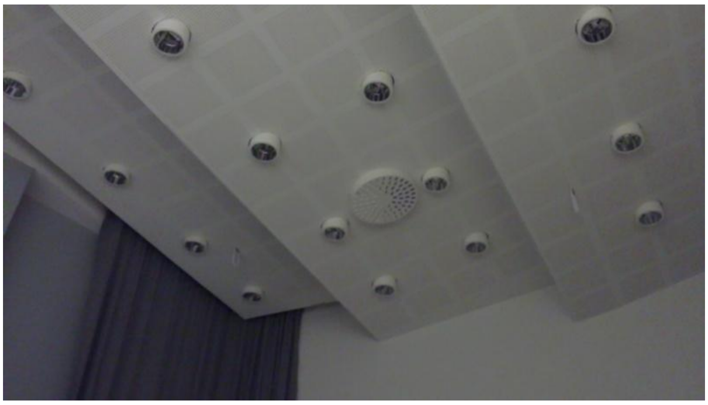
o Recording studio

This ultimate room for music recording, being the main reason why it has been chosen for the tests. It is a well acoustically cared room, which offers the possibility of taking advantage of the acoustic environment.

If we record a source from a mid long distance, the sound will be picked mixed with the ambient of the room, which is created by sound reflections on the walls. The effects of this contribution of the room are a colouring of the direct signal, which means a variation in its frequency response, and a modification of the temporal response of the signal, setting the firsts reflections and adding some reverberation.

In fact, both these effects are produced by the same reason, the reflexion of the spherical sound wave in each of the walls of the room. It creates a signal, which is the result of adding all the reflections present at the point of the microphone for every instant.

With the finality of controlling this signal, the room has been built following some geometrical premises, and inside it we find absorbent material and other acoustic elements, used for setting a low reverberation time and creating a diffuse space with a relatively flat colouring.



18. Ceiling of the recording studio.

The acoustical elements used in this studio are non-parallel walls, diffusers, sectioned ceiling and absorbent materials like acoustical curtains.

The main parameter that we can profit is the short reverberation time. If we place the microphone close to the source, it will pick a dry signal (non-reverberating), which is really useful if we are thinking in apply some post processing to it.

o Particular home living room

This environment has been selected thinking in the opposite of the recording studio. Here we don't find any kind of acoustical conditioning. The only acoustical treatment is on the isolation. Nowadays there are official rules that regulate the acoustical isolation between different rooms to ensure an acceptable level of sound exposure. But the isolation is related with the behaviour of the sound that travels through the walls, it has nothing to do with the sound that remains inside the room.

The main feature is the shape of the room, being completely squared, when a wave reaches a wall in normal incidence (forming exactly ninety degrees with the wall), this wave is going to bounce directly to the opposite wall. Then, the process is repeated in this second wall, which creates an stationary wave at certain frequencies.

This effect has two main disadvantages. First, it gives an irregular colouring to the signal, reinforcing some frequencies and attenuating some others and second, this lengthens the reverberation time beyond the desired time, and moreover doing it only for some specific frequencies.

The usual way for avoiding or controlling this effect is by using absorbent materials that earlier attenuates the wave, but in this room there are not any specific absorbent material. We could consider the thick curtain and the beds as absorbent elements, but as they have not been designed with this purpose, they are going to contribute in the colouring of the signal. Then, they do not perform an appropriate solution for the bad acoustic response of the room.

19. Particular home living room.

As we mentioned before, every room gives its contribution to the signal, but unlike a recording studio, the contribution of a non-acoustically designed room is so pronounced and random that it becomes impossible to control or take any advantage of it. In the best case we can set the position of the source and the microphone just to avoid, as much as we can, the acoustics of the room.

4. Test procedures

I. ELECTION OF THE RECORDER TO USE

In order to decide the best recorder to use, their benefits have been rated from different points of view.

We start with assessment of technical features provided by manufacturers. From these technical characteristics, the best recorder is “Aaton Cantar X”, since it has been designed with this purpose. About the others, “HHB MDP-500” and “Sound Devices 702T / 744T”, they also present good features, at least they are good enough for the purpose of this project.

Recorder “Maycom Handheld II” is the worst quality one from the available list. For other purposes, specially field sound picking, it can be useful due to its small size and weight, but for this project, we prefer sacrifice portability in favour of better sound features.

The next point to consider is the judgment of technical personnel responsible of the equipment. This judgment is considered important because this staff knows the actual current state of the equipment. According to their advices, they reinforce the idea of rejecting “Maycom Handheld II” due to its low quality. Also, they suggest not using “HHB MDP-500” because it isn’t in its best state, it has some recording problems and moreover, it hasn’t been used for a long time.

Finally, we must add a personal assessment according to the project’s purpose. Assuming that the choice has been reduced to “Aaton Cantar X” and “Sound Devices 702T / 744T”, due to the above reasons, we only rest rating them from the point of view of the concordance with the project.

Three ones offer enough quality to be used in this project, but the most important difference among them is the price, which means affordability. The price difference becomes important since Aaton recorder costs more or less four times more than Sound Devices ones.

For big professional productions Aaton can be affordable anyway, but if we want to be consequent with the purpose of the project, we must remember that its main target group will be media student, and the equipment available for them is mid ranged (inside professional field), not top of the class one.

Then, we can conclude that best recorder to use in this project is “Sound Devices 702T” or “744T”. On one hand, they offer enough good features that allow us to precisely assess the characteristics of the recorded audio signals. On the other hand, these recorders are the most suitable according the equipment availability for the students.

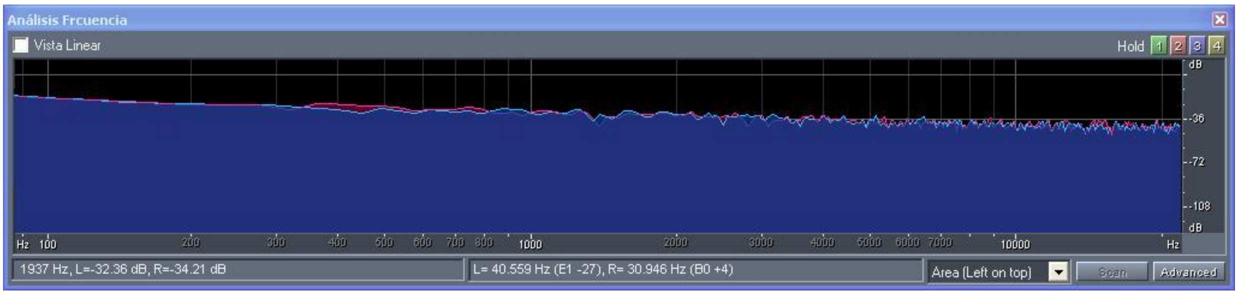
Between “702T” and “744T” models, the differences are mainly in the number of channels that each one manages, being the electronics and therefore the technical specifications the same. For this project only one microphone input is required (later, recorded signal is divided in two channels, allowing stereo listening), then we can set Sound Devices 702T as the most adequate recorder to use.

II. TEST PROCEDURE

Being the finality assessing the signal captured by the microphone in different environments and situations, the next test procedure is going to be followed:

The different sounds to capture will be generated by a PC, using the Behringer TRUTH B2030A loudspeakers provided by the school as a sound source. Only one loudspeaker is going to be used in order to simulate a punctual source, which imitates better the real situation of a speech take. Same gain level is going to be used for all tests, it will allow us to compare the differences in level due to the distinct distances, reverberation and features of the microphones.

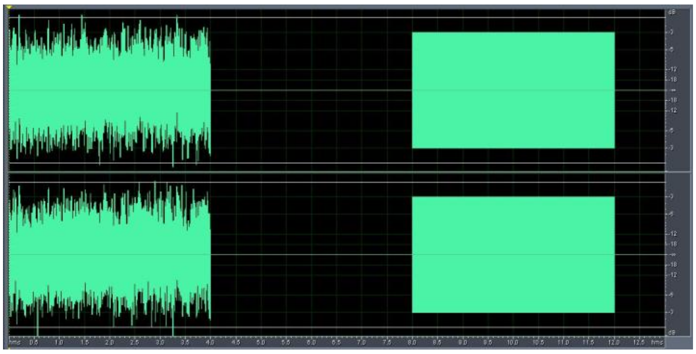
The PC will generate three different kinds of sound, first pink noise to assess the distortion produced by the microphone and the acoustical characteristics of the environment along the whole frequency spectrum. Pink noise is a wide spectrum signal where each octave contains the same total energy, then if we represent its spectrum in logarithmic divisions we obtain a flat spectrum as we can see in the following diagram:



20. Generated pink noise signal.

Contrasting the different obtained spectrums with this flat one it will be possible assessing the equalization produced by the room and the microphone. To difference if that distortion is produced by the room or the microphone, we will contrast the spectrums obtained with the different microphones, all common characteristics will be room made distortions and the differences will have been produced by the distinct microphones.

On the other hand, a single tone is going to be generated too. This pure tone will allow us to assess the differences in signal level. The generated tone is going to be 1kHz one, this is because 1kHz is a middle frequency within the flat frequency range of all the tested microphones and also, because this tone is representative of the human speech and it can serve to show the level differences in a speech take.



21. Waveform of test generated signal

Pink noise signal and 1kHz tone are included in the same cue, separated by four seconds silence that let time for reverberation to decay. It makes the results more practical to handle and easy to understand as we can see the two signals on the same grid.

Finally, it will be recorded a fragment of a song serving as a sample of the characteristics of the recorded sound for each take. Despite the previous signals are the technical ones, which allows us to take measures, this musical fragment is very clarifying to understand how the studied features affect the sound. Being elected with this finality, in this piece it can be easily differentiated a low frequency instrument, a high frequency one, and the midrange frequencies performed by the voice of the singer.

This musical fragment is extracted from the song *Anteroom of death* from the album of Tarja Turunen *What lies beneath* (2010).

About the assembly, it consists of two parts, the emitter that has been described above, and the receiver, which is the microphone under test connected to the Sound Devices 702T recorder. The microphone is always directed to the source, because it's the natural position in a sound take.

In each environment, each microphone is tested at three different distances from the source. First distance is the shortest one, the microphone is thirty centimetres from the audio source and it allows to asses two main features. On one hand it simulates usual distance for recording speech, for example in interviews, on the other hand this distance is always under the critical radius. It means that all recorded sound comes directly from the source, not being affected by the room, which let us know the actual behaviour of the microphone.

Second distance is between one and two metres from the source, being the middle one. This is out of the critical radius but very close to it, serving us to assess the change produced by the room in the audio signal and simulating a usual sound take in video recordings, where the microphones have to be maintained out of the shot.

Finally, third distance is the longest one being three metres from the source. In this position all the sound picked by the microphone comes from the reverberation of the room. This distance is very clarifying to understand the changes produced by the room if we compare these takes with the ones of the first distance. The most important feature of the microphones at this distance is their directivity, being able to anticipate, even before the tests have been done, that specially shot gun microphones are going to obtain better or purer sounds than the rest.

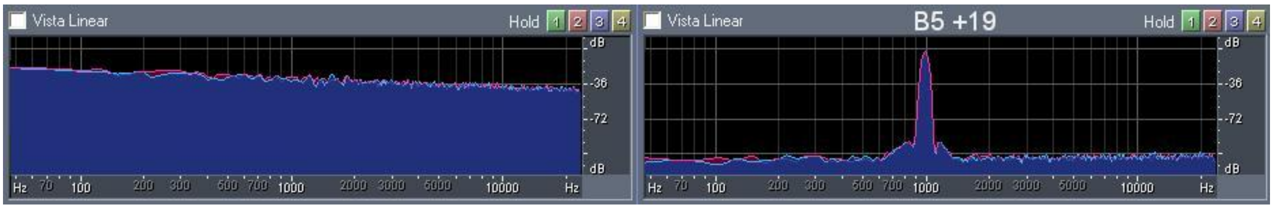
But there is a particular environment that escapes from all previously described, it is the open air. In this background there isn’t any reverberation that affects the signal produced by the source, so that it travels freely straight, making this environment the best one to assess the level loss caused by the distance. Then, there isn’t a critical radius in this background, because all picked sound is directly the one produced by the source.

Another new important parameter in this environment is the background noise. It always represents a problem to avoid in a sound picking, but it allows us to assess the background noise rejection offered by each microphone. In this aspect, we can suppose a higher behaviour for the Electro Voice RE50 microphone, due to it has been specially designed for interviews.

5. Test results

I. ORDERING THE RESULTS

The results are going to be ordered first by environments, inside each environment, the obtained spectrums will be displayed ordered by distances, accompanied by an assessment of all remarkable features.



22. Frequency spectrum of pink noise and 1kHz test signals.

Obtained frequency spectrums belong to the Pink Noise and 1kHz tone signals taken from each distance, being the ones that allow us to do the measures. The main features that are going to be assessed are:

Equalisation of the whole frequency spectrum. It is supposed that high quality microphones provide a flatter spectrum than low ones, for example we can expect a flatter spectrum from DPA microphones, because they are studio ones.

Sound level loss due to the distance. As acoustical law, sound level maintains an inverse relation with distance to the source, being the behaviour that we can expect in open air environment.

Sound level increase due to the effect of the room. The before acoustical law is not valid inside closed room, because passed a certain distance, called critical radius, the reflections with the walls produce a level increase that even surpasses the source-signal own level.

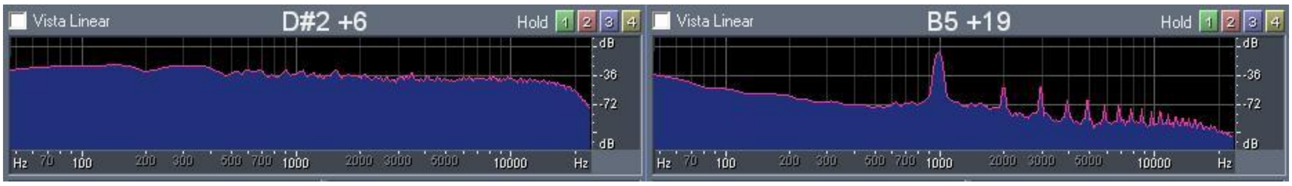
Effect of the environment in the recording. Closed rooms are controlled environments, but in open air, sound takes are always affected by natural agents. Being the finality simulate real sound pickings, takes influenced by these natural agents, as wind, have not been rejected, since they help to show better the environment characteristics.

To assess the peculiarities of each microphone we must compare its resultant spectrums with the pattern one. In this one, the average sound level for the pink noise signal is -36dB, being the spectrum flat from 23Hz to 23000Hz. About the 1kHz tone, it has an initial level of -3dB.

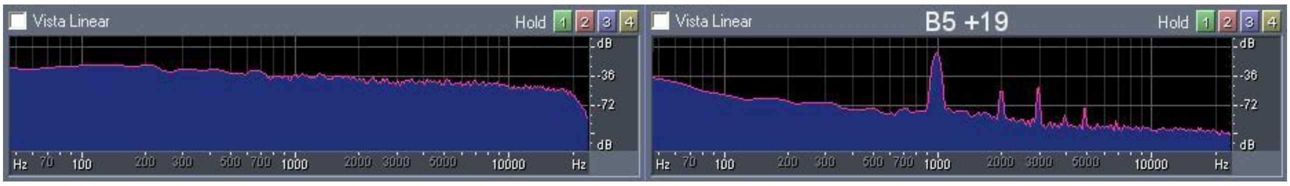
II. OPEN AIR

First position

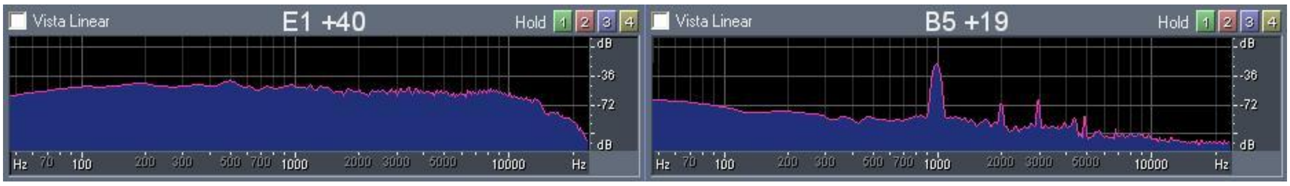
DPA 4006 Studio: Tone -7,8dB



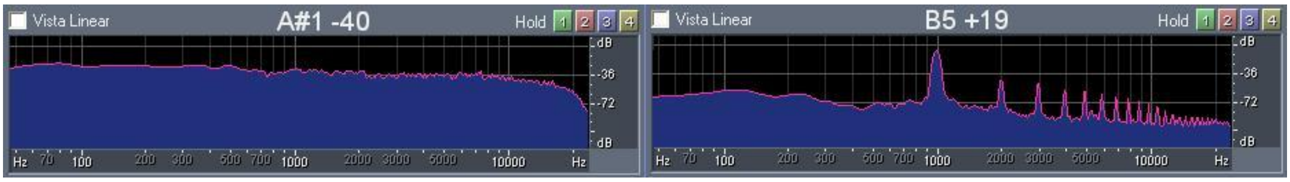
DPA 4011 Studio: Tone -7,7dB



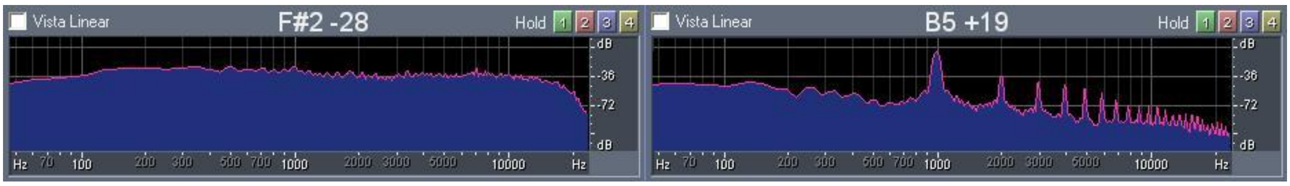
Electro Voice RE50: Tone -20,7dB



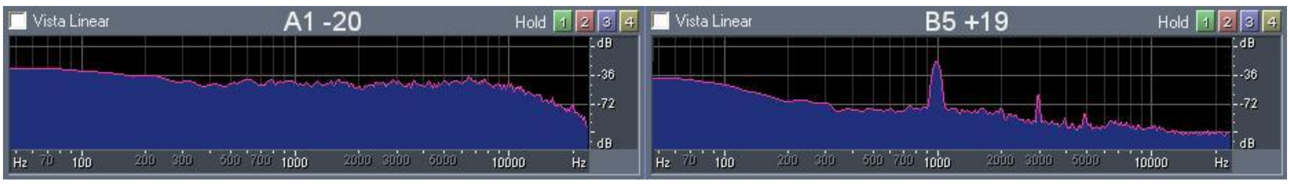
Sennheiser MKH416: Tone -6,6dB



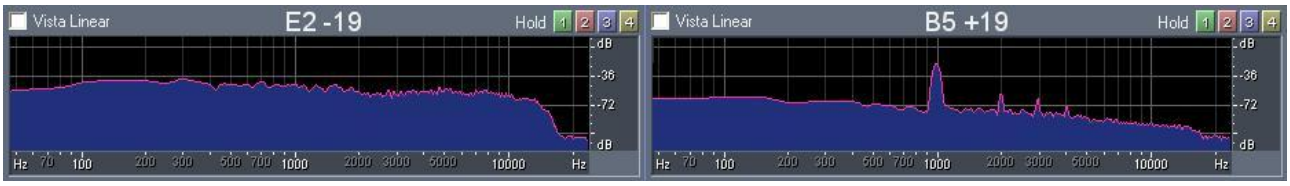
Sennheiser K6: Tone -5,5dB



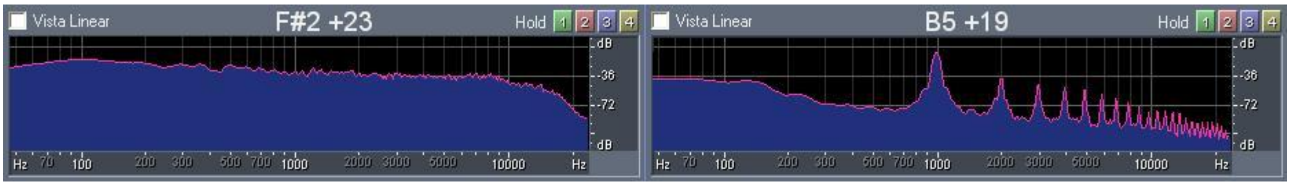
Audio Technica 871R: Tone -18,8dB



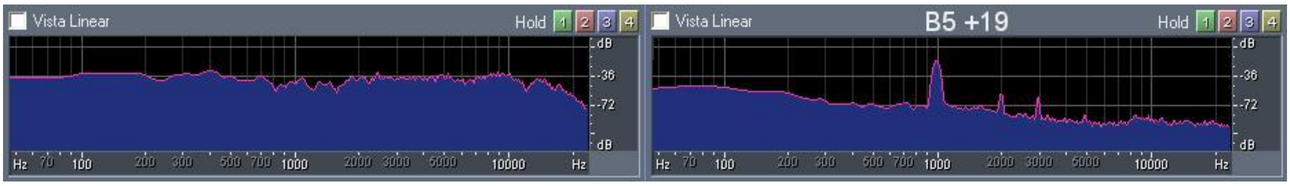
Shure PG58: Tone -20,2dB



Newmann RSM191 – M: Tone -6,2dB



Newmann RSM191 – S: Tone -16,9dB



From the pink noise frequency spectrum we can see first the low cut filter in some of the microphones. It is especially remarkable in EV RE50 and Shure PG58 ones, in both cases starting around 100Hz. These microphones have been specifically designed for vocal signals, so they should maintain a flat spectrum from 150Hz to 10kHz, with particular detail within 500Hz to 6kHz. This low cut helps rejecting the unwanted background noise without affecting the profitable spectrum zone.

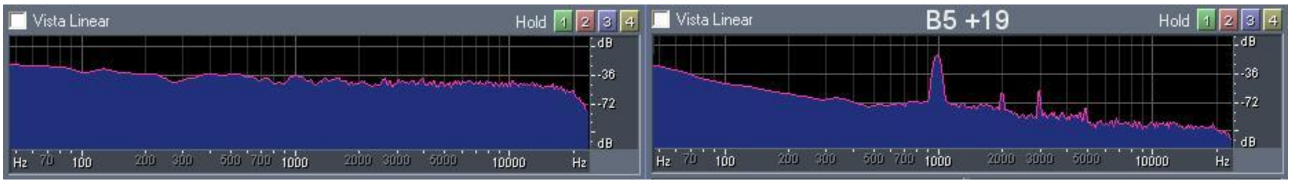
Both microphones also have a well marked high cut filter, but they are not the only ones since we can appreciate this filter in all the devices, but in a different way for each one of them. DPA ones, together with Newmann RSM191, are studio microphones, so they are supposed to keep a complete flat spectrum, because the EQ will be given to the signal in post-production, and as we can check, they meet expectations since they maintain a flat spectrum at least until 15kHz.

About the sound level, almost all microphones present a similar level. The main exceptions are, on one hand EV RE50 and Shure PG58, because they are dynamic devices and it makes their sensitivity lower. It means that they need more acoustical energy to produce the same electrical output level.

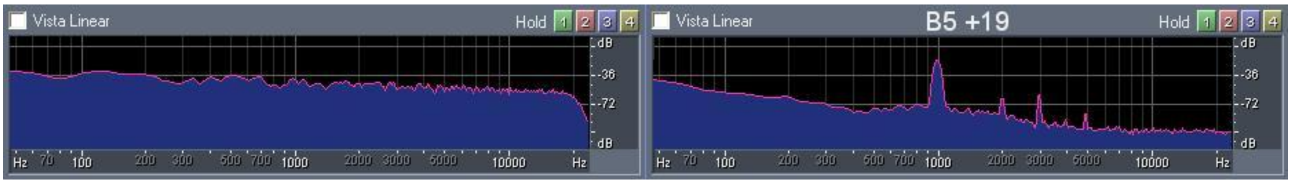
On the other hand Newmann RS191-S, because it has bidirectional directivity pattern, picking the sound coming from both sides of the microphone, but not the one coming directly from the source.

Second position

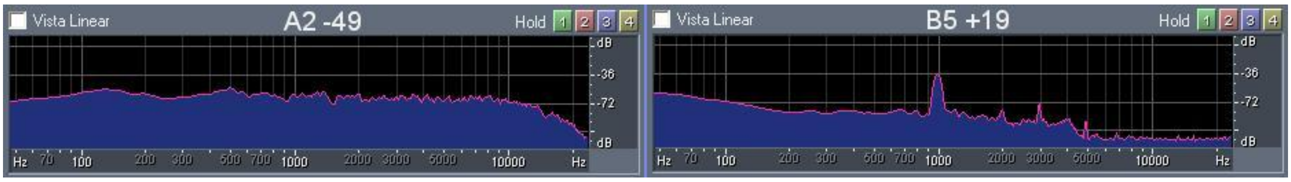
DPA 4006 Studio: Tone -12,3dB



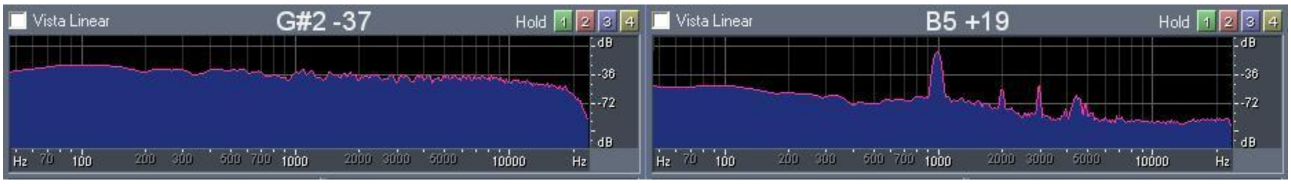
DPA 4011 Studio: Tone -16,7dB



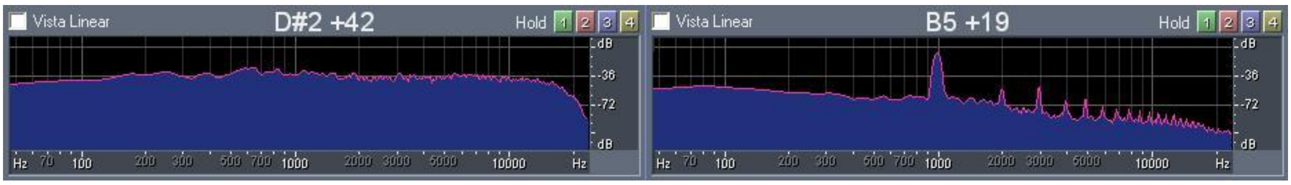
Electro Voice RE50: Tone -36,2dB



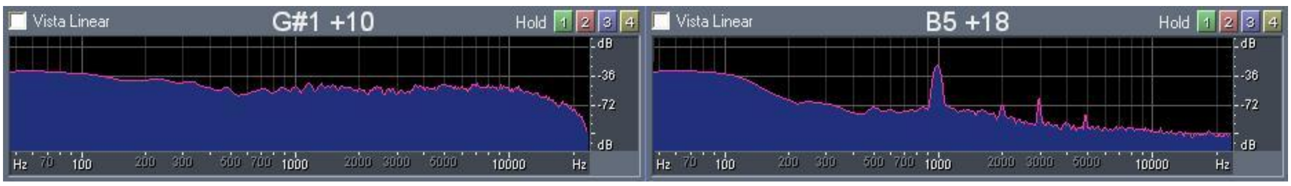
Sennheiser MKH416: Tone -7,5dB



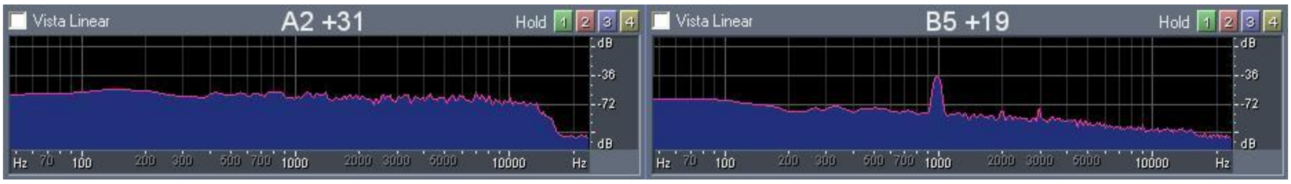
Sennheiser K6: Tone -7dB



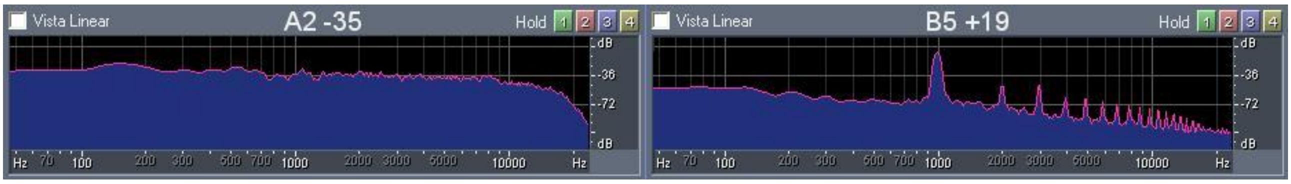
Audio Technica 871R: Tone -22,7dB



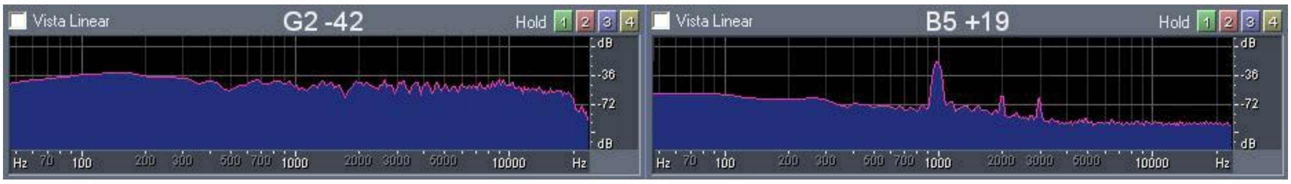
Shure PG58: Tone -37,3dB



Newmann RSM191 – M: Tone -6,9dB



Newmann RSM191 – S: Tone -18,3dB



Studying the results for this second position, we can appreciate how the pink noise spectrums are still fairly flat, but they start to introduce some irregularities. These ones are mainly produced by the background noise, which has totally random nature.

This effect was not so observable at first position because the source-signal level was well above the noise level, but as we start moving away the microphone from the source, the signal starts to be more affected by the background noise, making this effect more visible. Following this rule, we can expect that in third position it becomes even more pronounced.

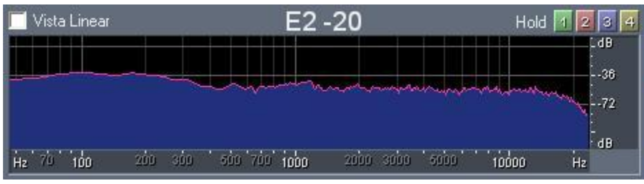
Specially remarkable is the increase on the lowest frequencies of the DPA microphones pink noise spectrum. Although background noise is totally random, it tends to be more significant at low frequencies and this is the case. The reason why we observe particularly in DPA microphones is because, as we mentioned at first distance assessment, they are studio microphones with good response at the whole frequency spectrum, including these low frequencies.

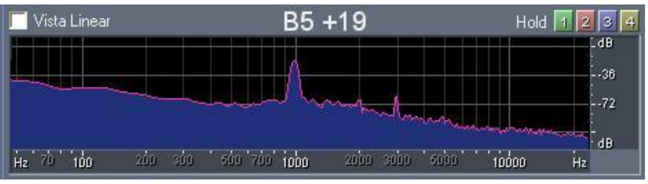
The second characteristic that we appreciate studying the results of the second position is the level decrease for all the microphones. But as we can see, this level reduction is not the same for all them.

DPA microphones show a decrease about 7dB, but Sennheiser ones and Newmann RSM191-M have only around 1dB decrease. It is due to their high directivity that allows them to focus in the signal coming directly from the source.

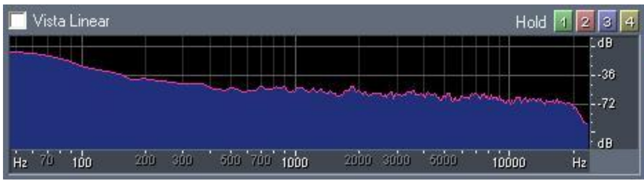
Third position

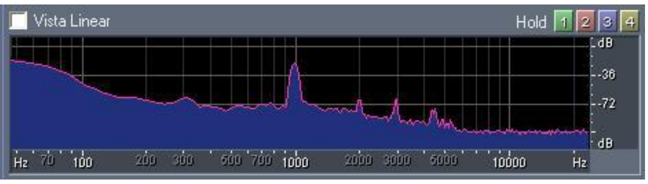
DPA 4006 Studio: Tone -17,5dB



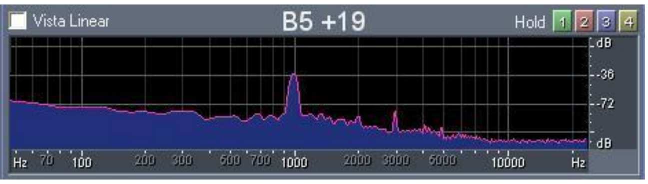


DPA 4011 Studio: Tone -21,5dB

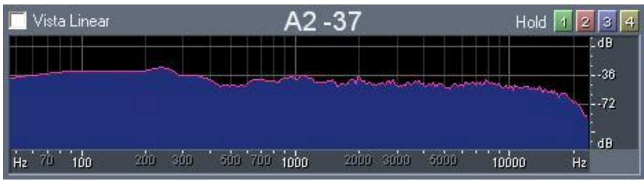


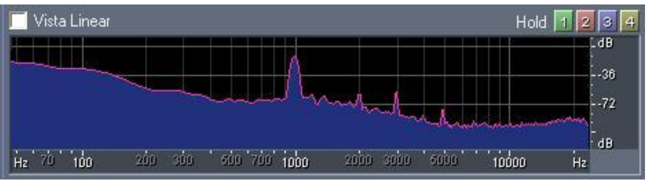


Electro Voice RE50: Tone -34,3dB

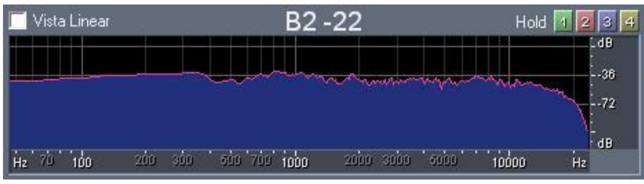


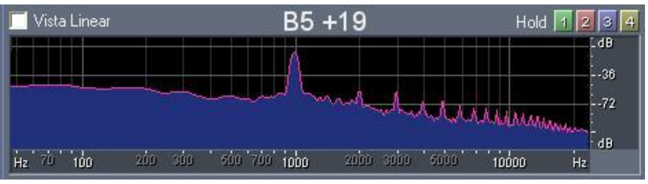
Sennheiser MKH416: Tone -14dB



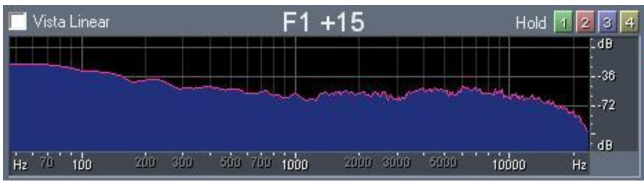


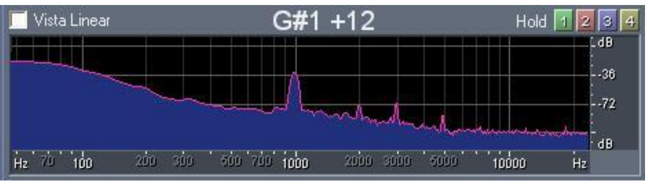
Sennheiser K6: Tone -7dB



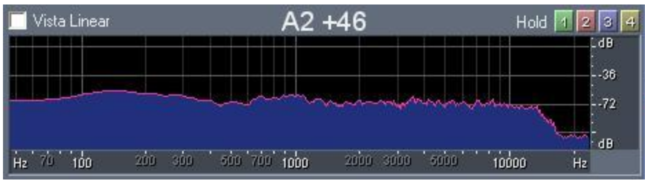


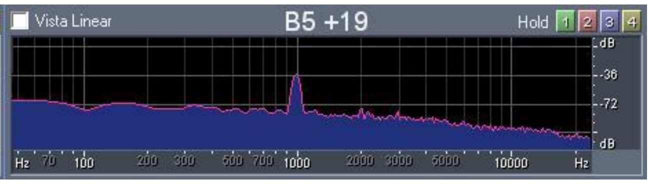
Audio Technica 871R: Tone -32,8dB



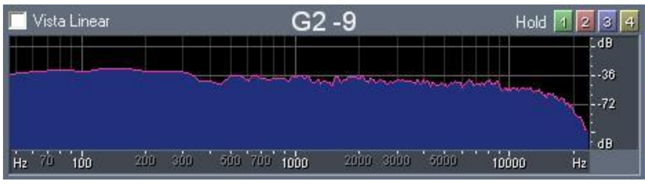


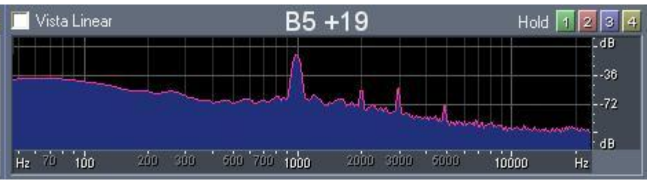
Shure PG58: Tone -34,7dB



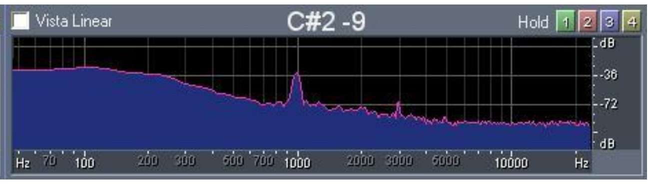


Newmann RSM191 – M: Tone -9,6dB





Newmann RSM191 – S: Tone -34,5dB



As we advanced at second position assessment, irregularities of the pink noise spectrums are much more visible at this third position. The larger we make the distance between source and microphone, the lower is the signal level and moreover, it is more affected by the surrounding noise, resulting this distortion of the spectrum.

The used tone is middle frequency, with harmonics in middle-high frequencies, but as we advanced, and we can ascertain from the graphics, the biggest background noise effects are produced in the low frequencies. This is because in this frequency range we find the background noise plus the effect of the wind, which moreover is not constant and it can vary its level to a large degree. Then, the wind becomes an important factor to keep in mind if we are recording music, because it is going to affect part of the frequency range of interest to us.

We have to pay special attention to Sennheiser K6, due to its good behaviour. It is the microphone suffering the less level loss, even in third position it is still able to pick all the harmonics above the background noise. The main reason is that this microphone is very directional, it means that it is able to pick the sound coming directly from its front side with higher level that the sound coming from their lateral sides. Then, only the noise coming from the same direction of the signal is going to interfere the signal, but not the surrounding one.

It specially helps to reject the wind, as we can see since it shows one of the lowest low frequency noise level.

Making a general view of the open air environment, it is a very noisy ambient, with high level of background noise. This background noise has random character, being able to influence any part of the frequency spectrum as we can see in the following two examples:

At third position, in Sennheiser MKH416 pink noise frequency spectrum, this background noise hit mainly mid frequencies.

At third position, in DPA 4011 Studio pink noise frequency spectrum, the effect of background noise is specially visible at low frequencies.

In open air, at low frequencies, background noise can be moreover intensified by the impact of wind. Being the main difference between them that background noise has random character, but it is more or less constant in time, while wind is totally unpredictable.

An extreme example of an outside noisy environment is a beach, where the sea waves produce a high level of noise and moreover, it uses to be a windy place. To try to control this hostile situation, the recommended solution is using Lavalier microphones, since they are protected from the wind by the body of the speaker and also they are very close to the sound source.

Unfortunately, sometimes it is not possible to use this kind of microphones because, for example in video recording, they must not appear on the shot. In those cases the best option is to use a shotgun microphone, since they are the only ones that continue picking the harmonics clearly even from the third position. It means that they are the most able to focus on the sound source and reject the surrounding noise.

In our test, specially the Sennheiser K6 shows well this feature. From first to third position Sennheiser experiences a level loss of only 1,6dB, which is a great behaviour for make sound pick from away sources.

Also Sennheiser MKH416 shows a good behaviour from this point of view, since it is a shotgun microphone too. But the most remarkable is the Newmann RSM191-M, because it is not a pure shotgun microphone, but it has only 3,4dB loss from first position.

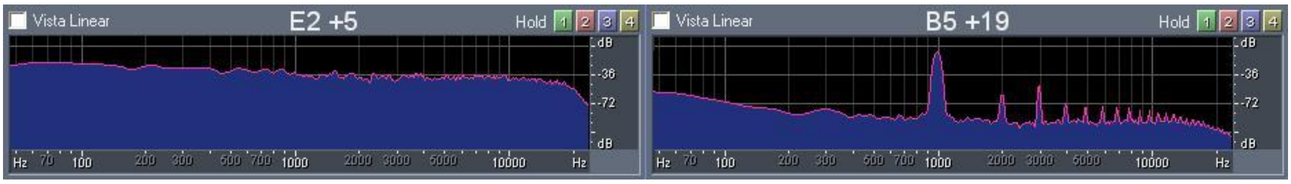
This is because, as we see from the technical specifications, this microphone has a particular set-up. In fact, the device is composed of two microphones: the RSM191-M and the RSM191-S, being the first one directive and the second one bidirectional. It allows making stereo takes with a single device, combining the signals of the two microphones, but for the effect to result, the RSM191-M has to be quite directive to avoid the lateral sound, which is picked by the RSM191-S.

This also explains why Newmann RSM191-S picks as high noise level as we can observe from its tone frequency spectrum of the third position. Being bidirectional, it picks better lateral sound than the one coming directly from the source.

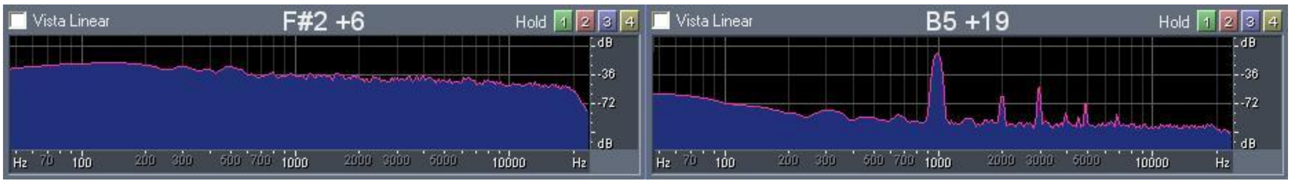
III. RECORDING STUDIO .

First position

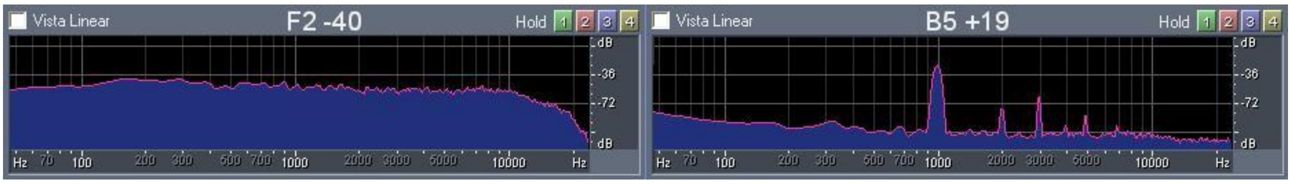
DPA 4006 Studio: Tone -7,1dB



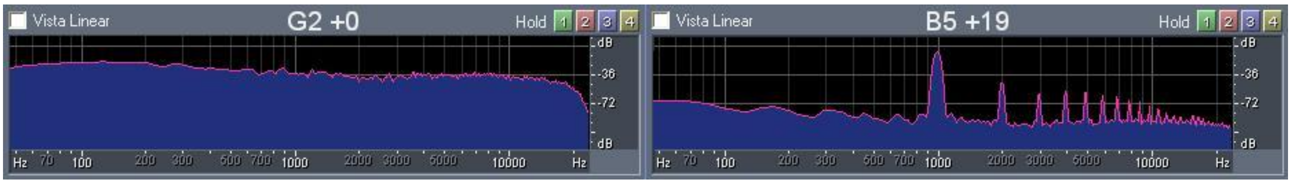
DPA 4011 Studio: Tone -8,3dB



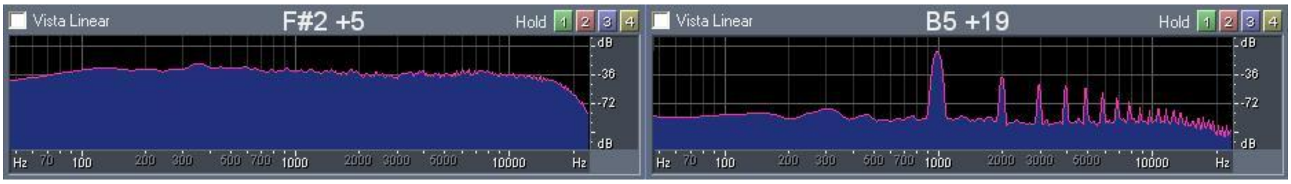
Electro Voice RE50: Tone -24,3dB



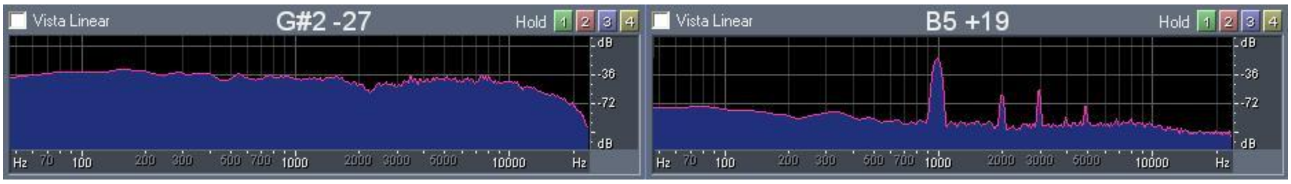
Sennheiser MKH416: Tone -6,6dB



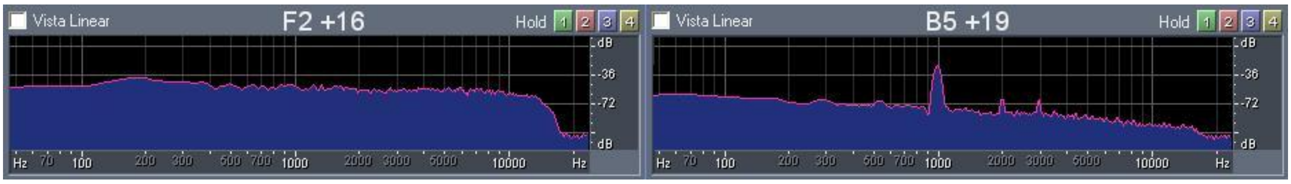
Sennheiser K6: Tone -6dB



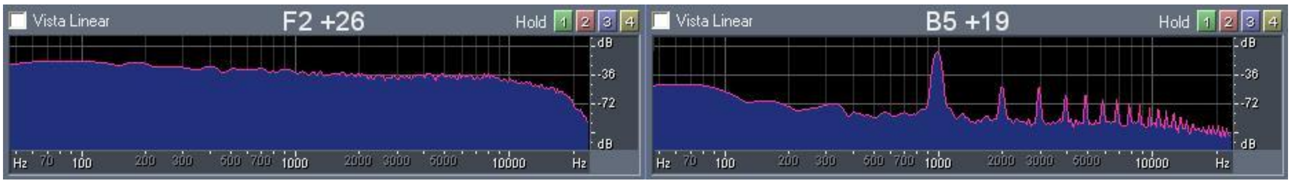
Audio Technica 871R: Tone -15,4dB



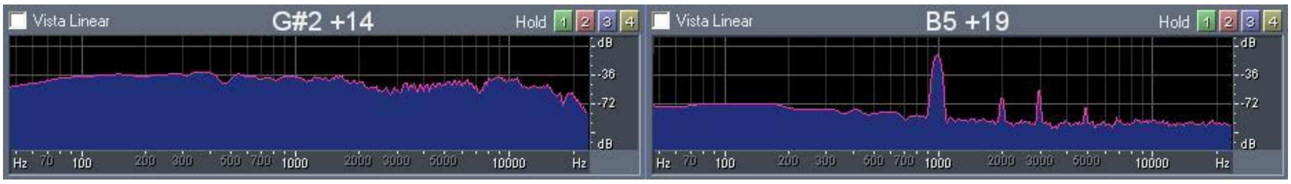
Shure PG58: Tone -24,8dB



Newmann RSM191 – M: Tone -6,9dB



Newmann RSM191 – S: Tone -9,8dB



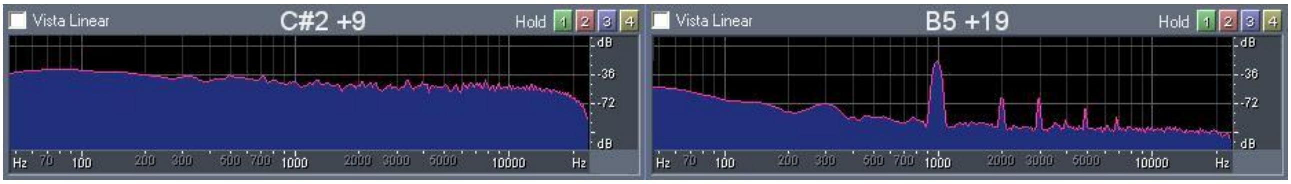
At this first position, microphones are picking the same level that they showed from first position of open air. It is because the source is the same, the gain has been adjusted to be the same, and at this short distance from the source, only the sound coming directly from the source is caught by the microphone. And, as in open air, sensitivity of each device makes that some of the microphones pick a remarkable lower level than the others.

Sensitivity is the level of electrical output produced by a microphone for a given acoustical pressure input, so one microphone is more sensitive when produces much electricity than another, for the same acoustic sound pressure level input. This is an important characteristic to consider if we want to give importance to details, as in studio.
In general, condenser microphones are going to be far more sensitive that dynamic ones, that is because dynamic microphones need much more energy to move the internal coil, which is, in fact, the dynamic transduction mechanism.
For this reason we can observe how Electro Voice RE50 and Shure PG58, the dynamic ones, display a much lower level.

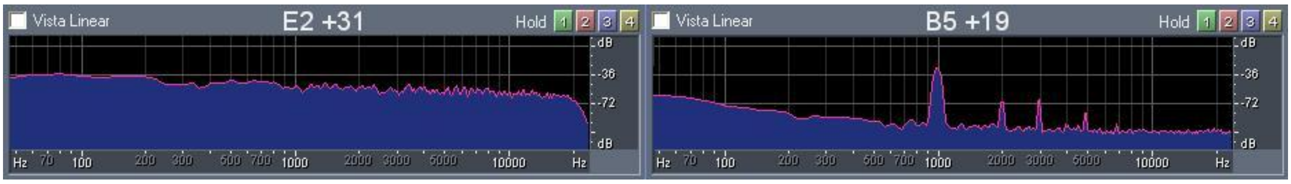
About pink noise frequency spectrums they are very flat, particularly at middle frequencies. This allows us to conclude that the environment is not affecting the signal, so these frequency spectrums display perfectly the frequency characteristics of each microphone. We can appreciate how, the only microphone influenced by the room at this short distance, the bidirectional Newmann RSM191-S, doesn't shows such a flat frequency spectrum, so it firms our conclusion.

Second position

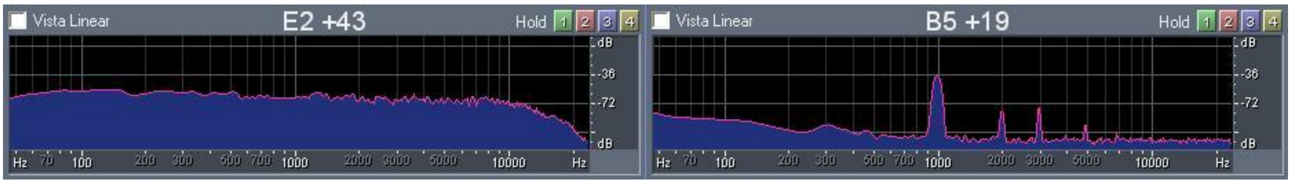
DPA 4006 Studio: Tone -18,4dB



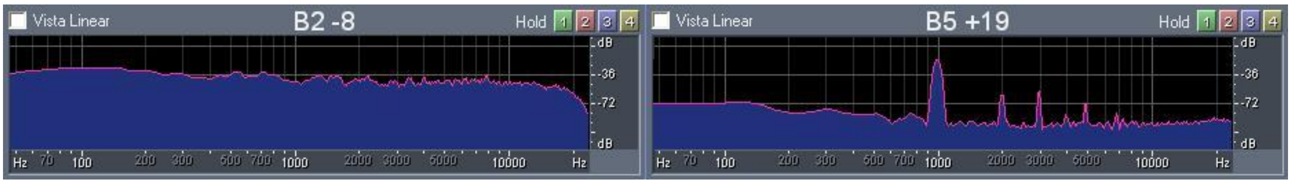
DPA 4011 Studio: Tone -26,7dB



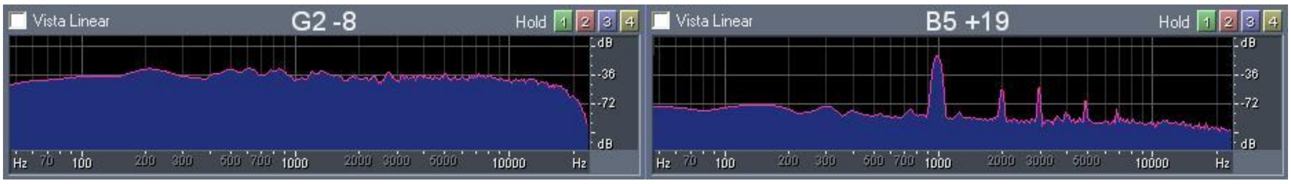
Electro Voice RE50: Tone -36,9dB



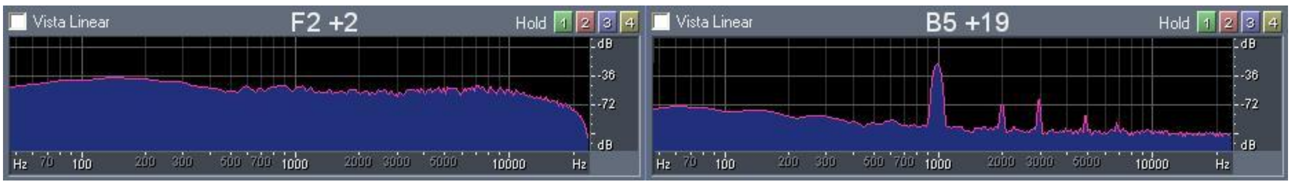
Sennheiser MKH416: Tone -16,7dB



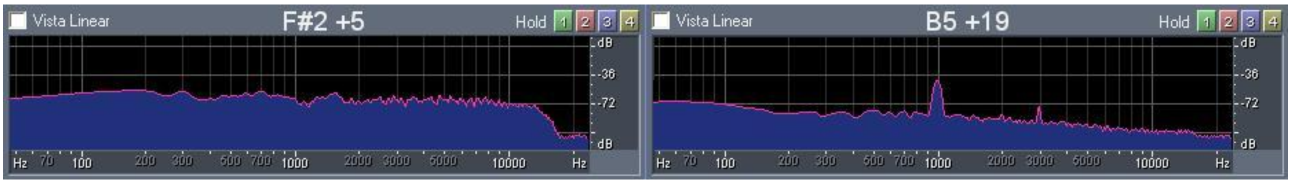
Sennheiser K6: Tone -11,1dB



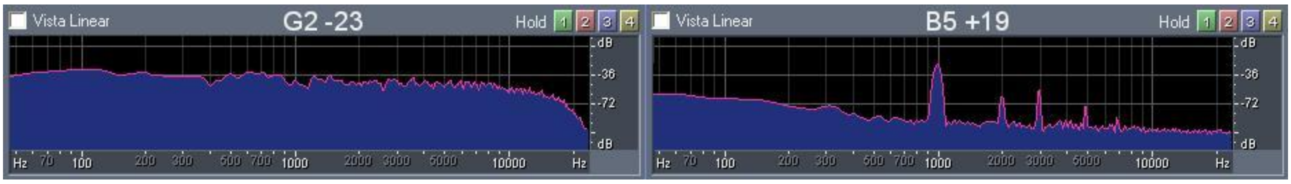
Audio Technica 871R: Tone -21dB



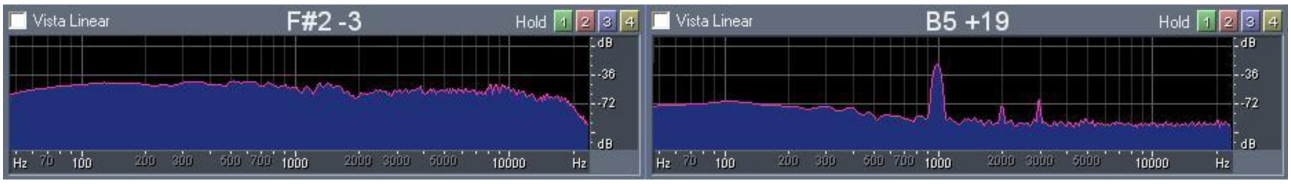
Shure PG58: Tone -43,6dB



Newmann RSM191 – M: Tone -22,6dB



Newmann RSM191 – S: Tone -22,8dB



At second position, in general, pink noise frequency spectrums maintain the flat shape that they had at first position, but we start to appreciate some irregularities.

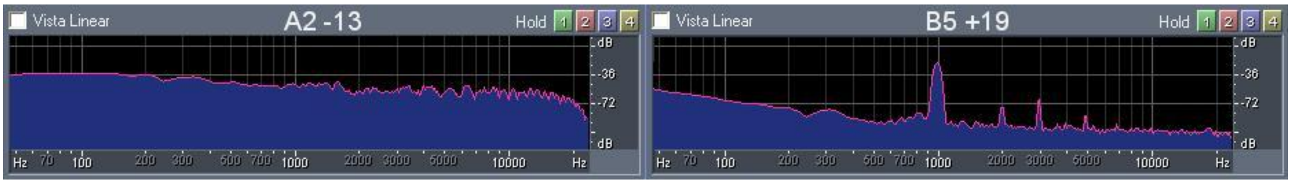
In the frequency range between 2000Hz and 10kHz we can observe some aberrations for all the microphones, and if we pay attention at the frequency spectrums of DPA 4011 Studio, Sennheiser MKH416 and Newmann RSM191-M, which are among the most precise devices, we can see that these irregularities seem to follow a similar pattern.
For this reason we are able to say that the room is producing these anomalies and if we are right, they will be increased in the third position, so we will extract conclusions studying the third position frequency spectrums.

Another fact to remark is the large loss of level for all the devices. Being inside a room, it means that this is a very dry room, that is, this room has a great capacity to absorb the sound. In fact, inside this room we can find absorbent material on the walls, together with absorbent curtains.

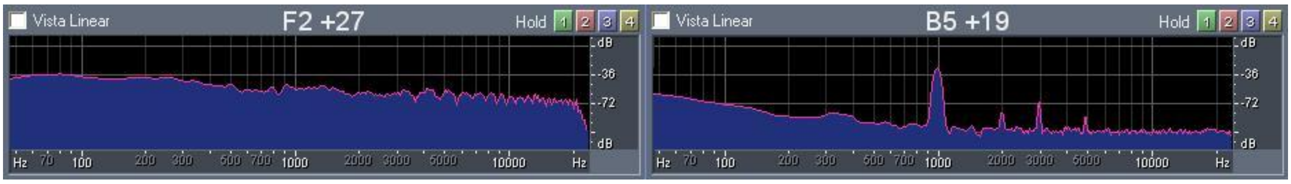
The last consequence that we can note is the loss of picked harmonics, specially for Sennheiser K6 and Newmann RSM191–M, but despite this loss on the quantity of harmonics, the level difference between them and the background noise is always superior than in open air. This evidences a reduction of the background noise level.

Third position

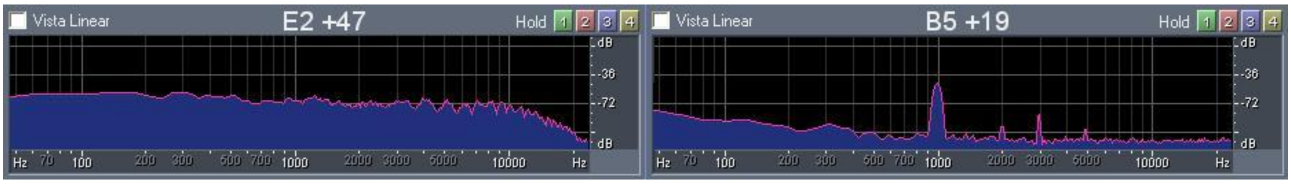
DPA 4006 Studio: Tone -20,6dB



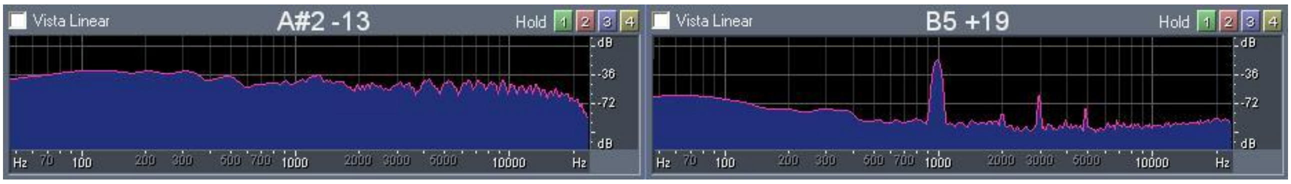
DPA 4011 Studio: Tone -28,2dB



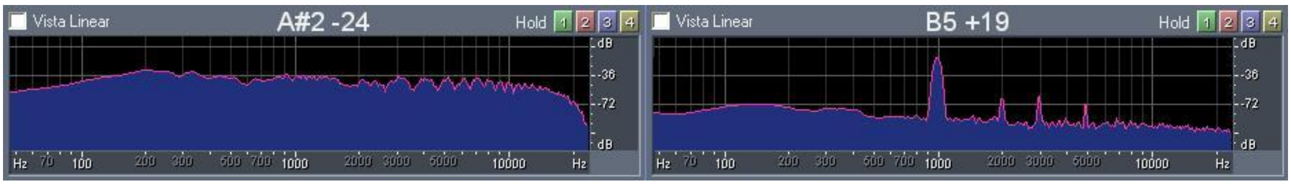
Electro Voice RE50: Tone -47,6dB



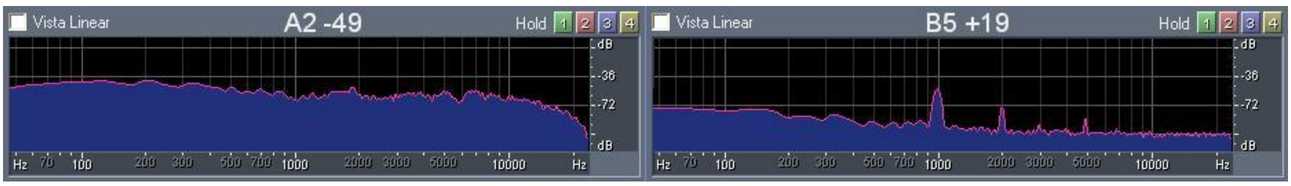
Sennheiser MKH416: Tone -17,5dB



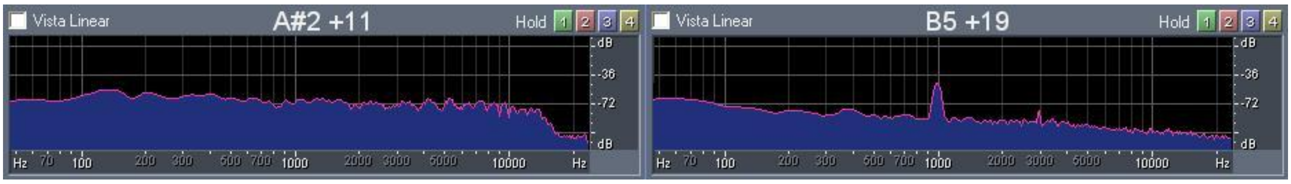
Sennheiser K6: Tone -12,8dB



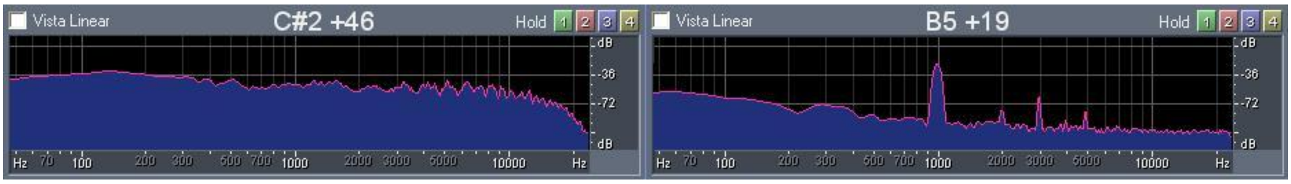
Audio Technica 871R: Tone -51,8dB



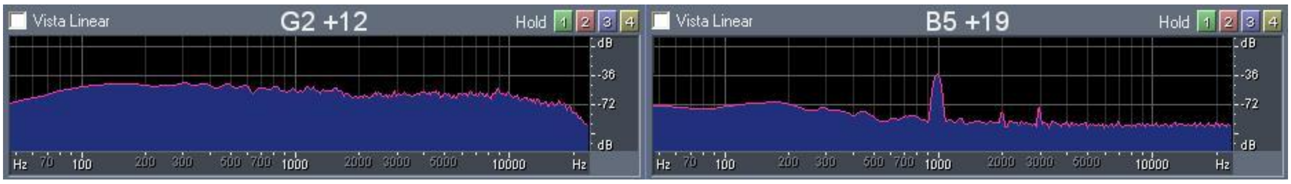
Shure PG58: Tone -46,3dB



Newmann RSM191 – M: Tone -21,8dB



Newmann RSM191 – S: Tone -35,1dB



Looking at the pink noise frequency spectrums, the first thing that we observe is the increase of the irregularities between 3kHz and 12kHz. As we are now outside of the critical radius, all picked sound is produced by the reverberation of the room, so it shows clearly all the effects introduced by the room.

Knowing the inside shape of this room, we can attribute this effect to one of the resources used for acoustic control. One of the walls and the roof are built as a sequence of surfaces at different distance of the opposite wall and soil respectively. This is made with the finality of avoiding the reinforcement of a single frequency, as more different distances, more different frequencies are in stationary mode making the ambient more diffuse.

The problem is that we have a discrete sequence of different distances, so anyway it is going to affect only certain concrete frequencies, producing the consequences that we can observe from the graphics.
But we cannot say that it is a bad design, in fact it is well done, because all the steps have the same level, so it is only an effect inherent to this kind of acoustic resource.

About the level of the tones, has to be noted that they have almost the same level than in second position. It is because we are now inside diffuse ambient, outside of critical radius, and theoretically, in any place inside the diffuse ambient, the sound level is the same, due to the independence from the sound source.

Making a general assessment for recording studio, the first important thing to point is the significant drop of background noise. This level reduction is mostly observed at low frequencies, where now we find levels around -50dB for third position, while in open air these levels were sometimes even higher than the tone itself.

Anyway, this is not a noise-free place, but almost. If we compare the tone frequency spectrums with the generated one (fig. 20), we can check how the noise level is almost the same that the generated, only suffering remarkable changes at low frequencies. And, moreover, the main difference between this background noise and the one that we found in open air is that here we don’t have the unpredictable effects of wind.

Despite the effect introduced by the room, commented in the second and third positions assessments, we can state that this is a well acoustical designed room. The main reason is that pink noise frequency spectrums continue being flat inside diffuse ambient (third position), meaning that the room have been well designed to treat all frequencies in the same way. It is specially remarkable the good behaviour of the room at low frequencies, as we can observe in the Newmann RSM191-S tone frequency spectrum.

Low frequencies are always a critical part of the spectrum to treat, since the different acoustical resources destined to this frequency range trend to be very selective. It means that these resources work well only for a small group of frequencies, but they are not able to treat in the same way all low frequencies spectrum.

A possible solution to avoid the frequency selectivity is to use several of these acoustical devices, each one tuned for different nominal frequency. Doing this, we obtain a similar effect that the discussed at third position for the medium frequencies, the more frequencies we can influence, the flatter spectrum we reach.

But it is interesting to notice that no matter how many acoustical cares you apply in a room, each room is always going to introduce its own characteristics on the signal. This let us to understand why some recording studios are preferred by some bands, because they give to the signal the sound that the band is looking for.

In our case, we have to learn that as important it is for video recording to choose an appropriate environment, it is for audio recording to chose the room that gives to the signal the sound that we want.

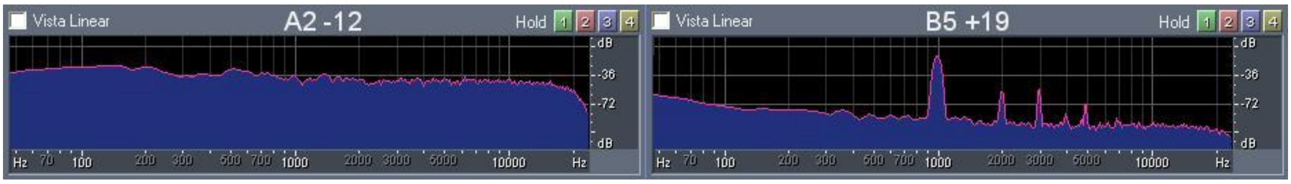
In audio recording, the kind of sound that we want to obtain has more to do with artistic preferences. Different music styles are related with certain kind of sound, but anyway each band decides about its own style. But if we are recording sound for a video production, we must take care that the ambient of the take agrees with the one present in the video sequence.

For example, if we are dubbing for a sequence of a walk in the park we can't do the recording session in a reverberating room. We must think that the video sequence is placed in the open air, being as we have seen before a non-reverberating environment, so we must obtain a completely dry sound.

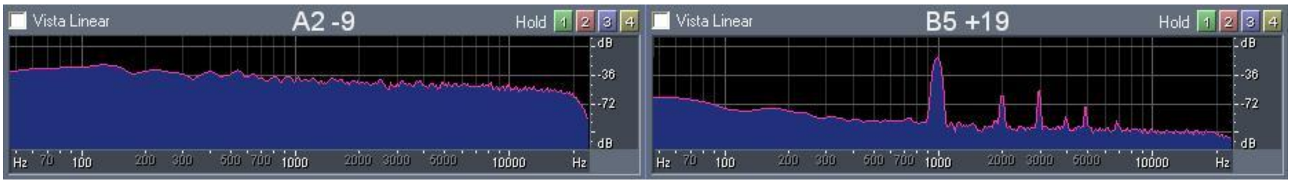
IV. LIVING ROOM

First position

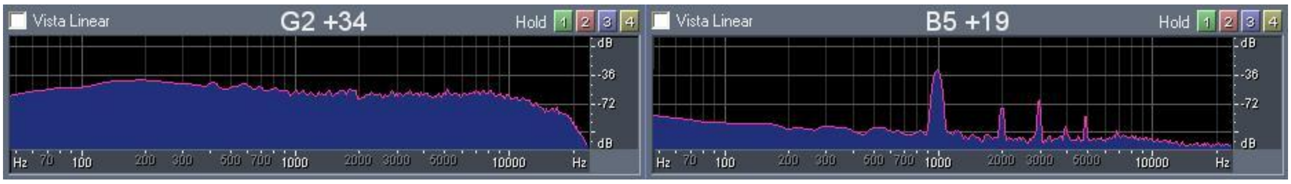
DPA 4006 Studio: Tone -11dB



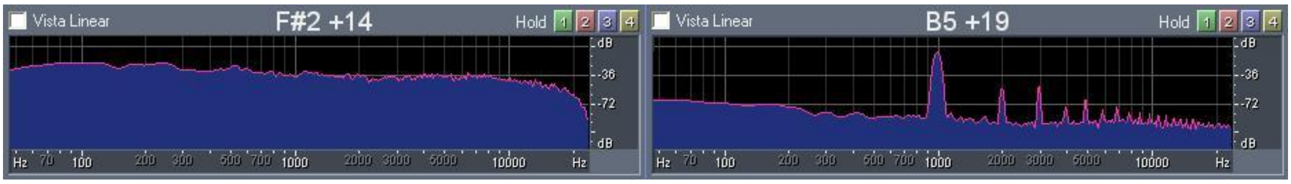
DPA 4011 Studio: Tone -14,3dB



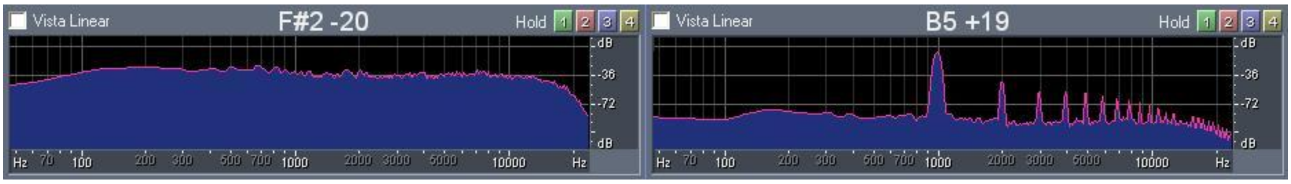
Electro Voice RE50: Tone -30,1dB



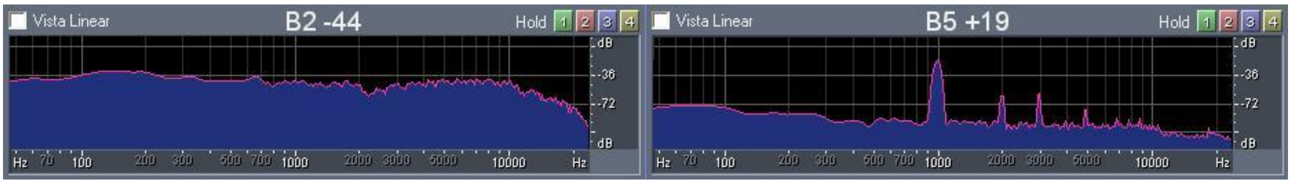
Sennheiser MKH416: Tone -6,9dB



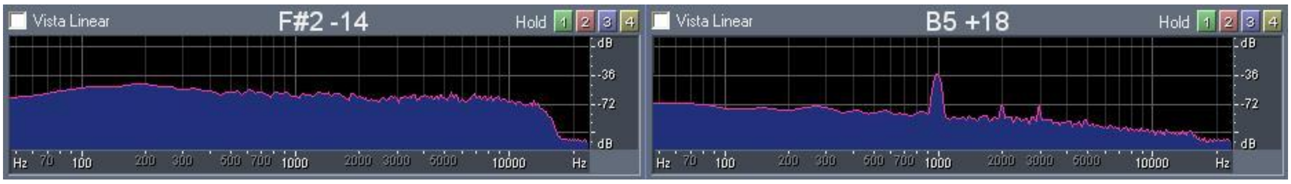
Sennheiser K6: Tone -6,6dB



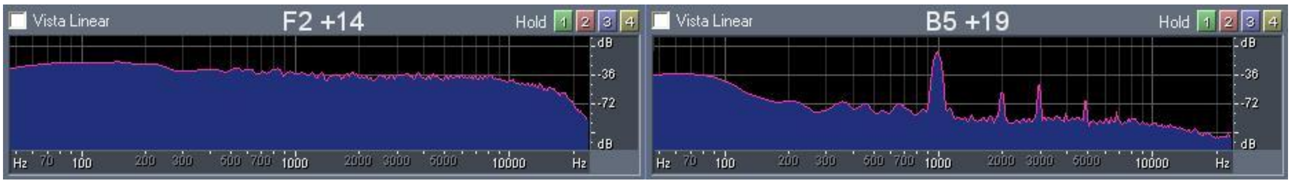
Audio Technica 871R: Tone -18,3dB



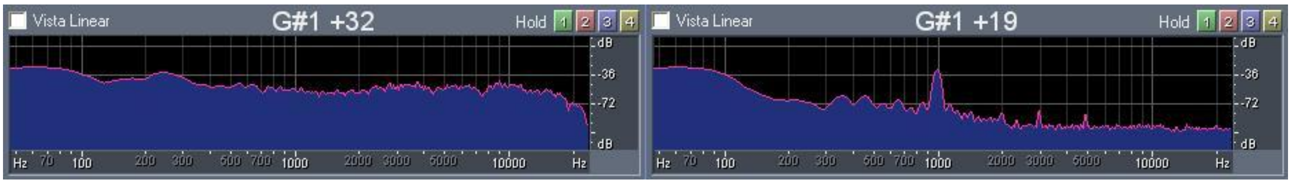
Shure PG58: Tone -34dB



Newmann RSM191 – M: Tone -7,3dB



Newmann RSM191 – S: Tone -29,3dB



The first thing to discuss is that, for this first position, tones levels are always lower than for open air or recording studio. Since the characteristics of the source have not been modified, we attribute it to the small error committed when placing the microphones, because in such a short distance, a small movement implies an appreciable change in the sound level.

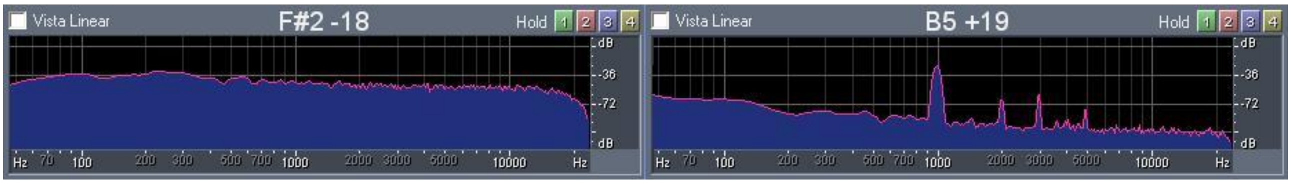
But we can state that these measures are valid since they keep the sensitivity relations. Despite levels are different from the other environments ones, the relation among these levels follows the same behaviour than in the other environments, and it is in fact, what we must perceive, which devices respond better than the others.

Pink noise frequency spectrums are, in general, flat, quite similar to the first position ones for open air and recording studio. It is normal because the sound has not been influenced by the room acoustics.

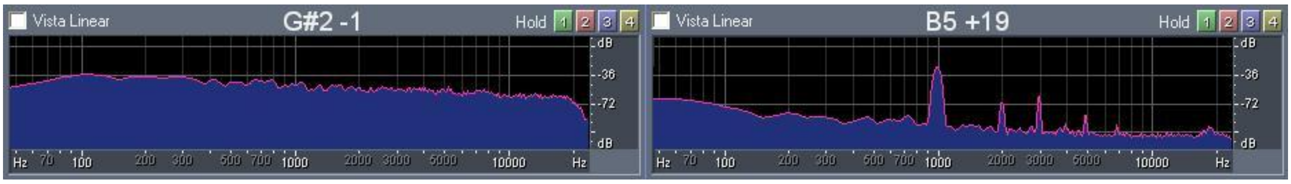
One point to remark is the high level of low frequencies for Newmann microphones. It could be attributed to the proximity effect, due to they are gradient pressure microphones and this effect is especially important in this kind of devices. Moreover, proximity effect consists just in an increase of low frequencies, below 100Hz, when the microphone is very close to the source. But before stating this, we must study the second and third position, where the proximity effect should not be present.

Second position

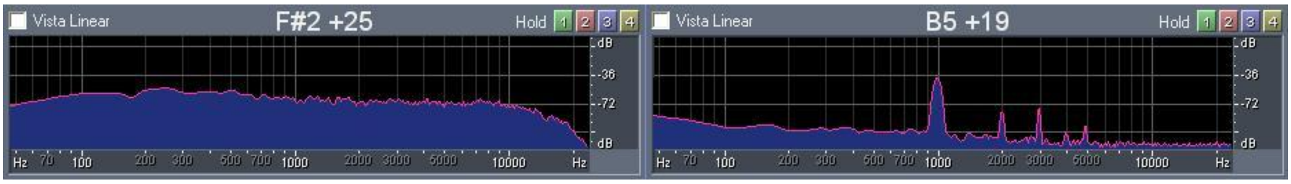
DPA 4006 Studio: Tone -24,8dB



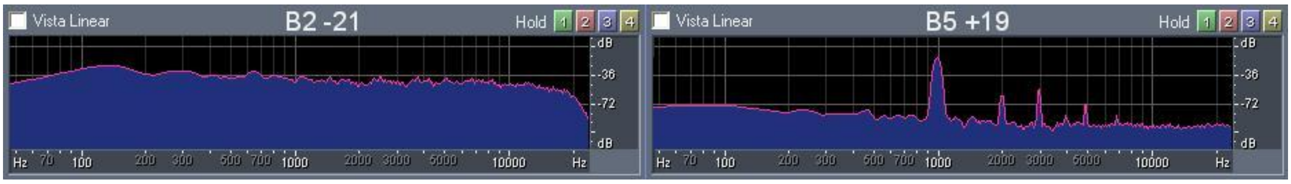
DPA 4011 Studio: Tone -25,5dB



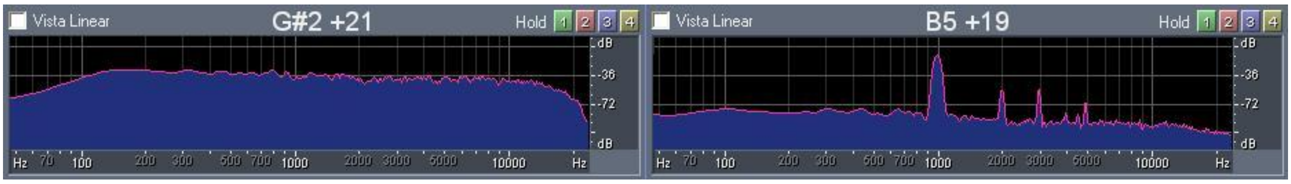
Electro Voice RE50: Tone -39,6dB



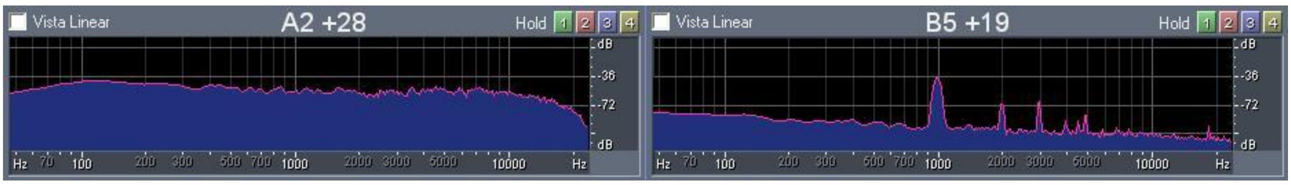
Sennheiser MKH416: Tone -14,6dB



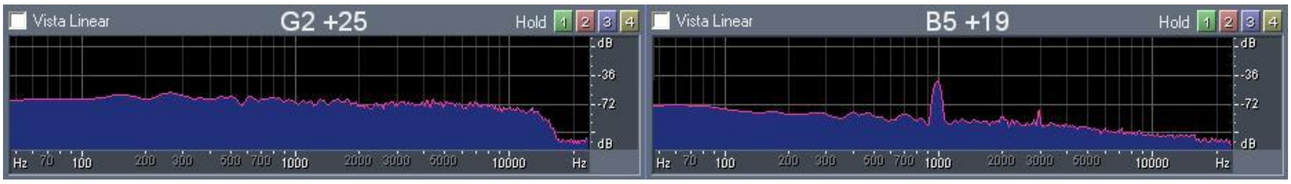
Sennheiser K6: Tone -10,4dB



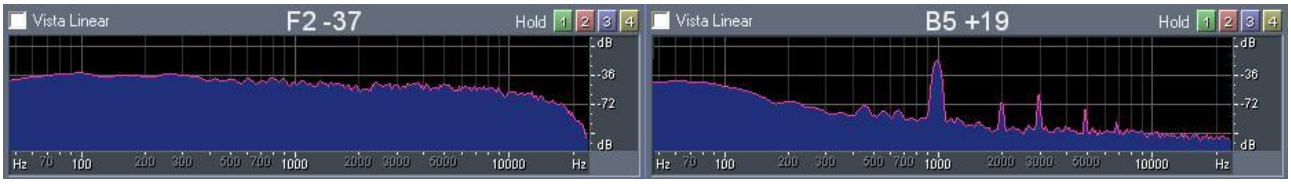
Audio Technica 871R: Tone -38,4dB



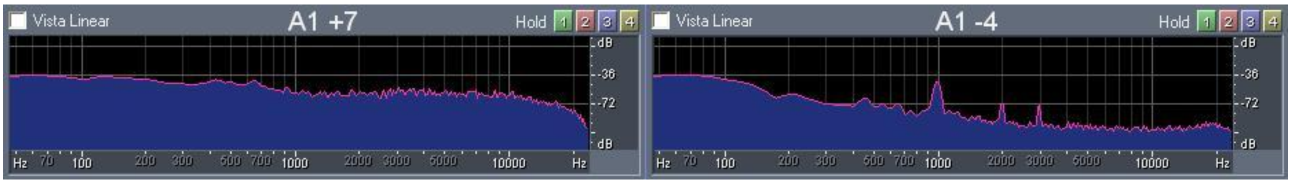
Shure PG58: Tone -44dB



Newmann RSM191 – M: Tone -17,4dB



Newmann RSM191 – S: Tone -45,3dB



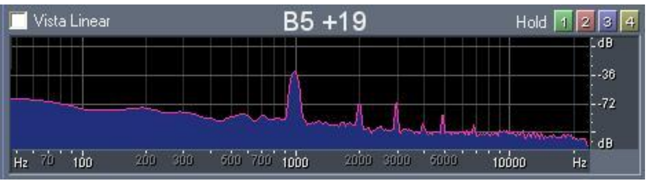
In general, all pink noise frequency spectrums start to display irregularities, as it happened for the second position of recording Studio. The difference is that in recording studio these irregularities seemed to follow a concrete patter while, in this room, they have random character.

One feature that we can appreciate are the low-cut filters, for example in DPA 4006 and 4011, Sennheiser MKH416 and Sennheiser K6. They are usual behaviours for microphones since only specialized measurement microphones are able to maintain a truly flat response for the whole frequency spectrum. The point is that we could not appreciate this effect from the open air spectrums, since they were affected by the low frequency background noise. This shows the importance of the environment for doing acoustical objective test, because, as we see, the ambient can totally distort our measures.

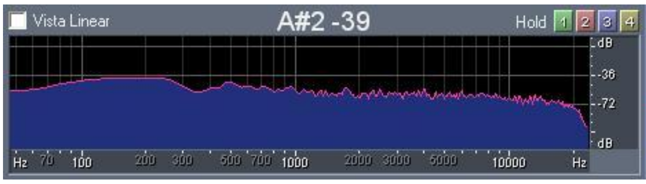
For general recordings, usually we can assume these ambient made distortions until certain level, not so restrictive as for measures, but we have to be aware that this effect is happening and establish the limit of distortion that we can assume.
About the tone levels, they have suffered a general drop between 10 and 12dB from the first position, except Sennheiser devices, which as always, follow their own behaviour.
The last point to discuss is the specially low level picked by Newmann RSM191-S. Remembering that this device has a bidirectional directivity pattern, the vast majority of the picked sound comes from the room reverberation, and displaying such a low level, it let us to state that this is a dry room, contrary to what initially expected.

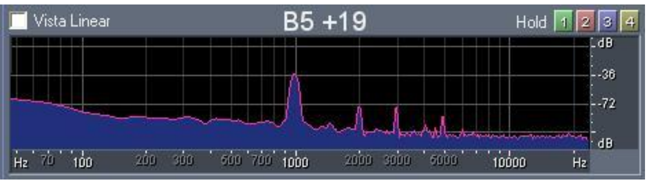
Third position

DPA 4006 Studio: Tone -30,3dB

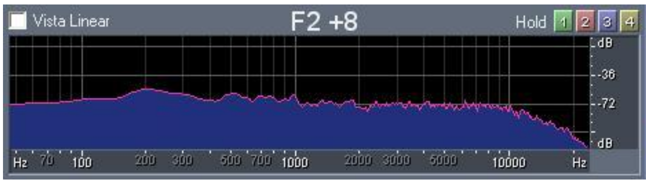


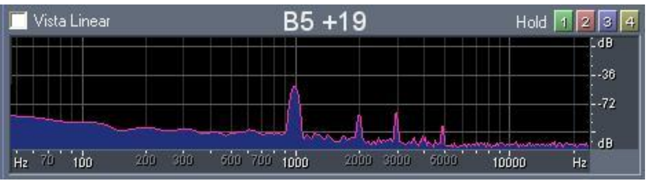
DPA 4011 Studio: Tone -34,7dB



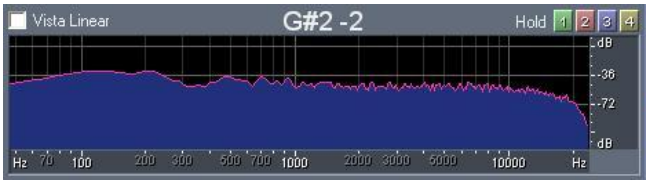


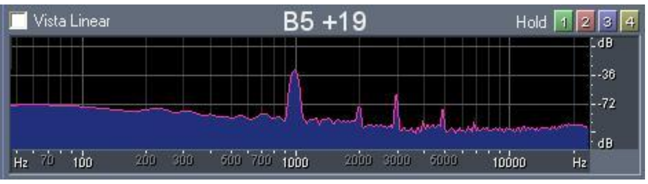
Electro Voice RE50: Tone -50,3dB



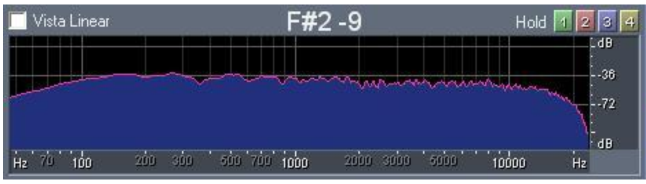


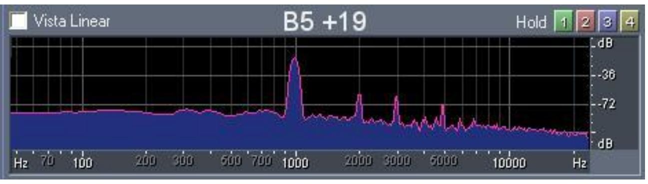
Sennheiser MKH416: Tone -29,7dB



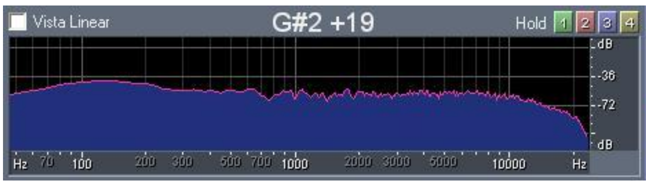


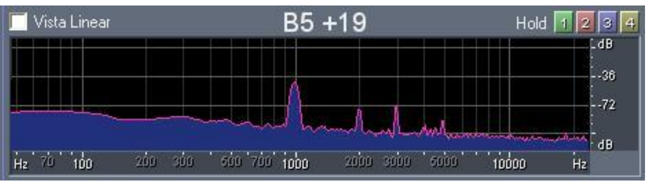
Sennheiser K6: Tone -14dB



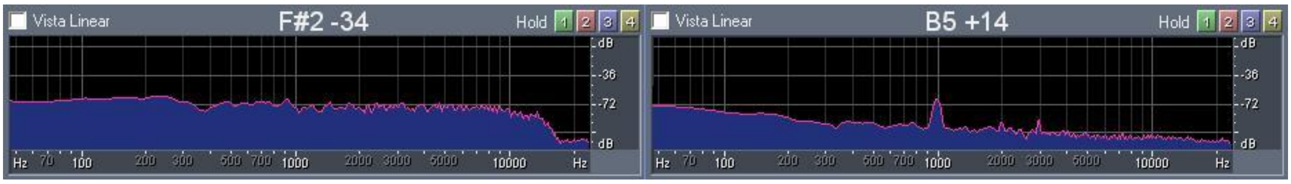


Audio Technica 871R: Tone -44,4dB

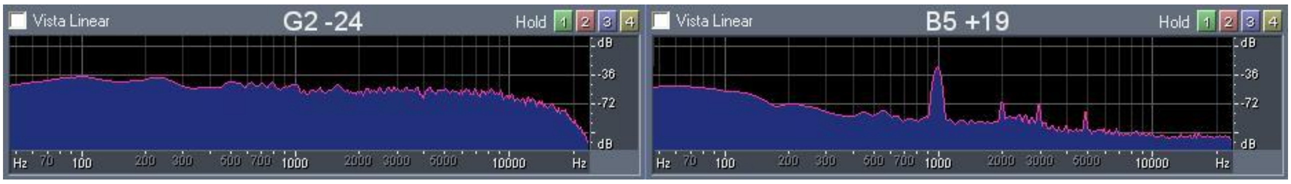




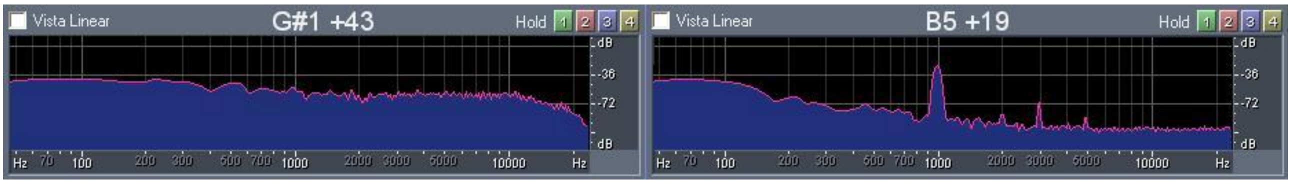
Shure PG58: Tone -66dB



Newmann RSM191 – M: Tone -26,8dB



Newmann RSM191 – S: Tone -24dB



From the pink noise frequency spectrums, we can observe that they maintain a flat tendency, but full of irregularities comparable to open air ones. This is because the room is not acoustically designed, so we can expect it no to be able to create a perfect diffuse ambient, intensifying some frequencies and reducing some others.

For example, we can observe a frequency gap in the range of 400Hz for almost all microphones, which would not be suitable for a recording studio, but completely usual in a non acoustical prepared environment.

The most remarkable result is the level drop for all the devices, comparing with second position levels. This fact shows that, despite we didn't expect that from a non-acoustical treated room, this is a dry room.

The drier is the room, the longer is the critical radius, so we can state that in the second position, microphones were still inside the direct field (before the critical radius), and from that position, the signal level continues decreasing until the critical radius. This explains why the level of third position is not so similar to the second position one as it was for recording studio.

Finally, we only rest to remark that Newmann RSM191-S is the only microphone that picks a higher level than for the second position. We attribute this to the bidirectional feature, that allows this device to catch better the reverberated ambient of the room, although it is not very pronounced.

In a general view of this third environment, we have to say that, at first, it was selected with the intention of showing a quite reverberant environment, but studying the obtained results it may be considered a dry room.

The lesson that we must learn from this handicap is that we always must check the equipment and the environment conditions where we are going to work in. For every recording we must ensure that the acoustic of the space provides the conditions that matches our project. For example, we must be sure that we are working in a dry room when dubbing voices for an open air images, otherwise the sound won't be coherent with the image.

This room was set as reverberant at first because it does not have any specific acoustic treatment, all the walls are reflectors and as absorbent materials we can just find a thick curtain and two beds.

Then, the main factor that explains the low reverberation is the small size of the room. In such a small rooms, sound waves travel short distances through the air between two reflections, so they fade quickly due to the great amount of reflections suffered in a short period of time. Also due to this small size, the two beds and the curtain are an important part of the total surface of the room, having a significant impact in the absorption coefficient.

This feature of the small rooms is explained by Sabine's formula for reverberation time:

$$Tr = \frac{0{,}162 \cdot V}{\alpha \cdot S}$$

V is the total volume of the room.
α is the absorption coefficient.
S is the total area of the room walls, floor and ceiling.

Maintaining $\frac{0{,}162}{\alpha}$ as a constant value, if we calculate the relation $\frac{V}{S}$, it is bigger for bigger rooms, so we can prove that reverberation time trends to be shorter in smaller rooms. Obviously, it depends on the dimensions relation, meaning that for rooms where one of the dimensions is quite larger than the others, the proportion $\frac{V}{S}$ is decreased giving them a different behaviour. But for rooms like the one that we are studying, with a nearly cubic volume, the stated argument can be taken as true.

The other important effect for this environment is the irregular shape of pink noise frequency spectrums. As this room has not any acoustic care, out of the critical radius there is not a real diffuse ambient, so we can find cancelations in certain frequencies or reinforcements in some others, producing the irregularities that we observe in the different spectrums.

Specially remarkable is the level decrease that we can appreciate at 300Hz in each frequency spectrum for the third position. No appearing in the first and second positions, it is clearly caused by the room, representing a wave cancellation at that frequency.

In rooms like this one, where there is not a real diffuse ambient zone, the effect of the room in the sound highly depends on the position of the source and the microphone inside the room. In the position that we set the test assembly, the most remarkable effect is the mentioned level drop around 300Hz, but if we change the positions of the source and/or the microphone the effect may be totally different. It is because the reflections on the walls produce specific wave cancellations or reinforcements for each point of the space.

This is one of the main reasons for acoustically prepare the rooms where sound is going to be recorded. Creating a truly diffuse ambient ensures that sound is going to keep the same characteristics for every point of the space, so we can set the recording equipment as best for us while maintaining more or less the same acoustical conditions.

V. PARTICULAR ASSESSMENT FOR EACH MICROPHONE

In general, if we pretend to assess the response of each microphone, the most logical would be to look at the results obtained from the first position, for all three environments, but it depends on the feature that we are going to discuss though.

Of course, the first position is going to show us the most similar signal to the response of the microphone, but for instance, in microphones like Sennheiser MKH416 and Sennheiser K6 will be interesting to pay attention to their features when used in second or third position, since they are going to be used in those positions at the vast majority of the situations.

It means that we must mind all three positions with the finality of extracting useful information for the larger amount of different sound picking situations as possible.

About the environments, in order to assess the capacity of the different microphones picking the details of the sound, we must pay attention to the results obtained from the enclosed environments, where we find an acceptable low level of background noise.

Instead, the outdoors environment will be useful to evaluate some features important for specific microphones. For example, we must think that if devices like Electro Voice RE50 have been specifically designed for interviews and outdoors applications, it looks interesting to pay attention to their response in this particular situation.

DPA 4006 Studio

It is a studio microphone that offers a great quality for the whole spectrum, which can be appreciated from the wealth and clearness of the sound and the loudness and body of the low frequencies. At the opposite side of the frequency spectrum, for high frequencies we must remark the flat response that this microphone provides.

Due to the omni-directional property, this device allows us to hear clearly the reverberating ambient at third position of recording studio and living room. This causes that as we start separating the microphone from the source the intensity level for the low frequencies starts rising up comparing with the level of the high part of the spectrum. It is due to the characteristics of the room, since every absorbent material we can find inside is going to cause some effect from midrange frequencies up, but not on the low ones because as we have mentioned before, they are the most difficult to control and attenuate.

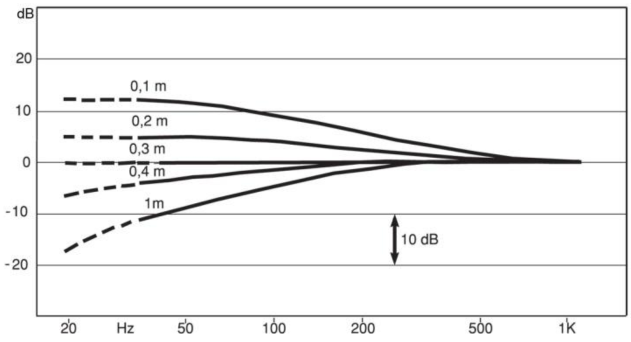
For these reasons this is a recommendable microphone to use in studio. Using it in a multi-microphone take, it is supposed to be placed very close to the source, so we are obtaining the purest sound of the one. If we like the acoustical characteristics of the room, we just have to increase the distance between source and microphone until we obtain the desired amount of reverberation, and the great quality of the DPA 4006 ensures that we will not lose any detail from the source, obtaining a very clear sound with the body provided by the natural reverberation.

DPA 4011 Studio

This microphone shares part of the great characteristics of the DPA 4006, producing a defined sound for the whole frequency spectrum.

One sensible difference, comparing with DPA 4006, is that DPA 4011 produces a slightly darker sound, being this fact attributable to the different directional pattern. For the second and third positions the frequency responses of both devices are very similar, since as we mentioned before, the low frequencies reinforcement is due to the effect of the room, but for the first position is easy to appreciate the slight slope falling towards the high frequencies in the frequency spectrum of the DPA 4011. Being a cardioid microphone, when it is placed next to the source, it suffers the proximity effect, which produces the low frequency reinforcement that we can appreciate.

This is a very important effect that we have to have in mind when using a directive microphone closer to a source. The more similar is the directivity pattern of the microphone to a bidirectional one, the more noticeable will be this effect.



23. Proximity effect for DPA 4011.

In some cases this can be profitable, because some microphones have a lack of level below 150-200Hz, so placing them in the right position, the proximity effect helps us to achieve a flat response in that side of the spectrum. But on the other hand, we have to use this trick with caution because if the proximity effect is too flagrant, the obtained signal can become too much coloured and even exceed the saturation level, causing an undesired distortion.

This microphone is also recommendable for the same use than the DPA 4006, being interesting the comparison between them because it let us see that, despite all them provide high quality signal, using different devices we can obtain a different sound in the studio.

Electro Voice RE50

The first fact that we perceive hearing the EV RE50 tracks is the high level drop. As we have discussed before, it is due to the dynamic nature of this device, because it makes the sensitivity lower. In order to treat most concrete data, we can see from the technical specifications that DPA microphones are in the order of 10-30mV/Pa, while EV RE50 has a sensitivity of 1,8mV/Pa.

This fact does not have to be always negative, because we can take profit of this particular feature by using this kind of device for sources producing high level of sound, like guitar amplifiers or drums. But when doing it, we must mind not only the signal level, but also the general frequency response of dynamic microphones.

We are not going to discuss the construction details of the different kinds of microphones, but it is necessary to understand that the transducer of a dynamic microphone is necessarily heavier than the one of a condenser device (dynamic one consists on a metallic coil, while the mobile part of a condenser transducer is simply a very thin and small metallic plate). This means that it is more difficult to quickly change the direction of the movement of the coil, because it has greater inertia.

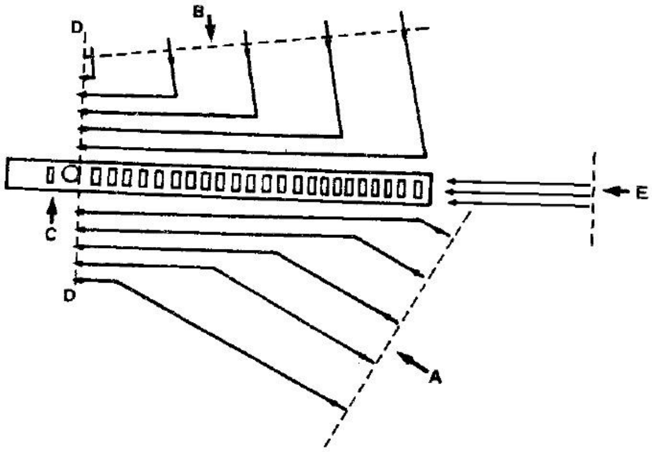
The direct effect in the sound signal is a lower response on the high frequencies, since they are the ones who quicker move the diaphragm. We can easily check that this theoretical explanation agrees perfectly with the obtained results. Observing whichever of the EV RE50 results we notice a level drop from 10kHz up caused just by its dynamic nature. It means that we are always going to obtain a rounded sound, with less definition in high frequencies when using a dynamic microphone, so we must think if this agrees with the source that we pretend to record, or it would be better to use a cardioid microphone but taking care of the maximum admitted level.

On the other hand, we can also use a dynamic microphone for standard sources and take profit of its low sensitivity in another way. As we can hear in *track 03*, using this microphone in adverse environments, but very close to the source, the low sensitivity helps to reject the background noise. If we compare this track with the ones corresponding to the highest sensitivity microphones, like *track 04*, we realise that this characteristic becomes really profitable.

Having explained all these features, it is very recommendable to use this microphone for voice recording, since it has been designed with this finality. And for this, the best way to use it is taking care of the distance between the microphone and the speaker in order to find an appropriate balance between the level of the voice and the level of the background sound.

Sennheiser MKH416

The first feature that we can observe in this microphone is the physical construction, being longer and thinner than the rest of the devices, except the Sennheiser K6 since they are sharing the same characteristics. This especial construction has the finality of focus on the sound coming directly from the axis, while the waves coming from the rest of directions are attenuated due to the interferences that this especial shape creates.



24. Interference principle in shotgun microphones.

The waves coming directly from the axis "E" are arriving to the diaphragm "D" in phase (at the same precise instant of time), so they do not suffer any interference. As the direction of origin starts moving away from the axis "A", the wave arrives to the diaphragm through different paths, which present some little length differences. It causes light interferences that start attenuating the high frequencies. When the direction of origin maintain a high angle with the axis of the microphone "B", the wave arrives to the diaphragm through different paths, which present very different lengths, causing the attenuation of the vast majority of the frequencies. This is the way the device is able to attenuate the sounds coming from undesired directions.

When we hear the *track 22*, where the MKH416 has been placed around three metres of the source in open air, the main feature to remark is the good intelligibility of the speech, explained by the good response of this microphone in the mid frequencies. This feature is a combination of the directive design and the high quality of the transducer, because we can appreciate it not only on the open air, but also inside indoor environments. For example, from the first and second position of recording studio, *track31* and *track40*, we can appreciate how all voice details have been picked perfectly.

As we move the microphone away from the source, we can note a slight loss on low frequencies, but the specific design of the microphone makes possible to obtain a drier sound, with less reverberation than the other microphones, for third position.

The Sennheiser MKH416 is suggested for using in situations where the microphone cannot be placed close to the source, looking that even at distances around 1,5 metres, it is able to pick all signal details.

Sennheiser K6

This is also a shotgun microphone with very directive polar pattern. The main difference is that Sennheiser MKH416 displays a good behaviour for the whole frequency spectrum, while Sennheiser K6 has a remarkable bad treatment of low frequencies, being it appreciable in any of the recording of this microphone; *tracks 05, 14, 23, 32, 42, 50, 59, 68* and *77.* It makes that the sound recorded with this device is lacking in body.

For this reason, Sennheiser is not appropriated to pick music, because it is going to lose an important part of the spectrum. But it is very recommendable to pick voices when it has to be done at long distances, this is because, as we can hear on *track50*, this microphone is able to reject all the reverberation of the recording studio even at this third distance.

We have focused on the enclosed environments for assessing the quality of the picked sound because open air takes are quite affected by the wind. If we would have to use this or any other microphone at outdoor conditions it is very recommendable using any complementary devices available for them, like the MZW 66 foam windshield.

25. MZW 66 pro foam windshield.

We have not used it in this project because we pretended to check the different devices in the different environments exactly in the same conditions and without any supplement, but for a real take it would be essential.

Audio Technica 871R

This device produces a sound level comparable to the dynamic microphones resulting specially bad in open air, because as we can hear in third position, *track 24*, the noise of the wind almost completely masks the signal. Probably it is due to the fact that the AT871R has a bigger surface exposed to the air, comparing with any of the other tested microphones. Initially, it is a bad result for this microphone, but we must have in mind that this is not a usual position for this device.

For its characteristics it has been built to be used up on a lectern, it means close to the speaker. For this shape it has also been designed to avoid the reflected sound, so we must think that it is going to offer its best results inside an enclosed environment, no at open air.

Inside an enclosed environment it produces a higher sound level than any dynamic device and moreover with a clearest sound, as we could expect from a condenser one, but it does not reach the good qualities of any of the other condenser microphones, being especially remarkable the difference in level between this one and any other one.

Its frequency response is quite flat, remarking a good behaviour at low frequencies. The main observable lacks are an important high cut filter starting at 10kHz, and a slightly more irregular spectrum than DPA ones for middle frequencies. This device has been designed to be used in speech takes, therefore the high cut filter is acceptable since it does not affect human voice frequencies.

After having assessed the results, we can state that this microphone is only recommendable to be used in the application that is has been created for. It would not be practical if we try to use it as a hand vocal microphone because of its shape, and on the other hand, it is not recommendable for music recording because despite it is a condenser one, as we have checked, it does not reach the quality level of the rest.

Shure PG58

This is a dynamic device, as Electro Voice RE50 is, and as we can hear in its takes, the sound level is also lower than for the rest of the microphones.

Specially for the second and third positions, even inside closed rooms, the signal level is so low that it is not useful for hardly any application, due to the great amount of lost details. Comparing with EV RE50, they offer similar signal level in almost any condition, but the main difference is that Electro Voice device is more able to reject the background noise. Observing the results of the recording studio, we can see how usually the EV RE50 is the device picking the lowest background noise level, while the Shure PG58 shows a level comparable to the one obtained with condenser microphones. It is not quite bad, but it is since the level of the wanted sound is not comparable with the one got with condenser microphones, obtaining the lowest signal to noise ratio.

About frequency response, it is quite irregular. As we could expect it has some deficiencies in both low and high frequencies, being normal for a dynamic microphones, but comparing again with EV RE50, the Electro Voice device presents this feature in a most natural way. We can easily check it just in the first look of the frequency spectrum, where we see that the curves for EV RE50 are much more soft. It is especially remarkable for high frequencies, where we can appreciate a completely well defined high cut filter in the Shure device.

For all these reasons it is an appropriate microphone to use for vocals, since the vocal spectrum is not going to be affected by the deficient response in low and high frequencies. Moreover, when using it for vocals it must be placed very close to the source for two main reasons. First, because using it in this way helps to obtain the highest signal to noise ratio. Second, because we can take profit of the proximity effect, since this is a cardioid microphones, helping to offset the deficiencies in low frequencies.

The main advantage of this microphone is its wireless property, which makes it perfect for handheld use. For this purpose, the bad treatment of low frequencies is good, since it helps to reject the mechanical noise produced by movements and hits on the hands.

But apart from this good feature, there is not any other reason to use this microphone in professional applications. Despite it belongs to a reputable brand, and it shares the same number than the vocal standard device, the Shure SM58, the PG58 is not sharing its good features.

Newmann RSM191 – M

This microphone displays a high level and a very good sound quality, remarking that for open air, it is able to pick even the background noise details as we can appreciate from *track 08*, before the musical fragment starts. Even inside the studio, it picks the background noise in a considerable level, comparable to DPA4006 or the two shotgun microphones. Thereby, we also appreciate the good frequency response from the closed environments, were this microphone shows a sound with even more body than the DPA devices, especially in first position, *tracks 35* and *62*.

At the furthest positions, this device displays a better behaviour than DPAs against reverberation, and moreover with higher sound quality than Sennheiser microphones. The reason is that actually, this is also a shotgun microphone, as we can observe from its construction. Together with a high quality condenser transducer, it is able to pick a very detailed sound even at far positions.
The main advantage over the other two shotgun microphones tested is its best frequency response, making the Newmann able to be used for music recording. In fact, the ensemble Neumann RSM-191 has been designed with the purpose of picking music using a specific stereo pair technique that will be explained in the next chapter.

As part of the set, the "M" capsule has the finality of picking the sound from the central source, if we think in an stage, so it needs a highly directive pattern, and as we have checked, it meets expectations. Moreover, it is usually going to be used at a relatively long distance to the source, according to the MS stereo technique, which explains the needless of a high sensitivity.

This is a very good option to use in studio, due to the great sound quality and body, and moreover with good shotgun features, which make this microphone very versatile, keeping a great sound in every situation.

Newmann RSM191 – S

This device is the other part of the RSM191 set. A bidirectional capsule that maintain the high quality of the RSM-191-S transducer. While it is a compact set, as we can observe in the figure 8 (pag.13), it has been considered to test the two capsules individually because it let us compare them better. Specially interesting is to compare the sounds picked with this RSM191-"S" device with the sound from any of the other microphones, because it show us how important is the directivity pattern of the microphone in the characteristics of the captured signal.

Due to its bidirectional characteristics, it does not shows great features with our set-up, mainly because it is not raised to pick the sound coming from our source direction.
In spite of this, we can state that it is a good device since it uses the same technology that RSM191-M, and shows good bidirectional behaviour, being able to pick the acoustic of the room from second or even first position in living room, *track 63*.

The rest of the microphones used to capture the effect of the room in the form of an increase of the reverberation and a modification of the temporal distribution of the different reflections of the signal, but we still had a lack of spatiality due to the monoaural sensation.

With the "S" capsule of the Newmann, the signal that we obtain is a combination of the sound coming from one side of the stage and the sound coming directly from the opposite side (supposing that we place the RSM-191 set pointing to the centre of the stage). Then, with a simple processing of the signal, we can separate the sound from each of these two sources in two different tracks allowing a stereo listening.

For these reasons, it is very recommendable to use this device combined with RSM191-M, just as it has been raised. With this set-up we can pick a very good quality signal, due to the high quality that both devices offer, and set the acoustical characteristics of this sound balancing the amount of signal provided by each one of the devices.

One specific situation where this microphone would be a good choice is for picking the sound of a group of sources playing inside an ambient with an interesting or desirable acoustical conditions. For example a chorus of voices in the choir of a church, or an orchestra playing inside a concert hall. This is because it let us obtain a very defined sound from the source, mixed with the ambient of the room, and with the possibilities that the MTX191A Matrix offers, we can even balance the relative effect of the ambient in the main source.

6. Practical applications

In this chapter we pretend to display some situations where a sound pick takes place. The main purpose is explaining which would be the best equipment to use in each one of the situations and how it should be used, depending on the desired results.

The first step is differencing the two main sources that we are going to find when doing a sound pick: speech and music. As we have discussed before these two sources have distinct nature mainly in two aspects:

On the technical side, usually music is going to fill the whole frequency spectrum, since we can find instruments like bass drum with sound starting at 30Hz, or violin with fundamental tones until 3,1kHz and harmonics until 15kHz. On the contrary, human voice has a frequency range from 100Hz to 1kHz, with harmonics until 4kHz, and this is for singing, but if we focus on speech, the frequency range is even smaller.

On the subjective side, usually, in music recording the purpose is picking the highest quality signal as it be possible, caring the environment to reject any unwanted noise and for having the best acoustical conditions. Even in live sound, at gigs, the main purpose is rejecting the unwished noise, trying to pick only the sound of the source, and there are some techniques to do it. For speech captures, they can also be studio takes, like dubbing, where highest quality and clarity of the speech are sought, but they can also be street takes. The main example for this are interviews, where we want to listen the reporter but also some level of background sound. It means that, for this kind of takes, we will have to use a microphone that allow us to set a desired balance between the level of the main source and the level of the background sound.

Then, the chapter has been divided in these two main sections; music and speech. Thereby, the different techniques that are going to be introduced will be grouped according to their common characteristics.

This chapter must be taken as a starting point or reference for choosing an appropriate recording set-up for a certain situation, but in fact, the only way of obtaining the desired sound is by testing different configurations and comparing the results, which means experience.

I. MUSIC

o Recording studio

Nowadays is very usual for modest bands recording themselves in an amateur or semi-professional way, mainly because the big-budget that involves a professional production. When doing this kind of productions, currently digital technology becomes very helpful, since it allows applying a huge amount of effects and processing to the signal, but we must take into account that in order to obtain a great product it is indispensable to start from a high quality signal, and this implies a good take.

26. Recording drums inside recording studio.

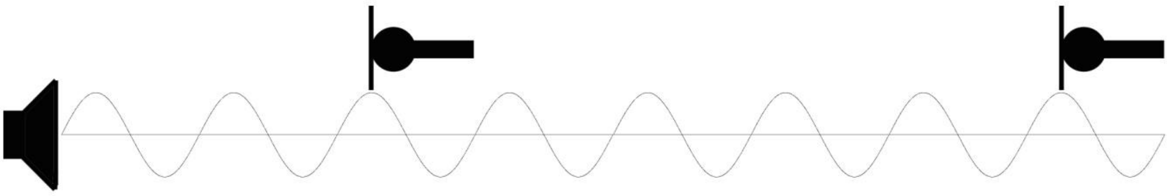
Inside a recording studio, as it is a controlled environment, signal level is not a problem, being able to adjust as best for us. So we should use the microphones that provide the highest quality, in our case the most recommended are DPA 4006 and DPA 4011 once we have heard their sounds. They are very general devices, having a great behaviour for the whole frequency spectrum, but at the market we can find more specialized microphones. For example, if we are recording a bass, we should use a microphone that focus on the low frequencies, displaying a good behaviour in this part of the frequency spectrum. For this reason, condenser microphones are the most used in studio, because they are able to provide the best quality signals, specially for high frequencies.

As we have seen from the results of the tests, each environment gives its own characteristics to the sound signal, and specially recording studios are designed to provide good acoustical characteristics to the sound. This, let us to use the room for giving the signal a special warm. The way of doing this is balancing the distance from the source to the microphone. If we want to achieve the purest sound of the instrument, we must place the microphone very close to it, but if we want to include the acoustic of the studio in the signal, we should start moving away the microphone from the source until we get the desired balance between direct sound and reverberated one.

Commonly, when recording in a studio, it is usual to use several microphones for the same source or for a group of sources. We can, for example, take profit of this for balancing the levels of the direct sound and the room sound. The procedure consists on placing an appropriate microphone close to the source, in the position where we like the sound that we got, and another microphone in a further position, picking the sound of the source from inside the diffuse ambient of the room. Then, having these two signals in two separate channels, we control the amount of reverberation from the room by mixing them until the desired level.

It must be said that doing this, we are not only controlling the amount of reverberation, but also the coloration that the room introduces in the signal, it is the equalization.

In this regard, when more than one microphone is used for the same source, an essential parameter to have in mind is the phase. The point is that when we mix the two signals, they have to joint coherently.

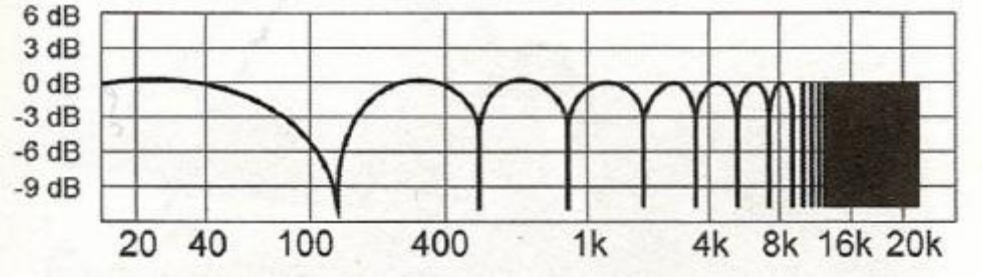


27. Emplacement of different microphones for coherent sound pickup.

In the example of the figure, the second microphone has been placed a distance equivalent to an entire number of wavelenghts from the first one. Thereby, when we mix the two channels, we must apply a delay to the first microphone corresponding to the time that sound lasts to arrive from the first to the second microphone, then we have the two signals perfectly synchronized.

Anyway, if we want to be accurate in the results, measuring the distance can be a pretty imprecise method of phasing the microphones, so at the end we must also mind our hearing perception. Is recommendable toying with the value of the delay around the calculated position, so we can perceive that while the two sources are out of phase the sound presents some strange effect, similar to a *flanger* together with a strong coloration. If we find the point where the sources are totally in phase, the sound would become more rounded and natural so we have the right value.

This out of phase effect can also be checked graphically by observing the frequency response of the combination of the two signals. In the representation we will find what is called a "comb filter", which produces the cancellation of some specific frequencies introducing a strong coloration.



28. "Comb filter" produced by out of phase signals.

To minimize the effect, there is a rule that states that the distance from the source to the microphone must by as much one third of the distance between microphones. So, we ensure that the level of the second source is lower enough for not covering the first one.

A different situation is when we have different sources playing at the same time and we pretend to record the sound from each one of them with a different microphone, for example recording a drum set. If we use several microphones for recording a drum set, each one is going to catch the sound coming from more than one source. We cannot avoid that, but we can minimize the effect or make that it happens in a coherent way, as we have seen before.

If the different sources are close one to another, the only solution is putting each microphone as closer as we can to its corresponding source. The problem is that drums are powerful sources, and at this close distance we can exceed the maximum sound pressure level supported by the microphone. For this reason it can be best using dynamic microphones for such powerful sources, because due to their low sensitivity, they tolerate higher levels of sound pressure.

Obviously, it also depends on the source, because as we have discussed before, dynamic microphones generally present limitations in their frequency response.

These could be considered technical reasons for microphone emplacement, and minding them ensures a correct signal, out of distortions due to the recording process itself. But the recording studio is the best moment to experiment with the sound.

For example, one interesting parameter that we can use when looking for a concrete sound is the directivity of the microphone. The usual tendency is pointing the microphone directly to the source since it is supposed the optimal position for the microphone, but by moving it around its edge we can take profit of the directive pattern for colouring the picked signal.

Apart from placing the microphones, we might also mind the placing of the source itself. Usually, recording rooms use to have different walls made of distinct materials and it is something to take profit of. If the source is pointing to a reflective wall, we will normally obtain a more defined or even aggressive sound, with well defined transitory sounds. On the contrary, placing the sound in front of an absorbent wall, the obtained sound is going to be more *legato*, with less defined high frequencies.

If we have in mind all these ideas, we will be able to obtain a great sound directly in the moment of the sound pickup, without any processing necessity.

o Indoor concert

With indoor concert we pretend to propose the situation of a large band, like a symphonic orchestra, playing in a concert hall. On this assumption, the usual purpose of the sound take is a recording session, since a concert hall is a place specifically designed to suitably treat the sound, then a sound reinforcement is not normally needed. This setup presents two main ways to be recorded, on one hand a multi-channel recording can be made, dividing the orchestra in some sectors, on the other hand we can use a stereo pair recording technique.

29. Orchestra playing inside a concert hall.

For a multi-channel recording, the main difference with the recording studio is the distance from the source to the microphone. In this case the different sources are groups of instruments with similar characteristics, instead of a single instrument, making that microphones have to be placed at medium distances around one or two meters from the source. In this situation we have to consider two main aspects.

The microphones should be directional, because we want that each one picks only the signal produced by the group of instruments to which is focused, in order to take profit of the multi-channel recording. But they should not be as directional as shotgun microphones, because all the instruments making up the source must be inside the region caught by the microphone. So, some good options are Newmann RSM191-M or DPA 4011, since both of them meet the requirements.

The microphones should also be condenser ones. It is because being placed at this distance of the source they are not going to be saturated, on the contrary the high sensitivity of this kind of devices is going to help to pick the sound in a better level.

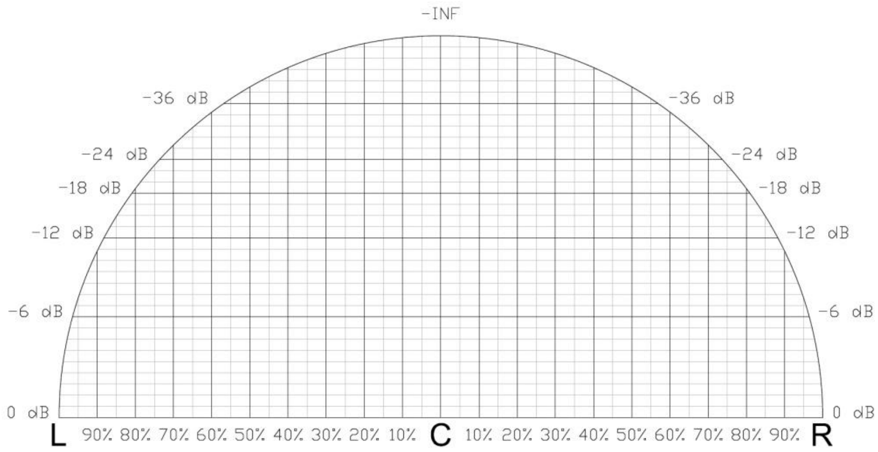
When using this multi-track recording technique, in order to obtain the spatial characteristics of the take, some post-processing needs to be added. It means that the different obtained signals have to be mixed minding the actual position of the sources on the stage. It is because as we use one microphone for each group of sources, if we mix them directly all the sources seem to be placed in the same position; centered and at the same distance to the listener. The way of obtaining this spatial effect is by adjusting the following parameters.

The panorama control (the PAN knob that we find in the vast majority of mixing desks or music production software) allows us to adjust the lateral spatiality. It works by balancing the level of signal that is sent to the left or right channel in a stereo mix and it can be adjusted from full left to full right, although these extreme positions are rarely used for setting a spatial position. If we think of the space between the monitor speakers as a pallet on which to place sources left to right, we must use our ears while toying with the PAN knob until we hear the source in the position that it is set on the stage.

As usual in acoustics, parameters are related among them and in this case the panoramic adjustment not only affects the lateral placement, but also the level of the signal. Thus, a source tends to sound with less level when its driven to one or the other side of the panorama.

The level of the signal is the parameter that defines the depth of the source on the stage. As it is obvious, the same source is going to sound louder if placed at the first line of the stage than if it is placed at the backside, just due to the distance to the listener. Then, we can use the volume control to adjust the profundity of a source on the stage.

In order to obtain a realistic effect, this volume variation must be applied together with a delay. Due to the different distances that they must travel, a signal coming from a rear source is going arrive to the listener some time later than the one coming from a first line source. And the way to create this effect in a post-processing is by applying this delay to the rear sources using a signal processor.



30. Scheme for spatial situation of the sources.

In the scheme we find some values that gives us an idea of the position that we would obtain by applying them, but as always, the best way to set the proper values is by relying on what we hear. It is important to remark that the final values that we apply are going to depend directly on the mix that we are making, because what we want is to define the relative position among the different sources inside the mix.

The second option for recording a big group of sources consists on a stereo pair recording, that is based on the use of two microphones placed away from the source, at ambient position. The main characteristic of this setup is that we are going to catch the signal with the ambient created by the room, implying a little loss in definition of the signal, but obtaining very special acoustical characteristics. This is typical in symphonic music, because main concert halls of the world have their own characteristics that experienced listeners like to appreciate, and this is the technique that allows us to catch this acoustic ambient accompanying a group of spatially distributed sources.

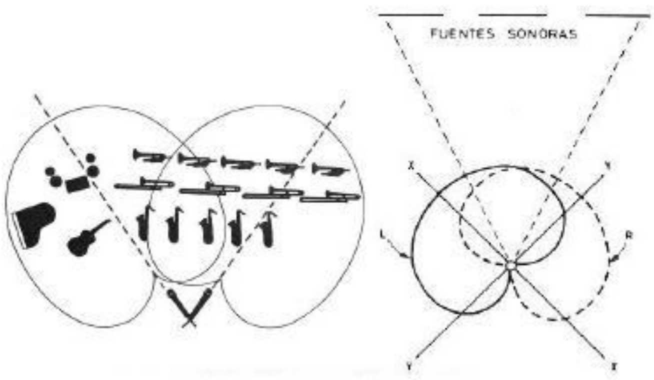
Due to the large distance to the source, for this assembling, we should use very sensitive microphones. About the directional pattern, it depends on which setup are we going to use, because there are some different setups for making a stereo pair recording. Of course, to achieve the stereo signal all these setups involve two microphones, the differences are the kind of microphones and how are they placed.

Coincident microphones.

There are two setups where the microphones are placed at the same point.

XY configuration.

In this configuration cardioid microphones are used, where each one points to one side of the stage. As we can observe in the figure 31, the directive pattern of each microphone picks up the sound coming from one of the sides of the stage. In the central position the two directional patterns overlap, so the sound produced at the centre of the stage is picked by both devices. By varying the angle between the microphones we adjust the amount of stereo effect that we obtain, but it must be done carefully, always minding the possible level loss of the central position if the angle is increased too much. The standard angle between microphones used in the XY configuration is 90º.



31. XY assembly for stereo pair recording.

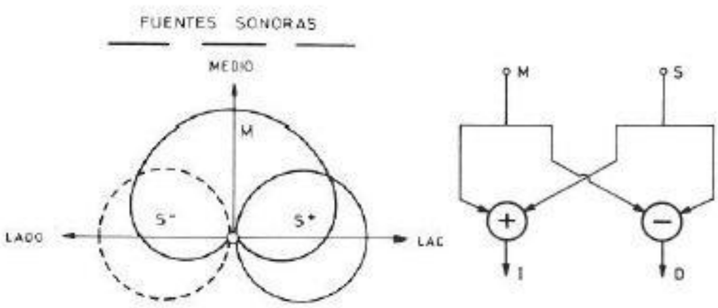
In order to maintain the stereo effect in the resultant take, the signal from the microphone pointing to the left have to be sent completely to the left channel of the stereo mix using the PAN control, while the other signal have to be sent completely to the right channel.

32. XY stereo pair.

For this configuration is really important that both microphones share a very similar characteristics, since it ensures a correct balance of levels of the two picked signals. With the finality of adjusting this balance, a good technique consists on speaking at the centre axis between the two microphones, while the two signals are mixed in opposite phase. Then, the optimum adjustment will be obtained when the resultant signal has the lowest level, because it means that we are obtaining an almost-complete phase cancelation. Of course, once the adjustment is done don't forget to switch off the phase offset button again.

MS configuration.

This configuration comprises one cardioid microphone and one bidirectional, disposed as we can see at the figure. The cardioid one is pointing to the centre of the source, while bidirectional microphone is set so that each of this lobes point to one side of the stage. As both microphones are placed exactly in the same position, the resultant polar pattern of the set creates the shape that we can observe in the figure 33.



33. MS assembly for stereo pair recording.

When using this technique we obtain two signals where the first one contains the information of picked by the cardioid microphone and the second one contains the information picked by the two sides of the bidirectional device mixed in a single signal. Then a post-processing becomes absolutely necessary in order to obtain the stereo image.

This post-processing consists on a matrixing of the two initial M and S signals, where the left resultant signal is formed by M+S and the right resultant signal is formed by M-S (the subtraction is achieved by adding the S signal in opposite phase).

The resultant directivity pattern has the shape of two directivity lobes that maintains an approximate angle of 45º with the central M axis. In the same way that we could vary the opening of the stereo capture in the XY configuration, we can achieve it in the MS configuration by balancing the levels of the M and S signals before the matrixing.

As we can note, the MS configuration is exactly the mounting that offers Newmann RSM191, but in this case all the set is contained inside the same device. But apart from this device, the normal situation consist on using two separate microphones mounted as we have explained.

34. MS stereo pair.

With this assembly, we also have the possibility of managing the amount of reverberated ambient that we have in the signal, by balancing the signals of the cardioid microphone and the bidirectional one, because is this last one who provides the vast majority of the reverberated ambient.

This was a very popular recording technique for radio, since it ensures a total monophonic playback compatibility by only disabling the S signal contribution.

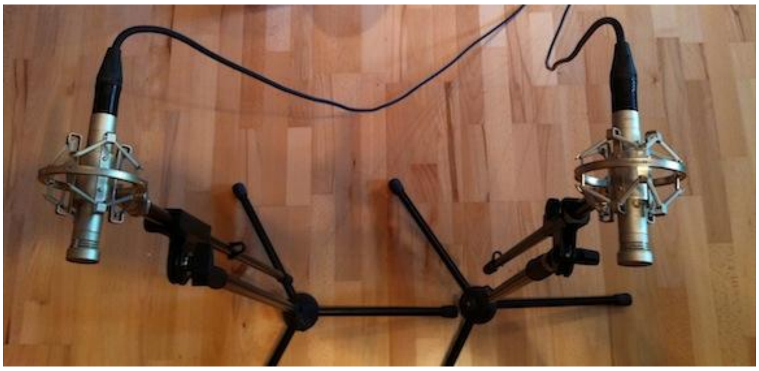
Separated microphones.

In the rest of setups, the microphones are placed leaving some distance between them, therefore they are called separated stereo pairs. Thus, the stereo effect is achieved in a similar way that we do with our ears, the same wave has to travel different distances to reach each microphone, this produces phase differences between the signals caught by each microphone and creates the stereo information.

The difference among the distinct setups of this technique is basically, the distance and the angle between the microphones.

A-B configuration.

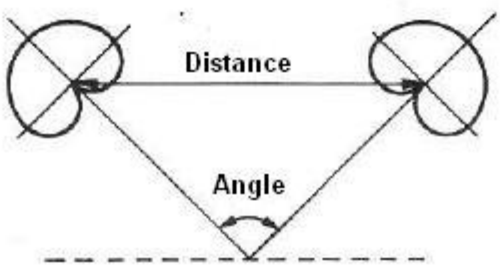
For this assembly, we can use both cardioid and omni-directional microphones. Since the operating principle of this technique is based on the phase differences of the signal caught by each one of the microphones, there is a recommended distance of at least, one fourth of the wavelength of the lowest frequency that we pretend to pick up. Setting it around 150Hz, The microphones must be separated by a large distance, between forty and sixty centimetres. Sometimes minor distances, around twenty centimetres, are used when recording near sources, otherwise the sound of a near source placed in the middle of the two microphones could be lost if we are using directive devices.



35. AB stereo pair.

This configuration is usually used for distant takes, because using omni-directional microphones it allows to catch the low frequencies without any level loss even at long distances, but in this case we must mind the balance between the direct sound and the reflected one in order to control the amount of reverberation.

Usually, in the A-B configuration the angle between microphones is 0º, meaning that they are placed completely parallel separated by the mentioned distances. But there are some other configurations that come directly from this one, where the main differences are the distance and the angle between microphones.



36. Assembly for separated stereo pairs.

DIN configuration.

Consists of two cardioid microphones, separated twenty centimetres and an angle of ninety degrees. As it uses cardioid devices, we are going to have a loss in the low frequencies when used at far distances, so this configuration is more recommendable for being used at short distances, for example for the recording of a piano or small ensembles.

37. DIN stereo pair.

NOS configuration.

This setup also uses cardioid microphones, separated a distance of thirty centimetres and an angle of ninety degrees. As we can observe, this configuration is very similar to the DIN, offering a slightly more defined stereo image due to the larger distance between microphones.

ORTF configuration.

In this setup we must use cardioid microphones, separated a distance of seventeen centimetres and an angle of hundred and ten degrees. This configuration is the most similar to the human ear observing its disposition. It provides a more defined stereo image than the XY configuration, but the loss of level of the central sources can be more appreciable due to the distance between microphones and the higher angle between them.

The most remarkable difference between the A-B and the rest of separated microphones configurations is that, as the A-B can use omni-directional microphones and they are usually placed at long distances to the source, it is the configuration that most is going to pick the acoustical contribution of the room.

As each one of these setups provides a slightly different stereo signal, the best option is making some tests before the recording, trying with the distinct assemblies. Which one is going to produce a better sound depends, as we have seen, on the acoustics of the room and the characteristics of the source. Although, it must be said that, as the human sense of location is more sensitive to the time differences between the two signals than to the level differences, the separated microphones techniques, in general, provide a best stereo sensation than the coincident microphones ones.

o Exterior gig

Here, we propose the typical situation of a small band, from three to fifteen components the biggest ones, playing on a noisy environment. This noise can be caused by the environment itself in a exterior gig or for example by the public inside a local. Then, while the usual purpose of a sound take inside a concert hall was a recording session, in this chapter we pretend to discuss a sound reinforcement situation.

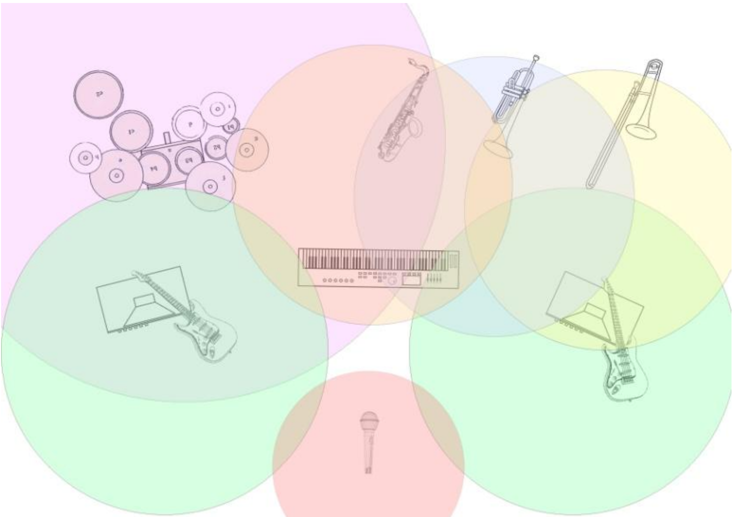
38. Exterior gig with sound reinforcement system.

The sound reinforcement is normally needed for two main reasons. First, because the place where the source is playing lacks the appropriate acoustical conditions. We can think for example in an outside gig, where the lack of reflections produces a quick loss of the sound level, which is not able to reach all the audience surface. The second main reason is when we have a group of sources with very different characteristics, which is a typical situation for example in world music. In this style acoustic folk instruments, which uses to produce a soft sound, are usually combined with percussion instruments or some extremely noisy ones like electric guitars and drum set. Then the sound reinforcement is needed in order to match the different level of all these sources.

Whichever the problem, when we introduce a sound reinforcement system, two different separated spaces are created. On one hand we have to care the sound on the stage, on the other hand, the sound that goes out from the stage to the public zone must be adjusted too.

Typically, in nonprofessional shows the sound upon the stage uses to be underestimated, focusing the attention on the sound that is emitted to the public, but where comes the sound that the public listens in a live show from? Yes, it comes from the stage, so we are never going to be able to achieve a good sound for the public if we don't care the sound upon the stage. So the first step in order to achieve a good sound for a live show is getting a good sound on the stage.

The basic problem on the stage is that we have quite different sources usually playing at positions that have been decided to create a cool or visual show, but most of the times it is not appropriate for creating a good sound space. Then, musicians need a monitoring system to be able to listen what each other is playing. The ideal goal would be to create a diffuse ambient on the stage where every musician could hear the same sound, but it becomes impossible since the different sound sources are distributed on the scenery, creating different sound pressure zones.



39. Contribution of the different sources on the stage.

As we observe in the figure 39, the distribution of the sources on the scenery causes an irregular distribution of the sound. There are even some special cases, like the keyboard, where these sources are not producing any effect on the stage, since they are not acoustic ones, but the player is equally affected by the irregular sound.

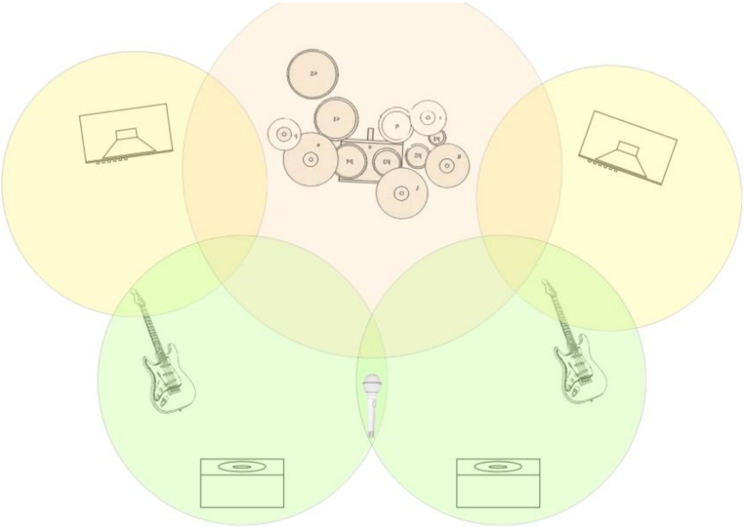
This irregular distribution of the sound makes that one musician can hear the nearest sources, but he is unable to listen the furthest ones, due to the level difference.

Then the solution consists on using an independent monitor for each musician or for each zone of the stage. With this, we can create an independent mix for each monitor, balancing the lacks of sound for each zone.

But even using this technique, the main factor for achieving a well mixed sound on the stage is minding the relative position among the sources. For example, if we place an acoustic guitar just next to a drum set, we can make the guitarist to hears to himself by using a monitor next to him, but we would be using an unnecessary amount of energy which moreover will help messing up the total sound of the scenery. A better option consists on move away the acoustic guitar from the drum set so the level of the drum set is lower at the position of the guitarist and this one needs les intensity from his monitor.

This practice becomes even more relevant for small stages, where the sound level uses to be louder than it should be, specially inside enclosed rooms. Then, in this situation is really important to control the level of every source on the stage, avoiding any unnecessary waste of energy.

For example, if we have the typical formation of four members playing inside a pub, an appropriate set up that allows us keeping controlled the level of sound on the stage is using only two front monitors. Supposing a small stage, it should be enough to cover all the front line. The only trouble would be the drummer. He is sitting behind a bulky instrument that produces a sound level around 90dB, so it is impossible for him to hear the signal of the front monitors. The possible solutions are adding a third monitor near the drummer or making him use headphones. A third monitor increases the total sound pressure level, moreover it features another source distributed on the stage, messing up the sound of the place. Then, the best option in this case is using headphones for drummer's monitoring.



40. Small stage with two way monitoring set-up.

As we can observe in the figure 40, we have created a regular zone in at the front line of the stage where the musicians can move around without variations of the heard sound. The only trouble in this zone could be the contribution of the direct sound of the drum set, but we only have to adjust the level of the monitors until it matches with the drum set one.

The back part of the stage is divided in three zones. At the middle it is the sound of the drum set, with the drummer using headphones monitoring. At each side it is the sound of each guitar, where each guitarist can check his sound out of the monitoring mix.

If we have more elements on the stage, as in the figure 39, the procedure can be the same. The point consist on grouping the sources that shares similar characteristics (specially the emitted sound pressure level) in separated zones and creating well defined mixes for each of this zones. There also exist some devices which can help us separating the different sources in complicated situations like acoustic screens.

41. Acoustic screen for drums.

They are not absorbent materials, only reflective walls that prevents the sound to go to the front of the stage. Then, they can be freely used outdoors, because the reflected sound will be sent to the backstage being attenuated by the background curtain or trough the open air, but when they are used indoors we must be careful. In this case we will normally have a background wall that, together with the acoustic screen, can create a high pressure zone really annoying for the drummer.

The other main part of the on-stage equipment that we have not already introduced are microphones. Of course they are the main equipment which starts the sound chain in a sound reinforcement system, but it has been considered to explain the topic in this order because now it is immediate to understand how important becomes their emplacement on the stage.

Observing the figure 40, if we focus on the singer microphone, depending on the position of the singer on the stage the microphone is going to pick up the voice accompanied by the background sound of that zone. Once more we can appreciate the importance of creating constant sound zones where the singer can move around without any change in the background sound, thus the background sound can be controlled.

Obviously, in order to optimize the signal to noise ratio (the source sound level respect to the zone background level) the microphone should be positioned as close to the source as possible, being the reason why dynamic microphones become so recommended for this kind of applications. They support much better loud sound levels and reject the low level background noise. On the market we can find some dynamic models specifically recommended diverse applications and offering higher quality than the ones that we have tested, but the models tested in this project have served to show the general behaviour of this kind of devices.

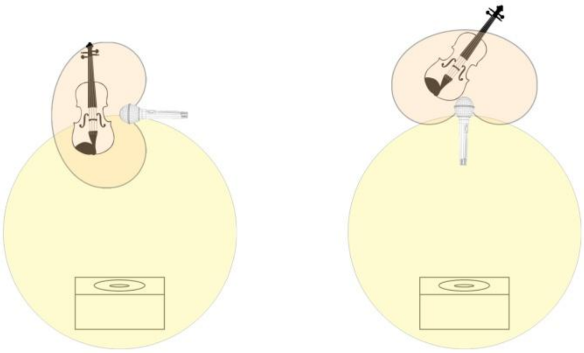
Now a days, there are also a lot of models of condenser microphone that can be used on a stage (understanding a noisy stage) due to the improvement in the capacity of supporting higher sound pressure levels, but in fact, dynamic microphones are still the standard device used in a exterior gig. This is mainly because as we have stated before, the lowest sensitivity of the dynamic devices, together with the worst capacity of picking the lowest frequencies help to reject the background noise.

Having stated the best way to create an appropriate environment for obtaining a clear and controlled sound on a stage, we are still not out of all trouble. The main problem that we will find on a stage is the feedback.

It appears when the sound emitted by a monitor is captured again by a near microphone, creating an audio loop. When it happens, the gain of a certain frequency starts to be increased without end, producing a high-pitched squealing noise familiar to those who have listened to bands at house parties, where the sound setup is less than ideal. The noise is produced around a certain frequency that depends on several parameters like the resonance frequencies of the microphone, the amplifier and the speaker or the distance between the monitor and the microphone.

The feedback is an important parameter that actually determines the placement of the microphones. It can be attenuated or eliminated by cutting out the affected frequencies by means of an equalizer, but with this procedure we are introducing some coloration to the signal, so a better option consists on setting the microphone in the right position. Minding the distance from the monitor to the microphone and the directivity of the microphone, the sound emitted by the monitor can be left out of the microphone pickup range, so we are avoiding the loop that produces the feedback.

This is an important reason for using directive microphones, which as we can see in the figure 42 help to keep the monitor signal out of the pickup pattern of the microphones.



42. Use of the directivity pattern to prevent feedback.

Whit all these premises we are able to achieve a defined and controlled sound upon the scenery which makes the musicians play comfortably. Moreover it lets us picking up the sound in a way that guarantees a well defined multi-microphone take.

All the signals obtained from this multi-microphone take will be mixed, processed, amplified and emitted to the public zone. All fours steps require an accurate management which is not within the objectives discussed in this project, but a lot of related information can be found in the purposed bibliography.

The only point to remark is the importance in the collocation of the loudspeaker system. The main goal of a PA system (part of the sound reinforcement system dedicated to emit the sound to the audience) is obtaining a well mixed sound that spreads out with constant conditions throughout the whole audience.

With this purpose the sound signal is suitably processed and emitted from a convenient position. The processing helps in obtaining the wanted characteristics for the emitted signal, but in order to uniformly spread it out throughout the audience the emplacement of the loudspeakers assembly becomes the essential part. Just by caring it we can obtain the wanted results, while avoiding possible troubles that the environment could create. And this is really important since there are some effects that cannot be fixed by processing the signal or they represent a wasting of energy that can be avoided just by properly setting the loudspeakers.

II. SPEECH

o Studio

The typical situations where speech is recorded inside a studio are dubbing sessions or radio producing. In both cases, the aim consists on achieving the maximum intelligibility of the speech as possible, but they have different particularities, since radio is usually carried out live, while a dubbing session is a more slow and perfectly cared procedure. Due to these, the used equipment and the different techniques to apply are slightly different in each one.

For radio production, due to the live nature, the equipment must help achieving a regular take. The reporter is supposed to know the skills of broadcasting, but the quality and specially the level of the signal must be constant all time, regardless the ability of the speaker. It is so important in radio due to the nature of the listener. Radio is a mass media, which great part of the public uses to hear as an accompaniment while doing other tasks, then the listening environment uses to have an important and changing background noise.

For this reason dynamic microphones are highly used for radio production. Their characteristic frequency response is appropriate for picking the vocal frequencies while helps keeping out another undesired sounds. Therefore, it improves the intelligibility of the voice, which is the only important sound that we want to capture.

In order to improve the intelligibility is also important keeping a low reverberation time, and this is achieved in two ways. On one hand, radio studios uses to maintain a low reverberation time between 0,2 and 0,6 seconds. On the other hand, the lowest sensitivity of dynamic microphones helps rejecting the furthest or softest sounds, like the diffuse ambient of the room, being very useful in not so well conditioned rooms.

43. Radio production studio.

Directivity of microphones used for radio production depends on the concrete application, but cardioid microphones are normally used. This polar pattern helps rejecting the near sources when we have several speakers, because we pretend to pick up each of them through a different channel. Moreover, the cardioid pattern becomes useful for the artistic side of radio production.

An important effect used in radio broadcasting are the planes of sound. The normal one is called first plane, which recreates interpersonal communication, but there are also some others, being the second plane, the third, and the one called close-up. Second and third planes are a decrease of sound level generally used to recreate the effect of distance. Sound planes can be created both by technical procedures or simply by varying the distance from the speaker to the microphones. Using this last method, a dynamic microphone allows to create the effect requiring a smaller variation of the distance between the microphone and the speaker due to its lower sensitivity.

On the other hand, close-up plane is used for recreating intimate communication. This is made by speaking very close to the microphone, which for dynamic microphones means a high presence of the proximity effect. It helps to get the feeling of a close communication and can be particularly heard in night radio shows, which use to have a near atmosphere.

Common complements for microphones in studio are pop filters, placed between the microphone and the speaker, they serve to attenuate air strikes produced when pronouncing the consonants p, b and t at short distance.

Voice dubbing has, as we have said, some particularities comparing with radio production. When creating voices for a movie, expressivity is the most important point then, the take should not maintain such a monotone stile as in radio. It makes the sensitivity of condenser microphones more appropriated to be used in this purpose.

Moreover, unlike in radio production the whole frequency spectrum becomes important in dubbing. On one hand, the audio track of a film has not got the bandwidth limitations of a radio channel, which lets use the whole spectrum. On the other hand, not only the voice is important, since the sounds produced by the movements of the voice actor improves the realism of the take when it is mixed with the video. Then, condenser microphones are normally used in voice dubbing.

These are some proposals for using the equipment based on the explained reasons, but any new idea deserves to be experimented, since it is the only of finding new results. Maybe radio uses to be more strictly joined to the explained configuration due to its specific technical requirements, but dubbing is an open world for trying out the equipment.

In this regard, every effect or ambience that we can obtain in the recording process is going to improve the realism of the audio in the film. Specially the environments are a critical part which must be cared in the moment of the recording, non relying on the postproduction process to recreate them.

One of the most challenging situations can be for example dubbing the voices of an outdoors scene inside a recording studio. Exterior environments have almost zero reflections, which means an extremely low reverberation time, then we have to respect this feature in the moment of the dubbing in order to obtain a realistic take.

The options are to have a well prepared studio, which can be configured for an extremely low reverberation time or recording these takes outside, at a quiet place. With this last option we can also obtain some effects like a soft wind, adding realism to the scene. Obviously, in this suppose all cautions must be minded in order to maintain a high level of intelligibility on the voices, as using windshields or caring the distances to the speaker.

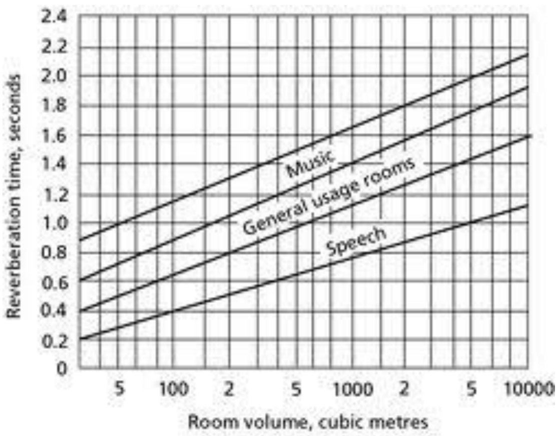
o Conferences

If we need to use a microphone in a conference, it is because there exists some kind of sound reinforcement system. This system must meet certain requirements in order to be appropriated for the voice reinforcement in each particular room, but first of it the room must have an adequate acoustical behaviour. As we have stated before, in order to achieve a good result, we must start from a quality signal, and it is also applied even before the microphone take.

The main parameter which affects the intelligibility of the speech is the reverberation of the environment. Then, it must be taken into account from the design of the room depending on the concrete application to which the place is going to be dedicated. Nowadays, it is quite usual that multi-purpose rooms are used for conferences, for example in hotels or universities, where live music or other performances like theatre use to be also performed in the same room. For doing it, the room must be a real multi-purpose space which can vary the acoustical conditions to adapt them for the specific performance.

This room features must be minded since the beginning of the construction project, because it is easier building a room with predefined characteristics than treating to obtain them in an already built location. Moreover, the shape of the room itself is one of the most important acoustic features to care.

For speech situations, a low reverberation time is required. It depends on the total volume of the room, but as we can observe it is always lower than for music applications.



44. Reverberation time depending on the volume of the room.

And this is the particular point, for a given volume, a multi-purpose room must be able to change its average reverberation time to adequate it to the specific performance.

The usual way for achieving this feature is by varying the total surface of absorbent material inside the room. There exist some different methods for doing it, but the vast majority relies on double sided mobile surfaces, where one side is made of reflective material and the other one is covered with a given thickness of absorbent material. Depending on the side exposed to the sound it produces a different effect on the sound waves.

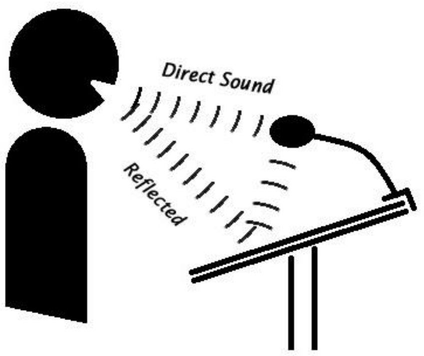
45. Three sided variable acoustic surfaces.

As it is not usual to find this kind of acoustic installations in the vast majority of the rooms, the reverberation time in the diffuse ambient uses to be higher than the desirable. Thus, multipurpose rooms are usually designed to maintain an average reverberation time, which as we can observe in the figure 44, is set treating to make it suitable both for music and speech.

Then, the microphone must be set close to the speaker treating to pick the direct sound alone. In this regard, it is normal to install the microphones above the used lectern, just in front of the speaker. This method allows to maintain the microphone under the critical radius, but it is still affected by another troubles.

Inside a room, reverberation is produced by the addition of all the late reflections. Earlier ones reach the microphone within a very short period of time after the direct sound, directly affecting the primal sound, since they are understood by the hearing as part of the direct sound. This is the effect which gives the voice its special sound features for each particular room.

If the microphone is set above the lectern, the very first reflection is produced on the lectern itself. The sound emitted by the speaker bounces on the surface of the lectern and it is picked up by the microphone within a short time after the direct signal.



46. First reflection on a lectern.

A solution consists on using a special kind of microphone which has been specifically designed for this application. Surface-mounted microphones have a special shape for being mounted on a surface, which allows them reject the signals coming from the bottom hemisphere.

The microphone Audio Technica 871R, tested in this project, belongs to this kind of devices. As we stated in its particular assessment, it shows a peculiar behaviour which makes it adequate for the specific purpose that we are discussing. Apart from setting it upon a lectern, it can be also attached to a wall, ground or any other convenient surface. In these all these situations, its special shape allows it to avoid the first reflections which uses to introduce undesired effects when a regular microphone is set close to a surface.

Another special feature in conferences is the nature of the speaker, who is usually standing, talking to the audience, not to the microphone, so we should use a means capable of following his movements without interfering in the performance.

The most appropriate microphone for this use is a Lavalier one, since it is attached to the body of the speaker and usually designed for voice picking. It is also a good option because the speaker does not have to pay any special attention to it and it is not very visible, non distracting the audience from the conversation.

47. Lavalier microphone and transmitter.

The standard kit comprises the small microphone, the transmitter and a receiver, since this kind of microphones use to be wireless. In this regard, this kind of devices needs some preparation before using them. On one hand, both transmitter and receiver offer some parameters to modify, which must be properly adjusted in order to obtain a clear signal free of interferences. On the other hand, they are battery fed devices, so batteries must be checked or even replaced just before every use of the kit.

The microphone must be placed close to the mouth of the speaker. For example it uses to be attached to a tie or the neck of the shirt. Then, the thin cable which links it to the transmitter can be hidden inside the shirt, since a usual place for the transmitting device is attached to the belt at the backside of the speaker. The receiver will be placed next to the technical control site and it is highly recommendable to avoid any object in the direct way between transmitter and receiver, which could cut off the transmission.

The main problem that we can find with this kind of device is that when the speaker moves the microphone rubs against the body or clothes, producing a very annoying low frequency noise. For this, it is very important to properly place the capsule of the microphone avoiding it to touch the clothes.

About technical specifications of a Lavalier microphones, they use to be omnidirectional. It helps to maintain a constant signal level regardless the position of the head of the speaker, while keeping a good balance between the voice and the background sounds due to the closeness to the mouth of the speaker.

o Outdoor speech takes

The main issues to keep in mind in this situation are background noise and mechanical noise, produced by sharp movements and hits. In fact, if we think in TV reporters, the majority of interviews are made at worst desirable environments to record audio.

From this point of view, we need a robust device able to support adverse conditions, so the best option is using dynamic microphones, since condenser ones are much more delicate. The main features that dynamic microphones provide for this environment are ruggedness, for supporting hits and ambient conditions, and lower sensitivity, for supporting higher sound pressure level and better rejecting background noise.

Thereby, dynamic microphones are highly used for outdoors interviews. We have hardly discussed how their lowest sensitivity and their narrowest bandwidth help rejecting the background noise.

The appropriate directivity depends on the concrete situation, but the reporter should know how to take profit of the specific microphone that he is using.

When the speech is the only important thing to be picked up, the best option is using directive microphones, since their lower pickup angle keeps the undesired sound away.

48. Outdoor interview in a noisy environment.

For example, a dynamic device becomes really helpful in the typical situation where there are a lot of medias fighting for the words of the interviewee. In this case, there are a lot of sound sources emitting from a short distance of the microphone, so the reporter must point the microphone directly to the source in order to obtain the best sound.

Although, when the main sound must be picked up within a background of ambient sound there are two main options.

On one hand, a single microphone can be used to pick both the main and the background sound. In this case an omnidirectional microphone represents the best option, since it picks up the sound of the ambient at the same level for any direction. The way of balancing the levels of the main source and the background is by varying the distance between the main source and the microphone. The closer we set it, the higher becomes the level difference, so we can decide the amount of background sound that we want to include in the take.

For this, a high quality monitoring system becomes essential. We obtain just one signal where the main sound and the background one are already mixed, so it is impossible to adjust the balance later. Then, the right level mix must be obtained at the moment of the take.

It is also important to mind the frequency response of the device when using this technique. We are going to record a background noise which is composed of a huge amount of sounds from different natures. It can even contain music, so the frequency response of the source is not as narrow as when we only record human voice. Then, the microphone must ensure a complete frequency response in order to obtain a quality background.

This technique is really useful since we only need one microphone, meaning that only one person is required for recording the take. But it could not be used for example in the situation of the figure 48, since the sources that produces the background noise are too close to the microphone.

Then, for noisy environments, if we need to pick up the ambient sound, a better option consists on using two separate microphones for the main source and for the ambient. This technique is a combination of the above two, where a directive microphone is used close to the main source just to pick up the speech and an omnidirectional one is placed away in order to capture the ambient sound.

When mixing the two signals they must be perfectly synchronized, since part of the ambient sound is always going to be captured by the directive device too.

Whichever the case, for outdoor recordings is always essential using a windscreen, since they largely avoid the annoying noise of the wind that we have heard in some of the recordings of open air. It is also really important to move the microphone carefully, avoiding rude movements which produce mechanical noise.

o Video takes

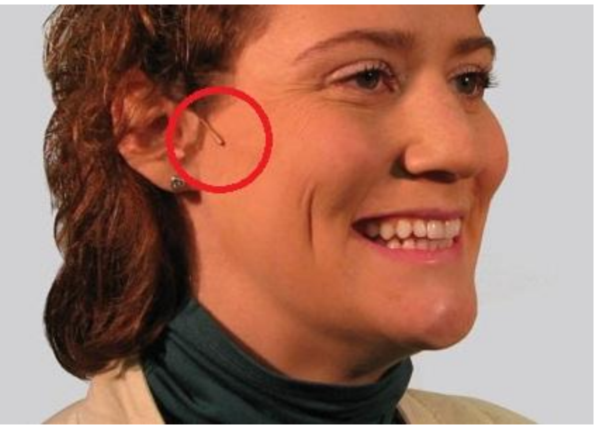
When we are picking up the sound for a video recording there are several situations with different characteristics. A main feature which divides these situations in two groups is the possibility for the microphone to appear in the image or not.

The typical situation where the microphone is present in the picture is in a report. Reporters generally use a handheld microphone which perfectly shows the logo of the TV channel.

In this suppose the microphone is normally used to pick up the voice of the interviewee alone, while the background noise is captured by an omnidirectional device placed on the camera. As we stated in the previous chapter, the microphone used for catching the speech uses to be cardioid, so it is very important for the reporter to point it directly to the source which is emitting in each moment. But these movements must be careful and the microphone must be protected with the appropriated windshield in order to avoid mechanical and wind noises.

In other situations the microphone can be present in the screen, but it does not have to be so evident. Then, the best option consists on using a Lavalier whose use has already been explained on a previous chapter. The only thing to add is that when the speaker holds a static position, like on TV news, rubbing against clothes becomes a less important problem, so the microphone can be set in the way it gets the best sound.

Lately, there are TV shows where the microphone uses to be hidden on the body of the presenter. On this purpose, there is a special type of Lavalier microphone whose capsule is even smaller than the regular ones. They are camouflaged in the skin using some make-up, which apart from making them almost invisible, avoids any mechanical noise due to hits against clothes.



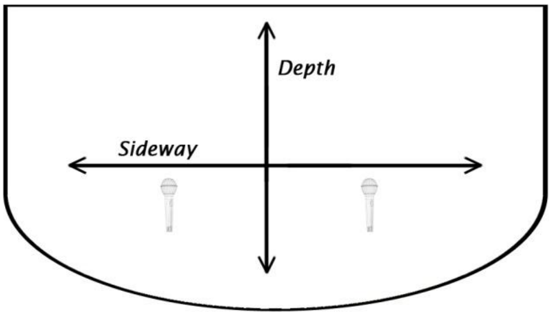
49. Camouflaged Lavalier microphone.

In spite of their small size and their emplacement, normally they are not completely invisible, so in the situations where the microphone cannot be shown on the screen another techniques need to be used.

The first procedure for avoiding the presence of the microphone started to be used as soon as sound was introduced in cinema. It consists on hiding the device among the stage objects. On the early films it caused an important injury to the drama since the poor features of those microphones forced the actors to speak directly in its direction, hardly limiting their movements.

Nowadays, due to the improvement on the features of the microphones, specially the increase of the sensitivity and the decrease of the internal noise, this technique can be used with better results. Obviously, this procedure is appropriated for quasi-static scenes, since the movement of the actors varies the distance to the microphones, producing sound level changes.

This effect, unwanted at the beginning, can be used in a profitable way though. Two microphones can be placed in defined places of the scenery obtaining a stereo take. It basically consists on using the AB configuration of the stereo techniques introduced in the above chapters.



50. Emplacement of microphones on the scenery for stereo pick-up.

The main point to consider is placing the microphones in a positions which produce a constant level regardless the sideway movement of the actors. For this, the addition of the two signals when the actor is at the middle point between the two microphones must have the same level that the signal of a single device when the actor is close to it. On the contrary, when the actor moves forwards and backwards on the stage, the level drop must be present, reinforcing depth sensation.

The main advantages of this technique are that the microphone is set on a static position, avoiding any mechanical noise and it can be close to the main sources even in long shots.

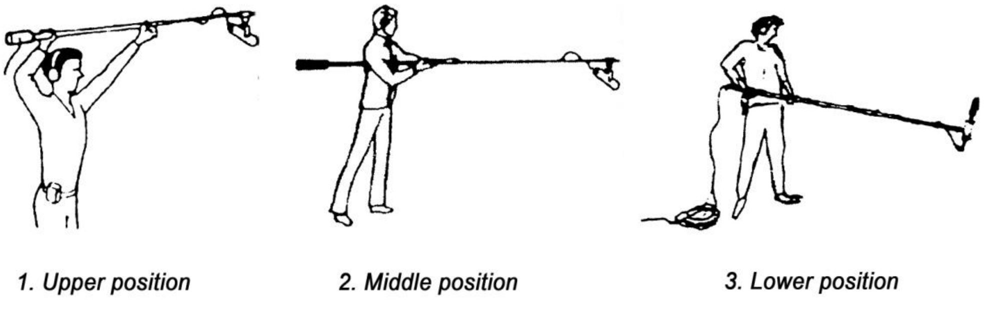
But there are situations where a microphone cannot be hidden anywhere on the stage, or there is too much movement on the scene for just using a static device. Then the only option is taking the scene from out of the shot, using a shotgun microphone.

As we have learnt from the results of the tests, the two main features of shotgun microphones are the high sensitivity and directivity, making them able to pick up further sounds while rejecting the majority of ambient noise. But, these specific features require a precise control of the microphone in order to obtain a quality sound.

In this suppose, the microphone uses to be attached to a boom. This allows the operator to control it close to the sources, while keeping it out of the camera shot.

51. Shotgun microphone installed on a boom.

In this regard, there are three main positions for the boom to be held depending on the kind of shot. The most common one is the upper position, where the operator holds the boom almost parallel to the ground, maintaining a "H" figure with his body. It ensures that the boom won't appear on the screen diagonally crossed, while procuring the maximum height for the boom.



52. Boom holding positions.

The middle and lower positions are not so used, but they can be useful depending on specific space restrictions of concrete shots. For instance, the middle one can be used when picking up a short and close source, like a series of steps or a ball rolling along the floor.

After placing the boom, it must be controlled following the movements of the source according to the perspective of the camera. If the camera is fixed on a position, the boom operator will usually be also fixed, being able to make two kinds of movements.

If the movement of the source is short or there are several sources very close to each other, the most suitable movement is a single rotation just over the axis of the boom. It is a slight and clean movement which does not create any noise, getting a soft variation of the sound. If the operator needs to cover a largest displacement, he can combine the previous movement with a rotation over the axis of his body. In this movement, his feet are fixed on the floor while the operator just rotates his waist, avoiding mechanical noises on the boom.

In this suppose, the movement over the axis of the body must be done before pointing the microphone to the next source, otherwise the microphone is moved directly against wind, producing an annoying noise.

The last movement which can be done using a boom is a travelling that follows the displacement of a source. It makes sense when the camera also follows the source, since the relative position of the actor on the screen remains constant and so has to do the sound level. It is a very delicate movement where the operator must walk carefully, avoiding any rough shock on the boom while he maintains a constant distance between the microphone and the source and keeps the microphone out of the camera shot.

The same as for outdoor interviews, shotgun microphones must always be protected by a windscreen, since working in open air we can never control the ambient conditions. Moreover, handling a microphone with a boom can create momentary wind flows which must be prevented.

7. Conclusions

One of the main goals of this project was to introduce the reader in the field of the sound. It has been done by presenting several acoustic parameters which explain the effects that we have gone finding in the diverse situations of the project.

Through these parameters we have learnt that sound is a physical concept which therefore is totally dependent on the interactions with the other physical elements of the environment. Thus, we have seen how maintaining the same source signal and recording set-up we can pick up a totally different sound by varying the characteristics of the environment.

This statement totally rejects the idea that we are going to obtain a best sound by using a better equipment. Of course, it is obvious that we must use professional equipment for professional applications, which ensures a minimum quality level that can be estimated from the technical characteristics of each device. But the result always depends on the way that we use that equipment in the specific environment where we are working.

Then, some ideas like this microphone is made for this application are not valid since the right approach is knowing the features of each specific device in order to decide which one is appropriated for each purpose. Despite we have seen it studying the properties and applications of microphones, it can be also applied to any kind of audio device.

Moreover, this approach is closer to real life situations, where we use to have a certain set of equipment, generally limited by the budget, and we have to decide the best way to handle it in order to obtain the best result. We can never say "I do not have any microphone for voice recording", but " this is the available equipment. Which device can I use in order to get the best out of the singer?".

We see that sound is a subjective field, where anything is wrong or right but depending on the concrete result that we pretend to obtain. Then, sound can be divided in a technical side, which is responsible of using the equipment within its capacities, and an artistic side, which profits the features of each device to obtain the wanted result.

This is the main reason that it was considered important to accompany the technical explanations of the different parameters with a real reference of the effects that they introduce in the sound. The results is the CD that accompanies this project, where the reader can listen the consequences of the different recording conditions in the sound for each take.

The main ability which allows the sound technician to effectively control a technical parameter to obtain a concrete artistic result is experience. Then, the attached CD becomes an important tool to get some of this experience in recognizing how the different acoustic parameters affects the sound.

In this regard, in *Practical Applications* chapter some premises and procedures on how to properly use the equipment have been presented to the reader. They are all based in the experience gained during years of studying sound properties, but they are just premises. It means that they must be taken just as a starting point which ensures that we are properly handling some of the typical issues of each specific situation. But the final procedure to use will always be the one which lets us to obtain the best result.

Then, we are in front of a totally open field, where experiencing is the main way to learn how to best handle the sound.

8. Annex

I. TECHNICAL CHARACTERISTICS OF ASSESSED RECORDERS

Manufacturer: HHB		**Model:** MDP500		
Recording format	**Recording/ playback time**	**Sampling frequency**	**Frequency response**	**Signal to noise ratio (playback)**
MiniDisc DAS	60min-120(mono)	32 / 44,1 / 48kHz	10 Hz - 20kHz	>89dB
Dinamic range (line input)	**THD 1kHz ref 0dBFS**	**Coding**	**Modulation**	**Channels**
>96dB	<0,02%	ATRAC 4.5	EFM	2

53. Technical specifications of HHB MDP500 recorder.

Manufacturer: Aaton		**Model:** Cantar X		
Recording format	**Recording/ playback time/ capacity**	**Sampling frequency**	**Frequency response**	**Signal to noise ratio (playback)**
Hard Drive/Flash Card	127GB HHD	44,1 - 96,096kHz	10 Hz - 40kHz	
Dinamic range	**THD 1kHz ref 0dBFS**	**Coding**	**Modulation**	**Channels**
		FAT32		5mic, 4line, 8AES

54. Technical specifications of Aaton Cantar X recorder.

Manufacturer: Maycom		**Model:** Handheld II		
Recording format	**Recording/ playback time/ capacity**	**Sampling frequency**	**Frequency response**	**Signal to noise ratio (playback)**
Compact Flash	>137GB	16 to 48kHz	20 Hz - 20kHz	>80dB
Dinamic range	**THD 1kHz ref 0dBFS**	**Coding**	**Modulation**	**Channels**
80 dB	<0,005%	BWF	PCM	2mic/line

55. Technical specifications of Maycom HandHeld II recorder.

Manufacturer: Sound Devices		**Model:** 702T		
Recording format	**Recording/ playback time/ capacity**	**Sampling frequency**	**Frequency response**	**Signal to noise ratio (playback)**
Hard Drive/Cflash I,II	40GB HardDrive	32 - 192kHz	10 Hz - 40kHz	>80dB
Dinamic range	**THD 1kHz ref 0dBFS**	**Coding**	**Modulation**	**Channels**
110 dB	<0,004%	BWF/wav/mp3	PCM	2mic/line, AES

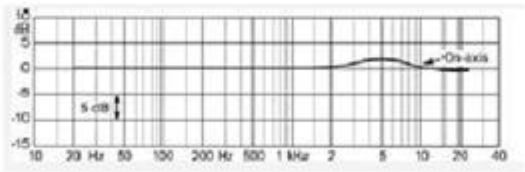
56. Technical specifications of Sound Devices 702T recorder.

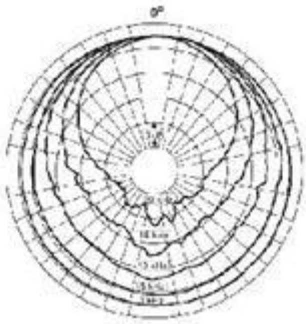
Manufacturer: Sound Devices		**Model:** 744T		
Recording format	**Recording/ playback time/ capacity**	**Sampling frequency**	**Frequency response**	**Signal to noise ratio (playback)**
Hard Drive/Cflash I,II	40GB HardDrive	32 - 192kHz	10 Hz - 40kHz	>80dB
Dinamic range	**THD 1kHz ref 0dBFS**	**Coding**	**Modulation**	**Channels**
110 dB	<0,004%	BWF/wav/mp3	PCM	2mic, 4line, AES

57. Technical specifications of Sound Devices 744T recorder.

II. TECHNICAL CHARACTERISTICS OF TESTED MICROPHONES

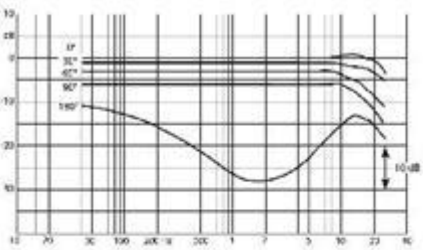
Manufacturer: DPA		**Model:** 4006 Studio		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ (open circuit)**	**Max. sound level**
Polarized condenser	Omnidirectional	<200 ohms	35mV/Pa	143 dB SPL

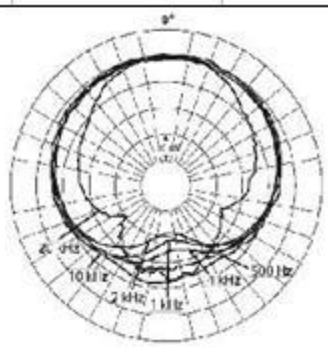




58. Technical specifications of Sound Devices 702T recorder.

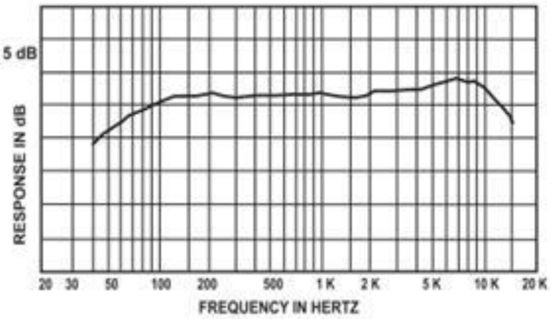
Manufacturer: DPA		**Model:** 4011 Studio		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ**	**Max. sound level**
Polarized condenser	Cardioid	<200ohms	10mV/Pa	158dB





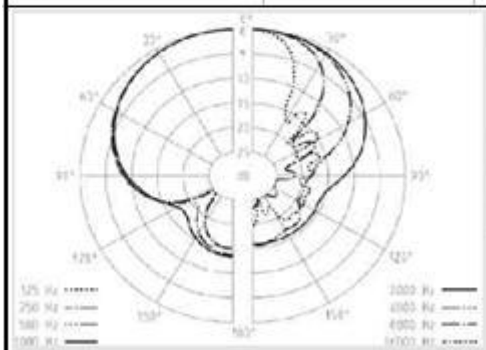
59. Technical specifications of Sound Devices 702T recorder.

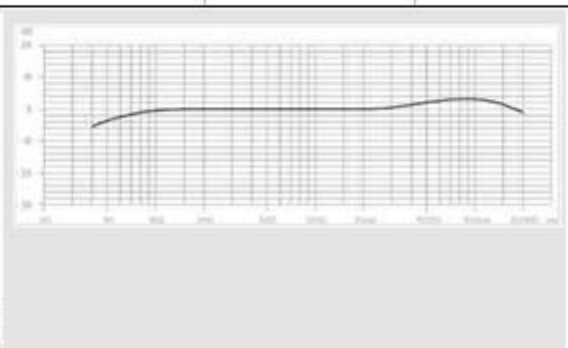
Manufacturer: Electro Voice		**Model:** RE50		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ (open circuit)**	
Dynamic	Omnidirectional	150 ohms	1,8mV/Pa	



60. Technical specifications of Sound Devices 702T recorder.

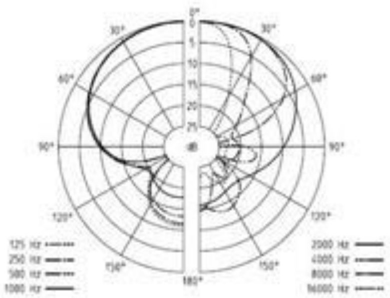
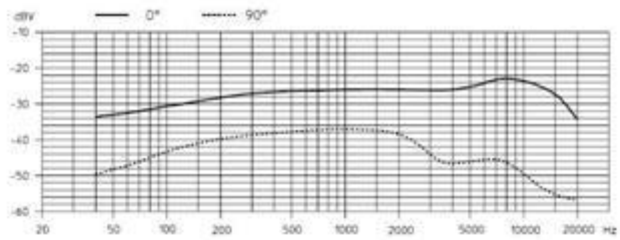
Manufacturer: Sennheiser		**Model:** MKH-416		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ**	**Max. sound level**
High freq. condenser	Super-cardioid	800 ohms	25mV/Pa	130dB





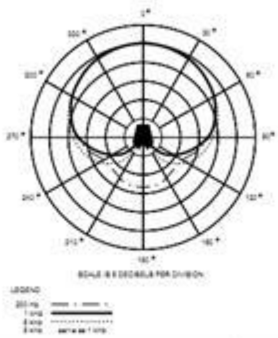
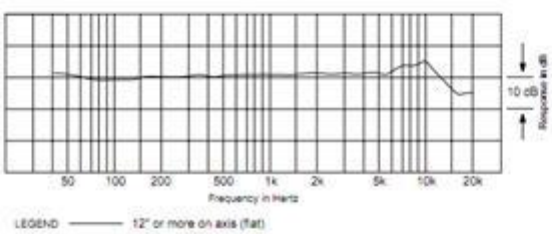
61. Technical specifications of Sound Devices 702T recorder.

Manufacturer: Sennheiser		**Model:** K 6		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ (open circuit)**	**Max. sound level**
Polarized condenser	Super-cardioid	200 ohms	50mV/Pa	126dB

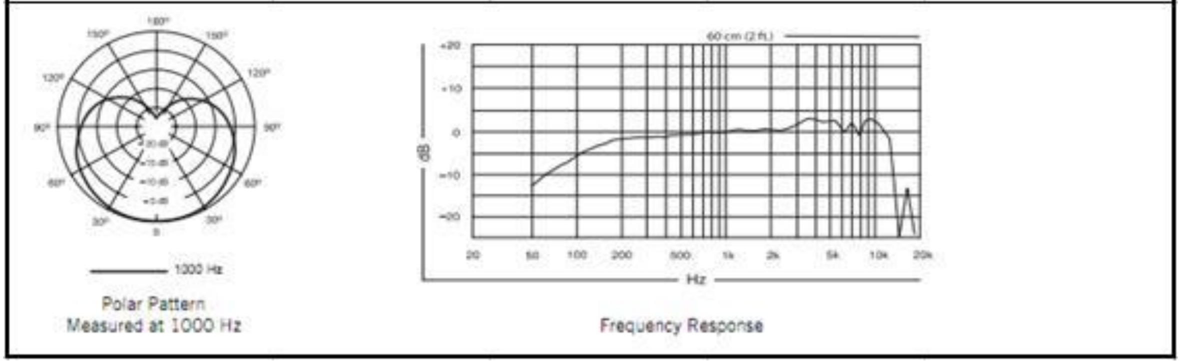
62. Technical specifications of Sound Devices 702T recorder.

Manufacturer: Audio Technica		**Model:** AT 871R		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ**	**Max. sound level**
Permanently polarized condenser	Half-cardioid (sup.hemisphere)	200 ohms	22,4mV/Pa	130 dB SPL

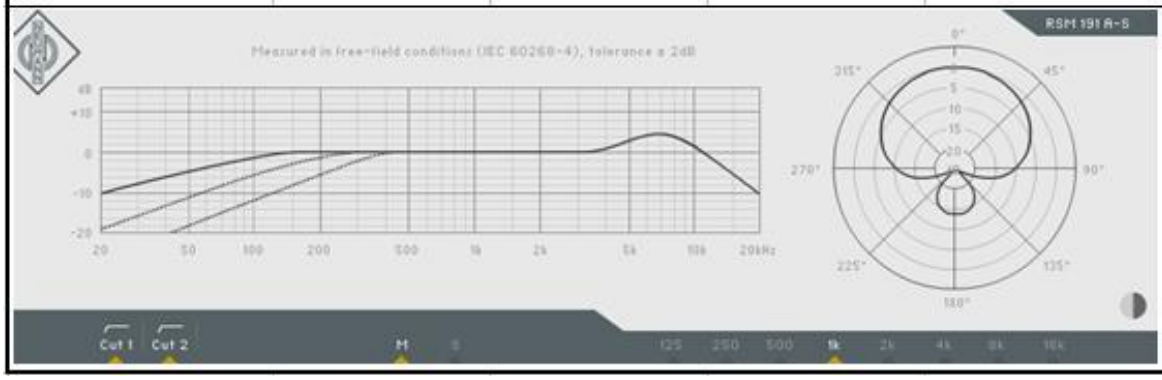
63. Technical specifications of Sound Devices 702T recorder.

Manufacturer: Shure		**Model:** PG58		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ**	**Signal/noise ratio A weighted (ref 94dB SPL)**
Dynamic: moving coil	Cardioid	300 ohms	2,2mV/Pa	



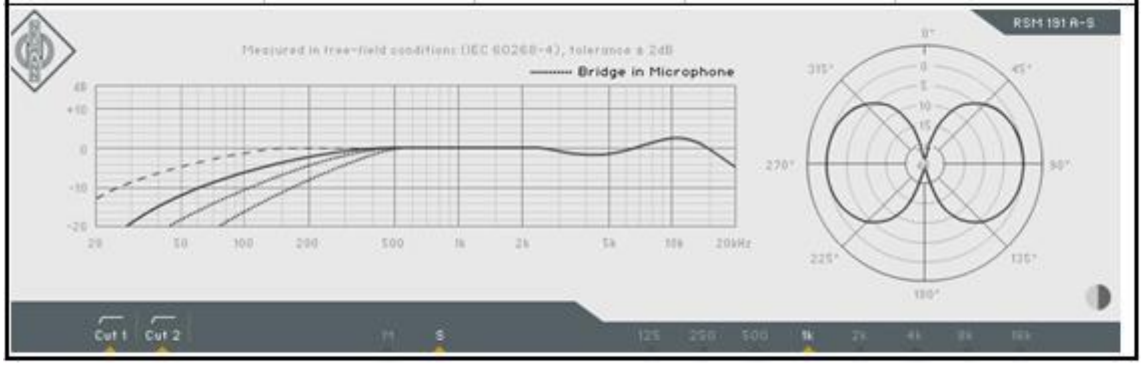
64. Technical specifications of Sound Devices 702T recorder.

Manufacturer: Neuman		**Model:** RS191 Stereo (M)		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ**	**Signal/noise ratio A weighted (ref 94dB SPL)**
Pressure gradient	Cardioid	1 kohm	23mV/Pa	78dB



65. Technical specifications of Sound Devices 702T recorder.

Manufacturer: Neuman		**Model:** RS191 Stereo (S)		
Acoustical operating principle	**Polar pattern**	**Rated load impedance**	**Sensitivity 1kHz, 1kΩ**	**Signal/noise ratio A weighted (ref 94dB SPL)**
Pressure gradient	Bidirectional	1 kohm	23mV/Pa	72dB



66. Technical specifications of Sound Devices 702T recorder.

III. CD TRACKLIST

T01 – Open air – pos1 – DPA4006
T02 – Open air – pos1 – DPA4011
T03 – Open air – pos1 – EV RE50
T04 – Open air – pos1 – Sennheiser MKH416
T05 – Open air – pos1 – Sennheiser K6
T06 – Open air – pos1 – AT 871R
T07 – Open air – pos1 – Shure PG58
T08 – Open air – pos1 – NewmannRSM191M
T09 – Open air – pos1 – NewmannRSM191S
T10 – Open air – pos2 – DPA4006
T11 – Open air – pos2 – DPA4011
T12 – Open air – pos2 – EV RE50
T13 – Open air – pos2 – Sennheiser MKH416
T14 – Open air – pos2 – Sennheiser K6
T15 – Open air – pos2 – AT 871R
T16 – Open air – pos2 – Shure PG58
T17 – Open air – pos2 – NewmannRSM191M
T18 – Open air – pos2 – NewmannRSM191S
T19 – Open air – pos3 – DPA4006
T20 – Open air – pos3 – DPA4011
T21 – Open air – pos3 – EV RE50
T22 – Open air – pos3 – Sennheiser MKH416
T23 – Open air – pos3 – Sennheiser K6
T24 – Open air – pos3 – AT 871R
T25 – Open air – pos3 – Shure PG58
T26 – Open air – pos3 – NewmannRSM191M
T27 – Open air – pos3 – NewmannRSM191S
T28 – Recording studio – pos1 – DPA4006
T29 – Recording studio – pos1 – DPA4011
T30 – Recording studio – pos1 – EV RE50
T31 – Recording studio – pos1 – Sennheiser MKH416
T32 – Recording studio – pos1 – Sennheiser k6
T33 – Recording studio – pos1 – AT 871R
T34 – Recording studio – pos1 – Shure PG58
T35 – Recording studio – pos1 – NewmannRSM191M
T36 – Recording studio – pos1 – NewmannRSM191S
T37 – Recording studio – pos2 – DPA4006
T38 – Recording studio – pos2 – DPA4011
T39 – Recording studio – pos2 – EV RE50
T40 – Recording studio – pos2 – Sennheiser MKH416
T41 – Recording studio – pos2 – Sennheiser k6
T42 – Recording studio – pos2 – AT 871R
T43 – Recording studio – pos2 – Shure PG58
T44 – Recording studio – pos2 – NewmannRSM191M
T45 – Recording studio – pos2 – NewmannRSM191S
T46 – Recording studio – pos3 – DPA4006
T47 – Recording studio – pos3 – DPA4011
T48 – Recording studio – pos3 – EV RE50
T49 – Recording studio – pos3 – Sennheiser MKH416
T50 – Recording studio – pos3 – Sennheiser k6
T51 – Recording studio – pos3 – AT 871R
T52 – Recording studio – pos3 – Shure PG58
T53 – Recording studio – pos3 – NewmannRSM191M
T54 – Recording studio – pos3 – NewmannRSM191S

T55 – Living room – pos1 – DPA4006

T56 – Living room – pos1 – DPA4011

T57 – Living room – pos1 – EV RE50

T58 – Living room – pos1 – Sennheiser MKH416

T59 – Living room – pos1 – Sennheiser K6

T60 – Living room – pos1 – AT 871R

T61 – Living room – pos1 – Shure PG58

T62 – Living room – pos1 – NewmannRSM191M

T63 – Living room – pos1 – NewmannRSM191S

T64 – Living room – pos2 – DPA4006

T65 – Living room – pos2 – DPA4011

T66 – Living room – pos2 – EV RE50

T67 – Living room – pos2 – Sennheiser MKH416

T68 – Living room – pos2 – Sennheiser K6

T69 – Living room – pos2 – AT 871R

T70 – Living room – pos2 – Shure PG58

T71 – Living room – pos2 – NewmannRSM191M

T72 – Living room – pos2 – NewmannRSM191S

T73 – Living room – pos3 – DPA4006

T74 – Living room – pos3 – DPA4011

T75 – Living room – pos3 – EV RE50

T76 – Living room – pos3 – Sennheiser MKH416

T77 – Living room – pos3 – Sennheiser K6

T78 – Living room – pos3 – AT 871R

T79 – Living room – pos3 – Shure PG58

T80 – Living room – pos3 – NewmannRSM191M

T81 – Living room – pos3 – NewmannRSM191S

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