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The peculiar placement of the Lavalier microphone as a way to improve the signal to noise ratio and speech intelligibility during the audio recording

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Acoustic conditions

It is not a closely guarded secret in the professional field that acoustic conditions for speech and for music are not the same. The communication process in terms of speech in quite complicated in the concert hall, whereas playing music in the conference room leads to the negative feedback of the audience; therefore, there are many professional sound designers and music producers who underline the significant role of the parameter named the reverberation time to describe the roots of dissatisfaction. There is a common understanding that the parameter should be around 0,7-1,1 seconds for speech and 2,1-2,5 seconds for music. As a TV sound producer and a recording engineer, I do agree that speech intelligibility is the «key point to improve» (Bruce Bartlett) if there is a task to build the TV studio to broadcast in that avoiding echoes is a paramount requirement; thus, the reverberation time has to be taken into consideration by the developers. Nonetheless, other factors have to be reckoned seriously to avoid possible dissatisfaction of the audience. One of the most significant points is a concern to control early reflections, due to the fact that, if there is the only one host in the TV studio, the direct sound is strong enough to satisfy the needs of the sound producer, but if there are more than 5 speakers in the room, speech intelligibility might easily be a problem in that acoustic conditions start playing the main role in the recording process. Therefore, under such circumstances, the peculiar close to the source of sound microphone placement is the key solution to improve intelligibility and to cut unwanted reflections.

Noise

The second factor is the signal to noise ratio. Noise is able to invade the studio or the recording room in a number of different ways. First of all, there is the airborne one. The modern studio is a complex structure with numerous sources of unwanted audio signals such as running ventilation plants, audience breaking noise, traffic noise, etc. It is significant to remember that should the broadcasting engineer doubles the distance from the source of noise it improves the signal to noise ratio to about 6 dB. Also, the second way noise might appear in the recording room is via transmission through «solid structures» (Bruce Bartlett), such as reinforced concrete walls and the floor. There are numerous well-known solutions to plummet the noise level, such as installing solid-insulating doors and double windows, insulating walls and «redesigning the ventilation system» (Michael Barron), but under some circumstances the well-known solutions might not be implemented due to a multitude of different reasons; thus, a specific close to the source of sound microphone placement is able to improve noise to signal ratio and increase the average quality of the broadcasting.

Human speech

The basics of any language in the world are consonants and vowels. Vibrations of vocal chords are produced with lungs. Resonant cavities in the throat, mouth, and nose, as well as movements of the tongue and lips, are also included in the process. It is known that vowels are louder and have a longer duration than consonants, but consonants are more impulsive; despite, their duration is shorter; consequently, the longer the reverberation time is the stronger the process of masking consonants by vowels. The human voice includes numerous frequencies with a huge range from around 75 Hz to about 7500 Hz., and there is an obvious difference between the male and female voices in that the male one possesses more low-frequency energy. This difference occurs seeing that a longer process of «opening of the male chords» (Michael Barron). The key point to take into consideration is the fact that the sound of speech does not have the same power and strength in all directions. Mostly this effect happens because the human head shadows the spectrum. Low frequencies from 50 to 250 Hz have a less directional effect, but from the range around 1000-1200 Hz, there is a significant directionality. It means that the human voice radiates different tone quality or spectrum of frequencies in all directions. The distance between the microphone and the mouth is imperative to choose in the correct way seeing that the human timbre is highly recognizable, the anchors of news, correspondents, and other celebrities don't want to lose it. This uniqueness exists because of specific relationships between fundamental frequencies and harmonics; consequently, the wrong choice might lead to loosing identity of the voice. It is obvious that the human voice sounds best at the distance since its essence, but under conditions of high levels of noise and unwanted reflections from the walls in the studio, the ordinary placement of the microphone leads to the counterproductive signal to noise ratio in terms of the recording process and unsatisfactory results in terms of voice intelligibility. In this case, the peculiar microphone placement improves intelligibility under complicated acoustic circumstances but leads to boosted low frequencies in the mix.

Proximity effect

Any microphone is a transducer that is designed to convert sound waves to electrical signals and «proximity effect detection is more than a simple analysis of the low-frequency content of the input signal» (Joshua Reiss). The omnidirectional microphones have a flat frequency response, so they are designed to pick up and convert sound pressure from all angles more or less equally. The directional microphone has its own feature; to pick up sound waves from a particular direction, so «microphones are made directional by controlling where the sound pressure arrives» (Joshua Reiss); thus, such transducers are widely used to improve the signal to noise ratio in the media industry. Indeed, this strong directionality leads to many issues with the flat frequency response, and as a result, there is a significant boost of low frequencies in speech, so additional equalization of the voice is imperative. Moreover, «proximity effect is not only an issue for close sources but also for distant ones in many cases»(Josephson, David); thus, controlling the distance between the source of sound and the transducer is the key factor to preserve recognition of the voice and to improve intelligibility. Anyway, we have to admit that due to the complexity of the changes, the distance between the success and failure is relatively short. Moreover, the proximity effect might be a reason of «distorting the audio signal» (L. Millot) in that the boost of the overall amplitude stems from the increase of low frequencies in the spectrum of the signal. Indeed, there are numerous different ways to tackle the problem. For instance, there are notch filters that effectively cut boosted low frequencies. This solution is not popular among professional sound engineers as the human voice has a complicated structure and cutting a number of low frequencies at once leads to significant changes in terms of timbre. The problem is to separate what frequencies to cut and what ones to leave without changes. Also, there are specific microphones on the market that have two separate diaphragms; thus, the second solution is the polar pattern that can be chosen by the sound engineer to fight against the proximity effect. This type of gear has the only one minus. It is heavy and bulky to be placed in front of anchors of news and guests during the broadcasting, so the Lavalier microphones are widely used due to the small size and some moderate quality of the sound they pick up; consequently, this type of gear with the omnidirectional and cardioid patterns is going to be used in the research. It is imperative to remember that every transducer has its own range of frequencies to be boosted if the gear is placed in a short distance nearby the source of the sound.

Method

Every media outlet, a news agency, or a broadcasting company has a number of different Lavalier microphones to satisfy the needs of the sound producers and broadcasting engineers. Measuring the distance between the microphone and the source of sound is the first step in the line. Every room, studio, or stage has its own pattern of reflections, and seeking out the right balance between unwanted ones that make the sound boomy and the direct signal that facilitates to improve intelligibility and clarity is the key task to execute. During the research, the detection algorithm was tested by recording different audio signals with omnidirectional and cardioid transducers. The distance between the source of the sound and the microphone was changed over time. The amplitude of the signal from the source is preserved constant and the output amplitude differentiates in that there are changes in terms of the distance between the source of sound and transducer. The omnidirectional microphone records all spectrum of frequencies equally with distance. Thus, there is no proximity effect. The amplitude of the audio signal simply increases linearly. The proximity effect on the cardioid transducer is meticulously documented. Consequently, there is proximity effect on the cardioid microphone, and there is no recognized one on the omnidirectional transducer. According to the measures that have taken place so far, the Lavalier microphone should be placed between 5 and 8 centimeters from the source to pick up a good amount of the direct sound and to avoid unwanted reflections if the reverberation time is more than 2 seconds. This distance is also good to improve the signal to noise ratio supposing the problem takes place.

During the second step, the sound designer finds ways to detect peculiar frequencies that provoke the proximity effect. There is a general understanding that the proximity effect mostly «happens between 400 and 500 Hz» (W. Dooley), so the spectral flux method is imperative to use in order to find and cut the unwanted and boosted parts of the signal. As we have discussed above, there are advantages and disadvantages to placing the Lavalier microphone as close to the source of sound as possible. In terms of recording human voices, such a placement does facilitate the sound producer to record successfully in a noisy environment and with some limitations avoid picking up lots of unwanted reflections from the surfaces in the studio, but the proximity effect, as well as possible “popping”, are among the main issues to avoid; thus, it is understandable that finding and cutting specific low frequencies from the signal is the ultimate priority to have a recording with satisfactory clarity and intelligibility. As the proximity effect is «a spectral effect, certain spectral features can be extracted» (Joshua Reiss). Measurements are imperative to have correct equalization. According to other researchers in the field, every Lavalier «microphone exhibits the proximity effect in different ways» (Clifford), so the measurements are valid for particular models.

Improving intelligibility by adding high frequencies is the last step in the line. It is paramount to know that some models have a specific boost within a range approximately 3-6 KHz in the output spectrum of the transducer to compensate a lack of clarity. Such models are beneficial for male voices and counterproductive for female ones since unwanted sibilants in the spectrum might occur. It is significant to understand that after first two steps there might be a lack or transparency in the voice, so additional boosting for male voices between 4 and 6 KHz might be a requirement to improve intelligibility. Nonetheless, there is no room for the approach that works well under all circumstances. Finally, the broadcasting engineer needs to take into consideration that adding some high frequencies using equalizers is a miles easier way to add some brilliance in the signal than the way of cutting distorted sibilants from the spectrum of the audio.

Conclusion

In all, the close microphone placement has its advantages and disadvantages. However, if the studio is noisy, there are many unwanted reflections, and should the reverberation time is bigger than 2 seconds, the broadcasting engineer is able to use the spectral flux method to detect the low frequencies that decrease intelligibility and clarity of sound and place the microphone in the peculiar way close to the source of sound in order to improve the quality of the broadcasting. At the same time, additional equalization is a significant point to take in order not to lose the unique tumble of the voice and to have a satisfactory broadcasting result.

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