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ORIGINAL RESEARCH ARTICLE

Wireless hearing aids for a theater show in Costa Rica: System design for 96 spectators

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ABSTRACT

Headphones allow to critical listen captured and recorded sound thanks to their ability to isolate the listener from the environment. This fact creates an intimate space that helps to immerse the listener into the aural experience. Providing headphones to Costa Rican theatrical shows attendees might help to increase acting dialog's intelligibility by using dedicated microphones, reduce outside undesirable noises that distract audience from on stage representation and to enhance the show's sound stimuli. There are no records of theatre headphone shows in Costa Rica, then it is difficult to evaluate its artistic impact. Given the fact there is no technical documentation that helps to replicate such a system, this paper addresses this issue by proposing a wireless headphone system design.

Keywords: headphones; wireless; theater; design; sound

1. Problem to be solved

As a sound designer for theatrical shows, my task is to provide the audience with a sound experience that allows them to connect emotionally with the events of the characters that act in the scene. This experience is achieved by designing and creating the sound stimuli to be reproduced in the show, through an audio system also designed, calibrated and operated under the aesthetic criteria agreed between the director of the production and myself. This type of auditory experiences is achieved through loudspeakers, arranged in most of the audio systems installed in the theaters of our country. However, these audio systems do not always satisfy the creative needs of Costa Rican theater directors. In this regard, I was consulted by colleague Natalia Mariño, theater director, about the need to have a sound system based on wireless headphones for a show in 2018, unfortunately, which was not possible to realize. Nevertheless, the concern to know if it was possible to create such a system remained latent. I set myself the task of researching our Costa Rican historical references to find out if in the past any theatrical show with the same or similar characteristics had been done. Unfortunately, it was not possible to find a record on this subject, so the present document poses as a problem to design a sound system that allows the transmission of its elements wirelessly, to each of the headphones of the spectators present in a show, using professional audio transmission devices, for sale in our country. The problem in question evaluates the feasibility of the components, the interaction of these

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with other devices of the system, the cost of acquiring or implementing it, which foresees the conditions of many independent theaters in our country, which seek to expand the sensory experience of their audiences. The problem presented in this paper tries to solve and anticipate situations at a technical level, but does not analyze or evaluate its aesthetic impact, since there is no show in which this system has been tested. However, it is hoped that the design of the sound system and its technical possibilities will inspire other sound designers and theater directors to approach, implement and explore this alternative in their shows.

2. Design approach: How the signal will flow in our wireless hearing system

Although wireless hearing aids are the focus of our proposal, I must point out that for them to have meaning or value as an active element of a theatrical performance, they must be placed in context and evaluated as part of a system. That is, I will not talk about hearing aids in isolation engineering and science that explains how hearing aids emit sound, but how they are hypothetically treated as a function of a theatrical performance. Understanding that hearing aids are connected to other devices, it is necessary to emphasize the components that make up the chain. The signal (hence, the narrative of this paper) will travel as follows theatrical room → actors/microphones → computer playback of sound sources → mixing desk → wireless transmitters → wireless receivers → headphones. Having made this brief clarification, let us move on, then, to quickly review some experiences of this type in theatrical performances in other countries.

3. Some background on headphone-based audio systems for theatrical audiences

The transmission of the sound of a show through the use of headphones is becoming a common trend in some British, German and American theaters. This is due to the benefits of headphones to create a more intimate and close sound environment than speakers. In that sense, Matt Trueman comments in the British newspaper *The Guardian:*

> In these theatre pieces, listening is a part of the overall experience, rather than the whole. Where sound is transmitted live, as in David Rosenberg's Contains Violence (where the audience watches the action through binoculars from a roof top across the street, hearing the dialogue through headphones), this sound-vision relationship makes sense. The audio device becomes a way of amplifying the dialogue, like a one-way walkie-talkie [In these theatrical pieces, listening is part of the experience and not the whole. When the sound is transmitted live, as is the case in David Rosenberg's Contains Violence (where the audience watches the action through binoculars from a roof top across the street, hearing the dialogue through headphones), this sound-vision relationship makes sense. The audio becomes a form of amplifying sound, like a walkie-talkie][1].

Martin Gimenez recognizes in his blog No Proscenium that "sound is the one design element that easily projects beyond the fourth wall-be that a stage or screen-and immerses the audience"[2]. Thus, sound in theatrical performances can create an immersive sensation through 5.1 systems, as in cinema (an experience that is already possible in our University Theater of the University of Costa Rica) or through binaural sound. Giménez also highlights the combined use of binaural headphones-microphone in the show. The Encounter by Simon McBurney, premiered in 2016 on Broadway, to intensify the spatiality and realistic effect of human hearing. About the same show, Gareth Fry and Pete Malkin, the show's sound designers, state that, "this is the first show on Broadway to use binaural sound, as far as we're aware. And also probably the first where the audience wears headphones [it is the first show on Broadway to use binaural sound and perhaps the first where the audience wears headphones]"[3]. The Encounter recreates the story of National Geographic photographer Loren McIntyre, who strayed deep into the Amazon forests in 1969. The show uses sound as its only dramatic device. Although the only actor on stage, Simon McBurney himself, performs in front of the Neumann KU-100 binaural microphone, the sound designers exhaust every available sound resource (sound effects, music, live voiceovers) to literally create the Amazon and Loren's adventures in people's minds.

As I explained at the beginning of this document, I have never experienced a show in which the acting dialogues, music, environments and sound effects have reached my ears through a pair of headphones, whether connected to a device or wirelessly. However, my colleague, director Natalia Mariño, who was recently in Germany, had the opportunity to experience the show Beute Frauen Krieg, directed by Karin Henkel[4]. The show, which recounts the events of the Trojan War, from the perspective of a woman, uses "custom-made" wireless headphones to allow the audience (about three hundred spectators), to move throughout the space where the theatrical event takes place. This is logical because of the inconvenience that cables represent for the displacement in a space and, above all, when there are three hundred people passing through the enclosure. According to Mariño, the entire sound design (including the dialogue between the actors) was emitted through these wireless headphones to the entire audience, who followed the dramatic action by moving to stations located at different points in the room.

As a theater director, Mariño sees a great benefit in the use of headphones for the spectators because it allows the vocal technique of the actors to be more intimate, cinematographic if you will, and that these parliaments are easily heard by the spectators. Of course, the fact that the headphones are wireless enhances the idea of designing a work in which the spectators can move around and the sound "follows" them everywhere, without losing any sound detail. In other words, displacement of the audience with the sound wirelessly. But, without limiting ourselves to these, the technology that makes possible the wireless transmission of sound, at least from the sound table to headphones on stage, is not something alien or novel just have a fixed transmitter and a wireless receiver that allows headphones connect directly. Personal monitoring systems or IEM (in ear monitor, for its acronym in English) allow musicians and actors receive messages and sounds discreetly and wirelessly. This is without the need for floor monitors on stage, which many directors and musicians find annoying because they are in the line of sight of the audience. When it comes to four or five musicians on stage, the system is not complicated at all and it projects just as simply for a hundred or two hundred spectators. We will explore, then, the possibilities that these systems offer to design a wireless headphone system applied to a fictitious staging of Hamlet, where, in addition, we will value wireless and binaural microphones as part of the system. Based on the above assumption, perhaps it can inspire another colleague in sound design in Costa Rica, who needs a starting point to create a similar transmission system.

4. Why IEM systems and not others?

To design a sound system for a theater show, in Costa Rica, based on headphones, we can anticipate two categories wired or wireless. In this article we will explore the wireless option. In short, the wired system can offer a good cost-benefit ratio, since it requires less investment and equipment acquisition. However, the wired system can be inconvenient, due to all the lines that would have to be distributed throughout the listening area, including the hearing aid cable itself. The wireless system, much more attractive these days because we are already used to our cell phones and mobile devices, demands a strong investment in equipment and accessories to achieve the system that, in this case, would eliminate all kinds of cables, except the one that connects the receiver to the hearing aid. Within the wireless systems there is a variety of transmission protocols, but not all of them are convenient. For example, systems based on infrared or IR light (not visible to the human eye) are cheaper, but they have a low sound quality and, since they have shorter wavelengths compared to radio waves, they need to have an obstacle-free transmission path between transmitter and receiver. Otherwise, the transmission will be interrupted just by passing the hand in front of the receiver. The other major contender is Bluetooth, which, for starters, has a latency that is hard to ignore, is easily accessible to anyone with a cell phone and can only pair with one receiver at a time. In addition, its protocol does not allow audio transmission in Wave format[1] at sampling frequencies of 44.1 kHz[2] in versions prior to 4.0.

Finally, it is not a protocol recommended for professional applications, because according to Huntington: "Bluetooth is a network primarily designed for short-range connection of peripheral devices, such as wireless headset to a cell phone. It's not well suited for the rigors of the show use, but you might find it useful to connect simple components to a show computer [Bluetooth is a network primarily designed for short-range connection of peripheral devices, such as wireless headset to a cell phone. It is not well suited to the rigors of a show, but it could be useful for connecting simple devices to a show computer]"[5]. It is worth noting that many bluetooth devices turn off automatically, when they stop receiving signal or activity. In the case of some bluetooth speakers, they go into "sleep mode" even when they receive audio signal, but it is sent at a low volume (e.g., if a cell phone is connected to a bluetooth speaker from the 3.5 mm audio output or headphones via a cable). This is a disadvantage for a show, as it cannot be guaranteed to operate continuously without shutting down in the absence of a signal.

Therefore, we come to radio frequency systems, where we find the IEM devices, which have proven to be more reliable professional audio level for decades. They also allow greater control and configuration of the devices. That is why the option of these systems will be analyzed in this paper, as an alternative for a wireless headphone audio system in a theatrical show. The fundamental difference between the IEM system and Bluetooth lies mainly in the fact that the audio does not require digital conversion, since it works with radio frequencies. Another advantage is that a single transmitter can send a signal to many transmitters at the same time and all they need to achieve this is to be tuned to the same channel. It works in the same way that a conventional radio tunes into a radio station to listen to your favorite program. Another advantage is that, if signal amplification is necessary, antennas can be added inside the theater to avoid signal loss or interruptions in the room. These and other features add value to the use of this wireless option. To develop my argument, I will create a fictitious situation for a real theater. An Italian show attended by ninety-six spectators, which is the usual number of people that the building can accommodate at "full theater". The venue we will use as a model will be the University Theater (UT) of the University of Costa Rica. On this building and its particularities we will design a headphone audio system for our fictitious staging of Hamlet. Let the show begin!

5. The University theater presents hamlet (theater hall → actors/microphones)

Whether wireless or not, hearing aids offer isolation from the acoustic environment, a phenomenon described as the "walkman phenomenon". Hosokawa describes how the Sony Walkman introduced,

[1]Wave, together with AIFF, is the format defined as a professional standard by the AES (Audio Engineering Society) for digital audio. Its specifications are defined in the AES "Red Book" and, in simple terms, define the minimum quality of Wave as follows: 44.1 kHz as sampling frequency and 16-bit resolution. These parameters apply for audio transmission over the Internet or in its duplication for compact discs.

[2]44.1 kHz was defined as the minimum standard sampling frequency for professional audio files for compact discs and, later, for Internet streaming audio files. This frequency is the result of the Nyquist theorem, which is beyond the scope of this paper. However, the reader can refer to chapter 35: DSP Technology of the Hand-book of Sound Engineer (see references) where highly detailed information behind the mathematics and techniques of digital sampling can be found.

not only the portability of sound and its control, but also, a kind of "secret theater" or "headphone theater"[6]. To achieve this "secret theater" with headphones, we must capture the voices of the actors in real time through microphones and this capture, in turn, must be sent to the audience's headphones. Before, we will define some parameters regarding the sound sources of our Hamlet show:

- The maximum recommended audience capacity for the University Theater, which will be the theater we will use as a reference, is 96 spectators. Therefore, the design contemplates this number of people.
- The most important source of pickup and delivery is the dialogue between the actors and actresses. This could be done by means of microphones, which capture the sound in different ways wireless lavalier monaural (for a stereo mix), or binaural (for a binaural mix). It is possible to combine both types of microphone.
- The music of the show will be previously produced and, then, played in the show. Not performed live.
- Sound effects, except those produced by the actions of the characters on stage in a natural way, will be previously produced and then reproduced in the show.
- The sound environments will be previously produced and then reproduced in the show.
- All produced and pre-produced sound sources from the show will be sent to the mixing desk, from which a single mix will be transmitted wirelessly to all the audience's headphones.
- Speakers shall be placed in strategic positions, within the stage, to provide the cast with reference to the feet of sound present in the show. The volume of these speakers should be carefully adjusted so that it does not leak into the actors' or stage microphones and create an "echo" in the audience's headphone mix.

In defining the above parameters, in this document, we will explore conventional wireless mono channel lavalier microphones and a binaural microphone the Neumann KU-100. Before deciding on one or the other microphone, it is important to analyze the space where the theatrical event in question will take place. By this, we refer to the ability of the theater building to not allow the entry of outside noise. In a simple way, we can define noise as those sounds that we are not interested in hearing, as opposed to the signal, which are those sounds that we do want to hear. In our case, external noises are motorcycles passing by on the street, horns of different automobiles, airplanes and helicopters passing by, urban train horns or conversations of passers-by. Another way of looking at noise is as that sound that does not belong to the world of on-stage representation. If for us, the story of Prince Hamlet takes place in a medieval castle, none of the aforementioned outside noises fit into our medieval sound world, and therefore will distract the audience.

Likewise, noises generated by the cooling systems of the mobile luminaires, smoke machines, footsteps or conversations behind the scenes or electrical humming emitted by the loudspeakers as a result of damaged or unbalanced cables, are the type of internal noises we want to eliminate or mask. Speaking specifically of the University Theater (UT), the building has serious acoustic insulation problems. Most of the external noises mentioned above enter with very little attenuation and, of course, the same happens the other way around the sounds generated inside the hall come out without attenuation. Recognizing this problem is very simple. It is enough to hear the horn of the 5:00 p.m. intercity train (the UT is located about 400 meters from the nearest station) or a motorcycle with an altered muffler passing by on the street in front of the theater, to understand that any external noise implies internal filtration. Of course, these noises interrupt and distract the spectators of the show because they do not belong to the world or the events of the play and happen randomly. The truth is that the microphones we use to capture the actors' voices will also pick up these intrusive noises to a greater or lesser extent. So, for a conventional stereo mix in our hypothetical production of Hamlet at the T.U., we would opt to use wireless lavalier microphones, using the technique typical of musicals: the microphone as close to the actor's or actress's mouth as possible. In contrast to the cinema technique, which would be on the chest. The technique of the musical ensures a better "signal/noise ratio"[3] , which means that we will have more present the voice of the interpreter at the expense of sounding less "natural" and that the microphone is minimally visible at a distance. Usually, each performer would have a dedicated microphone for his or her voice and this would mean a dedicated sound channel on the mixing desk, i.e. microphone 1 to channel 1 and so on. The usual practice in stereo mixing is to keep the heaviest sound sources located in the center of the stereo image. That is, 0% left/right. As an example, we can cite almost any commercial song. The vocalist's voice will be located in the middle of the two headphones to maintain its prominence throughout the song.

For many sound engineers, lavalier microphones are complex to handle in live performance because they tend to generate a lot of feedback (that squeal you hear when a microphone is close to or near a speaker) due to their omnidirectional polar pattern[4]. By performing the correct PAGNAG[5] calculations and adjusting the position and tilt of the speakers in a system, feedback can even be eliminated[7]. Since our setup is transmitted through headphones, the PAGNAG increases considerably, so feedback would no longer be an issue. This is why, lavalier microphones are projected as a good option. Let's remember that the objective is to deliver a very good signal/noise ratio, taking into account that our theater allows intrusive sounds to enter very easily. So, signal is everything. I am interested in capturing (such as the actress's voice) and noise is everything. I am interested in attenuating or masking with the signal (such as the train horn).

Now, if our hypothetical staging of Hamlet happens in a hypothetical acoustically isolated U.T., where intrusive noises are not a problem, we would explore the option of binaural sound by placing a binaural microphone such as the Neumann KU-100[8] center stage. These microphones omnidirectionally pick up sounds, but do so using a different method. The KU-100 consists of two microphones embedded in the ear canals of a mannequin's head, which even exhibits ears. This head, which has concavities like those of a real human head, makes it possible to bring the sound pickup closer to the way our human ears do. The shape of the ears, the ear canals, the hollow areas and the size of the head is what makes it possible for us to perceive sounds up, down,

3Signal to noise ratio is a technical parameter that explains how much noise a device inherently generates due to its components. Every device generates noise as part of the energy that is dissipated as heat and the value of this ratio is that the noise is very little in comparison to the captured signal. The term also applies when using a microphone (and assuming) with an exceptional signal-to-noise ratio to capture a sound source, noise is considered for all sounds other than the source. For example, if our intention is to capture the drum sound (signal) of a drum kit, all other instruments (bass drum, cymbals, hi-hat, unintentional stick strikes, drummer-generated sounds such as breathing) become noise. The goal is to capture the drum signal at very high amplitude or volume and the rest of the drums and noises mentioned above at low amplitude or low volume. The ways in which a signal can be isolated from noise depend very much on the type of microphone and its polar pattern, as well as the distance between the microphone and the source and, of course, the volume of the source itself.

4Microphones can be classified according to the way they pick up sound. Omnidirectional microphones (including binaural) pick up sound spherically, from all directions. In contrast, the conventional handheld microphone, such as the karaoke microphone, is a cardioid pattern microphone, or heart microphone, and its reception is effective by speaking directly into the microphone.

5PAG: Potential Acoustic Gain / NAG: Needed Acoustic Gain. Mathematical operation that allows estimating if a system has enough dynamic range to emit amplified sound before generating feedback. The greater the ratio of PAG to NAG, the greater the gain that can be given to a microphone without feeding back into the system.

forward and backward. By placing a binaural microphone in the center of the stage, the same sound environment of the stage in which the characters are performing is recreated in the headphones. Stereo sound, on the other hand, is heard in the headphones in a two-dimensional 180° arc (left, center and right) and depth can be created or accentuated using reverberations, which lacks the realistic sense of depth and localization that binaural sound generates. It should be noted that binaural sound pickups and recordings translate very well in headphones, but lose their three-dimensional effect when played back on speakers. What then would be the value of this type of microphone in a show? There are a number of videos on YouTube that are actually like small radio theaters made with binaural microphones, their greatest value lies in exhibiting the sensation of auditory spatial reality that they produce. One of the most common is that of the barbershop, where an Italian barber cuts the hair of the customer (in this case you, as you listen), while a friend of the barber plays the guitar. Even with the eyes closed, the brain is able to pinpoint the location and distance of each of the sources. I invite the reader to listen to one of these recordings to better illustrate what follows.

The objective of the headphones is to immerse the audience in the story being presented on stage. Using a single binaural microphone, located in the center of the stage, that captures the dialogues, steps and sounds generated by the actors in the scene, the spectator is audibly located in the center of the scene, generating a circular space. It is as if we were seating the spectator in the center of the stage and the cast was acting directly for him or her. A good example is the video on YouTube virtual reality for your ears[9]. From 4:52, the reader can get a clearer idea of how binaural audio is perceived and what the Neumann KU-100 microphone looks like. Also, why the center stage arrangement is optimal for capturing dialogue and on-stage actions. Of course, one could argue that two identical conventional microphones arranged to capture stereo can generate the same effect. But, as I mentioned earlier, the pickup techniques are different and both yield different results in terms of the spatial level of the sounds.

It is necessary to emphasize that all captured dialogues will be sent to the mixing console, regardless of whether the microphones are binaural, mono (single channel) or stereo. It is also necessary to clarify that no special recommendation is made about one format or another, but simply explain the differences between them and that both formats are equally valid and functional and even combinable. In addition, the success of the sound experience depends not only on the microphones and their location, but the mixing table (and of course, the person operating the table) plays a key role in the distribution of sounds, both in volume and space. In addition to the table may or may not record the sound of the show, but it is not a requirement. In short, if the show is done with conventional wireless lavalier microphones:

- One microphone is needed for each cast member, which increases the intelligibility of the actors, because the proximity between microphone and performer is constant.
- It would work best for an Italian-style or single-front show.
- Each microphone will occupy an individual channel on the mixing console.
- The microphone will pick up the interpreters' voices to a greater extent and, to a lesser extent, the acoustics and internal and external noises at the time of capture.
- Voices will remain located in the center of the stereo spectrum of the audience's headphones most of the time. Unless, manually, another location is available for a specific effect.
- To add spatiality and depth to the actors' microphones, one may choose to place two microphones in the center of the stage, in the center of the theatrical room, or both

and add these channels to the headphone mix. If microphones are added center stage, they should be masked or camouflaged with the set design so as not to draw attention to them, unless the sound designer and director agree to their exposure as part of the visual artistry of the show.

If the show is performed with a binaural microphone:

- A microphone is needed to capture the entire cast. However, it must be considered that speaking at a distance away from the microphone reduces the intelligibility of the actors' speech.
- It would work best for a three-front, semicircular show, or where the actors have their backs to the audience, without this being a drawback.
- The microphone will occupy two channels in the mixing console, which should be set to 100% right and 100% left, respectively.
- The microphone will pick up to a greater extent all sounds generated on the stage including its acoustics and, to a lesser extent, the sounds of the audience and the theater itself.
- There will be vertical and horizontal spatial correspondence between the voices of the actors and the one emitted in the headphones, without the need to add more microphones or make manual adjustments. In addition to giving a very realistic sensation of hearing.
- If this microphone is added center stage, it should be masked or camouflaged with the set design so as not to draw attention to it, unless the sound designer and director agree to its exposure as part of the visual plastics of the show.

Speaking a bit of economic investment, the Neumann KU100 binaural microphone (although expensive: $ 8,000), would represent a good alternative in a supposed cast of twenty actors and actresses, since a single microphone solves the pickup and spatial localization. Versus 20 individual conventional wireless microphone systems. Each wireless microphone system consisting of microphone, wireless transmitter, rechargeable batteries and wired receiver has a value of approximately $ 800. This results in about $ 16,000 in wireless microphones. If the budget exists, the combination of both types of microphones would even expand the creative capabilities of the sound designer. For example, you can use the binaural microphone for scenes in which there are several characters and if there is an aside, in that same scene, you could turn on only the wireless microphone of the character who plays the aside, to give greater prominence.

To conclude this section, it is necessary to remember that the success of binaural sound depends very much on how many undesirable noises enter the enclosure. It is even necessary to consider those generated by the theater's own equipment, such as lighting or stage systems, for example. Binaural sound represents a more "realistic and three-dimensional" proposal. Whereas, lavalier microphones offer better signal-to-noise ratio and definition of the dialogues at the expense of having a flatter and "less" three-dimensional sound.

6. Meanwhile, at the Front of House (computer playback of sound sources → mixing desk)

We receive from the microphones the signal to be mixed with other elements in the Front of and engineers who mix the sources and sounds conceived by the sound designer, and the wired and wireless technology that makes possible the transmission to the headphones. Next, we will talk about the Behringer X32 mixing desk, the Apple iMac computer that specifically has the Qlab 3 program

House (FOH)[6] . There, we find the sound operators installed[7] and the wireless components that will send the mix to the audience's headphones.

The Behringer X32 is a digital mixing console with 32 inputs, which can be assigned as analog or digital, 6 auxiliary inputs, 16 analog outputs, 2 monitor outputs, 32 direct digital USB outputs and 6 auxiliary outputs. This table, undoubtedly, allows to handle a large number of sound sources and send them to the stereo channels needed for headphones. The Apple iMac computer, running Qlab 3 software, communicates via USB with the X32. In a loudspeaker sound design, it is desirable to assign one Qlab 3 output channel per loudspeaker. But, in this case, one stereo channel would be needed for the whole mix. So, if we needed to connect 20 microphones (of the actors on stage) to the table, we would assign inputs 1 to 24 as analog (analog-card assignment can be done in banks of 8 inputs). Whereas, inputs 25 to 32 would be digital or "card". Since Qlab 3 allows individual volume control of each of the audio files imported into the working session, it is not necessary to have more than two channels assigned on the X32 to receive the entire mix from Qlab 3. So, channels 25–26 would be more than enough, assuming the show has no live music. Otherwise, we would be left with 4 free channels, 21–24, to add musical instruments if needed.

The next question is, how do we deliver sound to 96 headphones? It is understood, then, that the Behringer X32 mixing console receives the wireless signals from the lavalier microphones or the wired audio signal from the binaural microphone and combines them with the sounds reproduced from Qlab 3. In turn, these send the wireless signal to the headphones of the viewers. Of the total number of stereo outputs offered by the X32 mixing console, two are enough to feed 96 headphones at the same time, either stereo or binaural mix (both use two channels for pickup and playback). What differs is their capture method). Let us now comment on the IEM systems that will serve as transmitters and receivers of our wireless headphones.

7. Hamlet speaks in your ear: Powering 96 hearing aids (IEM transmitters)

The audio signal containing the entire mix comes out of Out 1–2 on the desk and connects to the fixed transmitter #1. For wireless transmission we will use the in-ear monitoring systems, or personal monitors that musicians often use to listen to themselves on stage during live performances. If you have seen concerts of your favorite artist on DVD or music channels and noticed that they wear some sort of headphones while playing or singing, well, then, you have seen an IEM system, specifically, the headphone connected to the wireless receiver. IEM systems consist of a transmitter (usually fixed and wired to some source) and a receiver (usually wireless and battery powered). These systems allow all musicians on stage to hear a mix dedicated to their particular needs; different from what the audience hears through the loudspeakers, wirelessly and with the

6Translated as "front of stage", it is the place within the stalls or seating area where the mixing engineer and, of course, the mixing desk are located. Traditionally, it is located at the midpoint of two speaker systems, usually left and right. Although for our type of mixing the location of the FOH is not so relevant, for live shows with loudspeakers or concerts, its position is crucial. There is a tradition in Costa Rica that FOHs are located inside control booths. These booths not only represent a distortion of the sound that actually reaches the audience because the mixing engineer or operator compensates for the sound deficiencies of their environment affecting the sound actually heard by the audience in the theater, but also because the booths often contain lighting system devices that generate noise by cooling fans. This factor of course makes it difficult to correctly mix the levels and tonal balance of the different sounds and music present in a show.

7Qlab 3 is a program for live show control and is exclusive to the Apple MacOS operating system. The program can control audio, microphones, cameras, video and send MIDI signals to remotely command other devices. Specifically in audio, Qlab 3 differs from audio recording programs such as Pro-Tools in that Qlab 3 only plays back, not records or creates sounds. Qlab 3 searches for audio files within the computer and allows you to play them simultaneously, each with its own volume level, panning, effects, fades and other functions. The sound designer programs each of the sound feet according to the needs of the show. For more information the reader can visit the site www.figure53.com ability to control the volume. But these systems do not stop there with a collective mix, because they add versatility to serve each musician independently and individually, if several transmitters are incorporated at different frequencies one for each one. In our case, we will take advantage of the ability to transmit a mix to all the spectators of our show, ensuring volume, spatial and tonal uniformity, as if we were tuning a radio station.

As I mentioned earlier, we will use two stereo outputs available from the X32 to send the signal captured from the microphones back to the headphones. On one of them will be connected a transmitter tuned to one frequency. On the other output, the second transmitter will be tuned to the same frequency as the previous one and will serve as backup equipment in case transmitter #1 fails. In this way, simply turn on transmitter #2 to re-establish communication between the transmitter and receivers again. For clarity, we will connect the stereo transmitters to the Out 1–2 and Out 3–4 outputs of the X32. The 96 wireless receivers (to which we will connect the headphones) will be tuned to the frequency of the transmitter. This will result in one transmitter feeding 96 (or more) IEM receivers simultaneously. According to the manufacturer Shure, the number that IEM receivers that can be connected to a transmitter is "unlimited"[10]. On the other hand, the manufacturer Sennheiser states that it is possible to tune "multiple receivers to the same transmitter," but does not specify a quantity[11]. Consulted on this subject, Professor Guillermo Rivero of the University of Costa Rica, specialist in radio frequencies and consultant of the Costa Rican Electricity Institute tells us:

This system (Shure PSM 900 and Sennheiser IEM G4) has the possibility of 9 operating frequency ranges, each range is 42 MHz bandwidth and each receiver has a bandwidth of 25 kHz, thus allowing the connection of 1680 devices by frequency synthesization.

Sigismondi reinforces Rivero's statement indicating that "with a wireless personal monitor system, however, the number of receivers monitoring that same mix is unlimited. Additional receivers do not load the transmitter, so feel free to add as many receivers as necessary without adding more transmitters [with a wireless personal monitor system, the number of receivers monitoring that same mix is unlimited. Adding receivers does not load the transmitter, so feel free to add as many receivers as you wish without adding more transmitters]"[7].

In Costa Rica, the objective of transmitting audio to 96 headphones can be achieved with either Shure or Sennheiser IEM systems, both of which are represented in the country. As explained above, for each IEM system we will need two transmitters and 96 receivers. For the Shure IEM system we will need 2 stereo transmitters model PT, 96 receivers model PRA, 244 AA 2,500 mA rechargeable batteries[8] (192 batteries to power all receivers and at least a ¼ part extra as a spare) and the headphones that connect to the receiver through a standard 3.5 mm connector[9] . If the sound designer opts for the manufacturer Sennheiser, the devices needed would be 2 transmitters model SR IEM G4, 96 receivers' model EK IEM G4, an equal number of 2,500 mA AA batteries and, of course, the 96 headphones.

The viewer will receive two components the wireless receiver and the earphones. The receiver should preferably be hung on the waistband of the pants or skirt. As a professional monitoring system, these are intended to be fitted on the fly, which would pose a problem for the designer and operators, as the controls and configuration of the device itself may be at risk to those audience members with "curious" hands. Fortunately, IEM receivers allow the operation of the entire device, including the volume knob, to be locked out. Some viewers may appreciate being able to control the output volume of the headphones themselves, although it is up to the designer to set the

8mA: milliamperes.

9mm: millimeters.

standard output level for the show, which we recommend to be 85 dB SPL, A-weighted on each of the headphones. This level should be marked at the point where the knob outputs the desired volume, so that the user can quickly return to the preset level. **Table 1** best summarizes how the signal will be transmitted from the table to the headphones.

In this way, we will be able to transmit all the audio captured through the microphones and sound played from Qlab 3 and those musical instruments that are connected to the table, directly to the headphones in a wireless way. Now that we have created a system, we have to talk about the last component of our wireless system the headphone.

8. Some technical considerations when choosing hearing aids for Hamlet

The first consideration is of a hygienic nature. As mentioned above, in Costa Rica you can get the Sennheiser EK IEM G4 or Shure PRA wireless receivers, which include a pair of earbuds as part of the purchase. But, the fact that these must be inserted into the ear makes them unattractive for our show. It must be considered that we will be lending in-the-ear headphones to many viewers over a performance season of no less than four weeks. Not only is it a complex task to remove earwax from 96 pairs of hearing aids at the end of each performance, but some of the audience will find it unpleasant to wear hearing aids that have been inserted into someone else's ears. We could argue that each viewer should be able to wear their personal hearing aids to avoid uncomfortable situations. But, part of designing the sound experience of a show is to ensure uniformity of sound and that implies having control and knowledge of the devices involved in the process. Therefore, we must consider a pair of headphones that are easy to put on, take off, clean and, of course, sound good. Understanding that now we must pay attention to certain specifications, to make the best choice.

Table 1. Connection of X32 outputs to stereo transmitters

Stereo output table X32	Stereo transmitter	Receivers/Headphones
Out 1–2	T1	1–96
Out 3–4 (Spare)	T2 (Spare)	1–96

Source: Own elaboration.

The second has to do with impedance. Impedance (which is measured in ohms and is represented by the Greek letter omega: Ω) is a force in opposition to the current passing through a circuit, when a voltage is applied. As a general and very simple rule in headphones, the lower the impedance of the preamplifier in relation to the impedance of the headphones, the lower the current flowing through the circuit, therefore, the higher the volume perceived by the listener. This rule can be expressed as a ratio of 1/8. That is, the preamplifier has an impedance of one-eighth of the impedance of the headphones[12]. Any of the wireless receivers Shure or Sennheiser mentioned above have output impedance of 32 Ω, so, an ideal headphone should be 256 Ω, according to the above statement. On the other hand, the theory behind the circuits indicates that matching impedances implies a maximization of the electrical energy transmitted to the headphone horns and, consequently, a distortion-free sound at a defined volume, which is not exempt from power loss[13]. However, the two approaches to matching impedances, described above, are almost never met to the letter, due to the large supply of consumer and professional audio devices with different circuit designs resulting in varying impedances. This fact impacts the sound of our performance as follows suppose a headphone A has 38 Ω, while a headphone B has 55 Ω. Each of these headphones will be connected to an IEM receiver that has 32 Ω. The knob controlling the volume on the IEM receiver is set to half its full turn, i.e., 12 o'clock. If you play the same sound through each of the earphones and measure it with a sound level meter you will notice that earphone B will emit a slightly lower level than earphone A, this due to the higher impedance of B, and will need a little bit more turn on the knob to match its volume level with earphone A. This should never be interpreted as an advantage of hearing aid A over B, but as a condition to be taken into account when deciding the average volume at which the whole show will sound.

The third specification is sensitivity. As with loudspeakers, greater sensitivity means greater ability to handle high sound pressures without distorting or even damaging the speakers. This parameter is measured as "n" decibels[10] sound pressure level or SPL[11] equivalent to 1 mW[12] . Take for example the AKG K240 Studio headphone. Its sensitivity is specified as 104 dB SPL at 200 mW. That is, when the power in watts of the device that feeds the hearing aid reaches 200 mW, the hearing aid will be able to produce the 104 dB SPL, but we will be pushing the hearing aid to its limits. This lets us know that this hearing aid will be able to generate this level for a very short period of time, or rather, at peak volume. Also, that taking the hearing aid to this level, constantly, will overload it. On the other hand, if we look at the specifications of the Shure PRA IEM receivers, we will notice that it is capable of generating 200 mW output. If we connect the AKG 240 Studio headphone to this receiver and calibrate its volume knob so that the constant output level is 85 dB SPL, we will obtain as a result that the receiver needs to generate 22 mW[14,15] . We see, then, that the wireless receiver can, in theory, supply enough power to operate the hearing aid properly. We also see that as we lower the volume, the power demand decreases exponentially.

The fourth specification to be considered in the selection of hearing aids is the frequency response and is defined by a graph showing the ability of the device to reproduce human audible frequencies. This type of graph is mainly used to illustrate how a microphone picks up sound or how a loudspeaker or hearing aid reproduces it. Very quickly, the user can tell whether the device accentuates some frequencies more than others. Even among commercial consumer headphones, it is possible to find such graphs. What usually separates consumer devices from professional ones is that the latter aim to have a "flat" frequency response. The term "flat" means that none of the frequencies or frequency bands that the hearing aid is capable of producing is increased or decreased. This is in contrast to consumer hearing aids where the goal is, rather, to preset response curves that accentuate and dampen certain frequencies to produce a pleasing sound. The main reason why professional audio needs and aspires to be as flat as possible is critical listening, in which the sound is evaluated for imperfections and corrections. By having predetermined response curves for pleasurable listening, consumer equipment fails as critical listening devices because it masks undesirable sounds or frequencies and gives a distorted sound image of what is actually sounding. However, the very physics involved in the phenomenon of sound dispersion, as well as the actual practical capabilities of the circuits and components, keep professional audio equipment away from the desired flat straight line, which is expected to be seen in a frequency response graph. Given this reality, it is accepted that a deviation of +/3 dB is considered flat, +/6 dB acceptable, and greater than these values as increases or attenuations of the frequencies in question. **Figure 1** shows the frequency response of the AKG K240 Studio headphone.

As shown in **Figure 2,** the frequency response is stable at 102 dB SPL from 70 Hz to 6 kHz. Then the hearing aid begins to lose its stability and accentuates frequencies between 7 kHz and 10 kHz. On the other hand, the hearing aid has difficulties in

10Decibel, or dB, is one tenth of a Belio and expresses a relationship between quantities. It should be noted that the decibel is not a unit of measurement, but the exponential change of energy. For example, a change of 20 dB means that the intensity increased 100 times, since the 2 of the decimals is actually 10 raised to the "n" power of the decimals, which in this case is 2.

11Sound pressure level or sound pressure level. A measure of sound volume intensity that can be calculated from atmospheric pressure. For example, 0 dB SPL, or the threshold of human hearing, is equivalent to 0.00002 Pascals, and the Pascal is considered the pressure exerted by the earth's atmosphere in square meters.

12mW: milliwatt or thousandth of 1 W, or Watt.

maintaining a uniform response in the low frequencies, specifically, between 20 and 69 Hz.

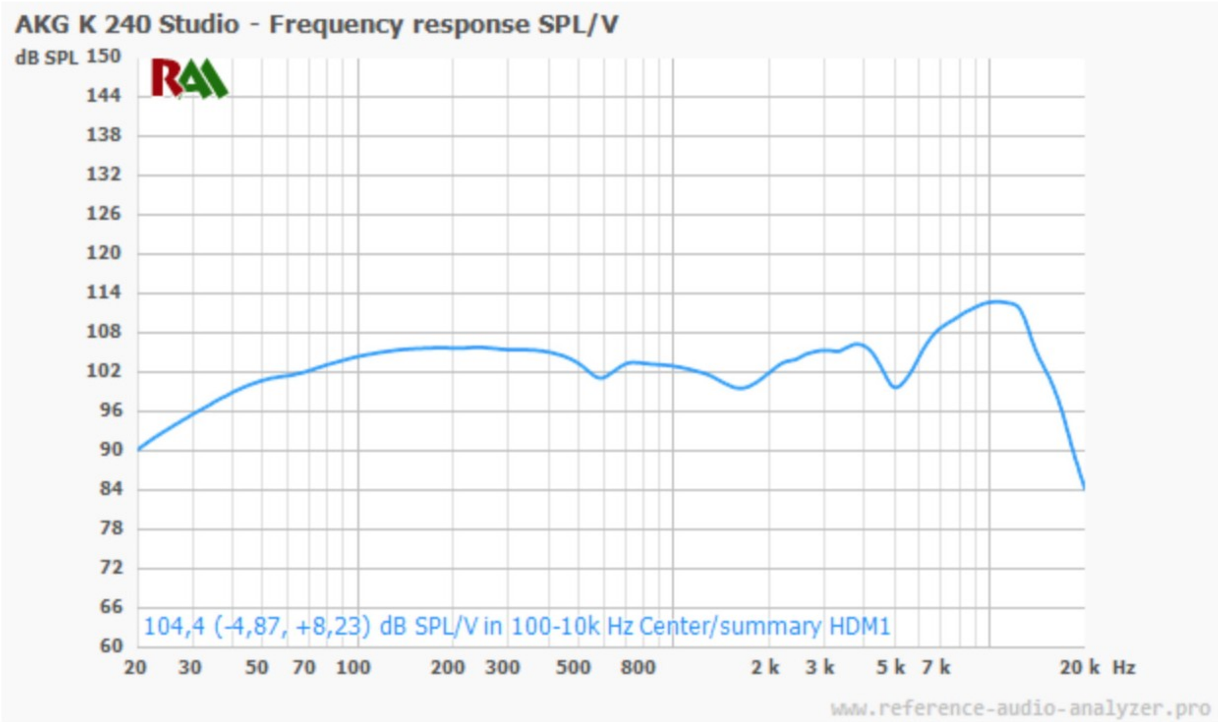

Figure 1. Frequency response of the AKG K240 Studio headphone.

Source: Device Measurement Reports: AKG K 240 Studio report[16].

Table 2. Recommended hearing aids

Specifications	AKG K240 Studio	Audiotecnica ATM M40x	Sony MDR 7506	Beyerdynamic DT 990 Pro
Type	Semi open	Closed	Closed	Open
Sensitivity	104 dB SPL	98 dB SPL	106 dB SPL	96 dB SPL
Power	200 mW	1,600 mW	1,000 mW	100 mW
Impedance	55 ohms	35 ohms	63 ohms	250 ohms
Reply	15–25,000 Hz	15–24,000 Hz	10–20,000 Hz	5–35,000 Hz
Disconnectable cable	Yes	Yes	No	No
Cost	$ 69	$ 99	$ 99.99	$ 179

Source: Own elaboration.

Let us now look at the ability of hearing aids to reject external sounds. This ability can be divided into three categories open back, semi-open and closed back hearing aids. Basically, open back hearing aids offer a better translation of spatiality and natural sound at the expense of allowing external sounds to enter the ear. These headphones need a very quiet space, but they are the ones that best replicate the sound of loudspeakers in a studio. At the other extreme are the closed ones, which offer a much better isolation to external noises, with the disadvantage of not replicating the spatiality of the open ones and affecting the stereo sound image, making it narrower. The choice of hearing aid will depend on external factors such as the noise level of the room and, of course, the cost. **Table 2** suggests hearing aids tested by the author and shows their most relevant technical specifications.

Of the contenders shown in the table, we have had the pleasure of using and listening to them all. Personally, I own a pair of the AKGs and a pair of the Audiotecnica, and I can say that I prefer listening on the former, mostly because of their semi-open construction. However, for our hypothetical show I am leaning towards the Audiotecnica ATM M40x for the following reasons having an impedance closer to that of the receivers, theoretically, at least, the need

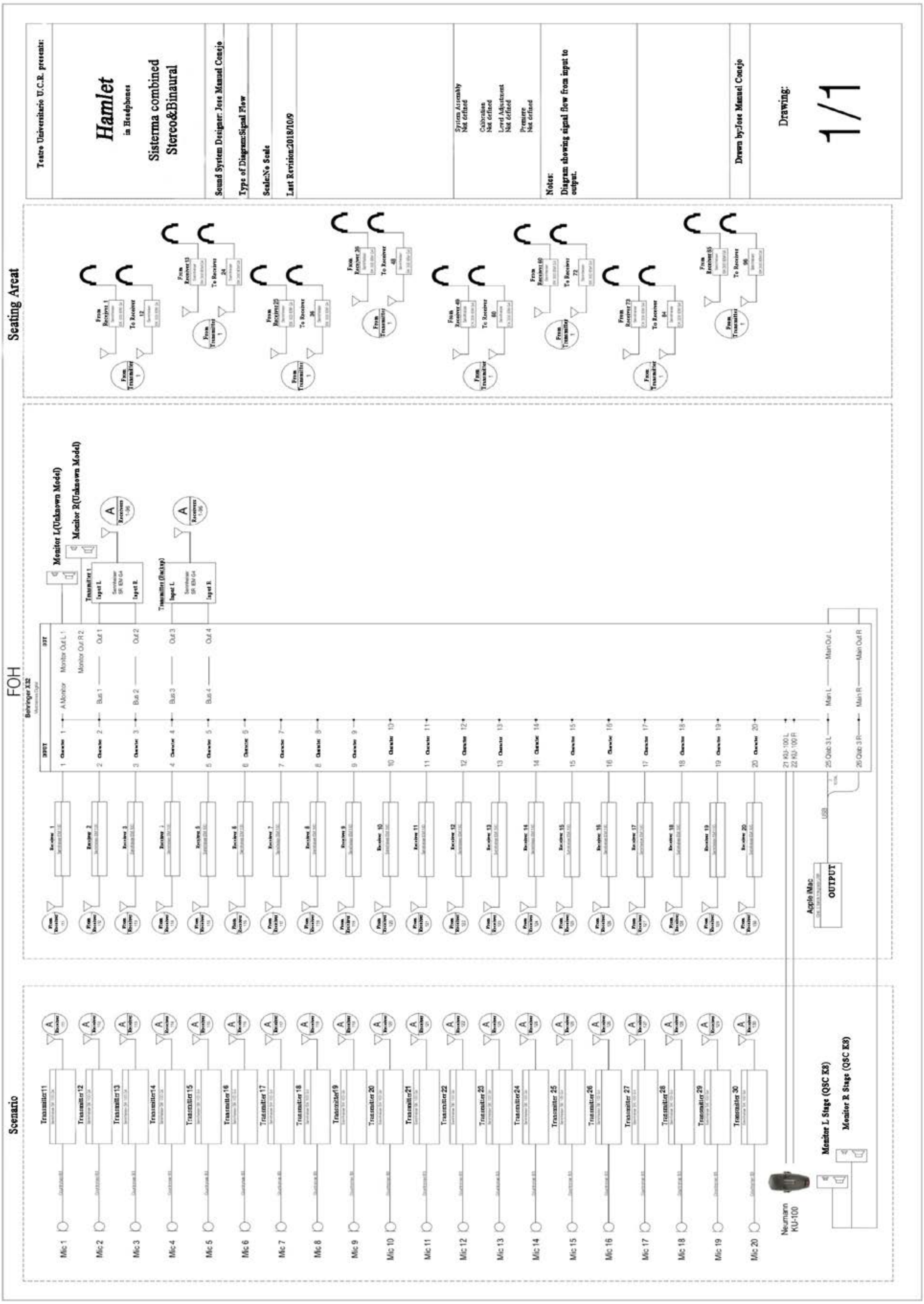

Figure 2. Summary of the design of wireless headphones for 96 viewers.

for volume on the IEM will be minimal. Being closed headphones, they offer more acoustic isolation and have a better bass response compared to the AKG K240 Studio. Its design is comfortable and comfortable to wear for a period of two to three hours. Its sensitivity is sufficient to handle the dynamic range of the show without distorting the sound. Its frequency response is flat enough for the mix engineer to detect undesirable sounds and correct before sending it to the audience. Although it is not the cheapest, it is still affordable and can be purchased in Costa Rica, in large volumes, at cost price. Finally, it is possible to replace the connection cable in case of deterioration and even has the option to connect shorter cables.

9. A lot of noise and how many nuts did you tell me it cost?

To exemplify the cost of making this system let's value what we need to buy assuming that you already have a console similar or equal to the Behringer X32 and a computer equal or similar to the Apple iMac with Qlab 3. **Table 3** shows the cost of making this show with the necessary wireless devices, with the brands of manufacturers Shure and Sennheiser, which are represented in our country. We obtain from these approximate costs that, if you want to mount Hamlet with a Sennheiser system in its entirety is necessary to invest $ 175,332 in 96 wireless headphones for the audience and 20 wireless microphones for the actors. If the show were to be captured binaurally, the total cost would be $ 86,586. If you want to do the show with Shure systems in its entirety, the investment amounts to $ 165,784 for 96 wireless headphones and 20 wireless microphones. If the show is binaural, the production must invest $ 82,642. It is important to mention that the actors must have monitoring of some kind to hear the sound feet on stage. To keep costs down in our hypothetical show, we will include two loudspeakers calibrated to a loud enough volume that allows the actors to hear while minimally filtering into the wireless microphones.

At the FOH, mixing can be done on speakers or headphones. Just to keep it simple, headphones are recommended, but fortunately, the X32 offers monitor outputs to connect a pair of speakers. If the show is entirely done in stereo, I recommend doing the mix on speakers.

Table 3. Estimated cost of wireless audio systems

Sennheiser wireless headphone system

Article	Brand	Model	Quantity	Cost Unit $	Total Cost $
Transmitter	Sennheiser	SR 300 IEMG4	2	749	1,498
Receiver	Sennheiser	EK 300 IEM G4	96	629	60,384
Headphones	Audiotecnica	M40x	96	99	9,504
Batteries	RadioShack	2,500 mA	240	30	7,200
Total wireless hearing aid system					78,586

Sennheiser wireless microphone system

Article	Brand	Model	Quantity	Cost Unit $	Total Cost $
Transmitter*	Sennheiser	SK 100 G4	20	599	11,980
Receiver*	Sennheiser	EM 100 G4	20		
Microphone**	Countryman	B3 for Sennheiser	20	219	4,380
Batteries	RadioShack	2,500 mA	60	30	1,800
Total wireless microphone system					18,160
Total investment in Sennheiser wireless systems in $					96,746

*Both items can be purchased as a package
**The package already includes a lavalier microphone, but this is not suitable for theatrical performances.

Shure wireless headphone system

Article	Brand	Model	Quantity	Cost Unit $	Total Cost $
Transmitter	Shure	P9T	2	649	1,298
Receiver	Shure	P9RA	96	590	56,640
Headphones	Audiotecnica	M40x	96	99	9,504
Batteries	RadioShack	2,500 mA	240	30	7,200
Total wireless hearing aid system					74,642

Table 3. (*Continue*)

Shure wireless microphone system					
Article	**Brand**	**Model**	**Quantity**	**Cost Unit $**	**Total Cost $**
Transmitter*	Shure	BLX1	20	549	10,980
Receiver*	Shure	BLX4R	20		
Microphone**	Countryman	B3 for Shure	20	186	3,720
Batteries	RadioShack	2,500 mA	60	30	1,800
Total wireless microphone system					16,500
Total investment of Shure wireless systems in $					91,142

*Both items can be purchased as a package.
**The package already includes a lavalier microphone, but this is not suitable for theatrical performances.
Source: Own elaboration

It is well accepted that the speaker mix translates very well into headphones, especially the sense of depth and spatiality. In any case, it does not hurt if the mix is done by monitoring the material in both headphones and loudspeakers. If, on the other hand, the show was entirely binaural or had sections in binaural, the mix should be done in headphones because this type of capture does not sound or generate the same sense of spatiality in the loudspeakers. Note, however, that the manufacturer Neumann states that its KU-100 microphone “carries over well on loudspeakers”[8]. It is up to the designer to evaluate with his own ears whether such a statement is true.

Finally, the sound designer must make sure that the transmitters and receivers are configured to receive pilot tone. This tone is located at 19 kHz. It serves, mostly, for the receiver to decode the stereo signal, otherwise, the receiver will modulate a mono signal or degrade the stereo signal. Sigismondi recommends that mixes should not have hard panning or that sound sources should not be arranged entirely to the left or right to ensure better transmission of the sound image[7]. This is especially important if there is sound material and pre-recorded music.

10. Possible implications of this design in the show

Now that we have assembled the parts, let’s remember for a moment how the signal flows in our show:

> Theatrical room → actors/microphones → computer sound source playback → mixing desk → wireless transmitters → wireless receivers → headphones → audience → spectator

Knowing that the hall of the University Theater has difficulty rejecting intrusive sounds from outside, it would seem that the use of binaural microphones is not an option because this microphone would pick up sounds from outside as well as those produced inside the theater. However, the beauty of doing a show based on headphones is that the acoustic isolation they create allows you to put sounds in the head of the viewer, without there necessarily being a real source that produces them. That is to say, a character like King Hamlet’s Ghost can be brought to the forefront using a binaural microphone, which allows recreating this sensation of forward-backward/updownward sound localization that is not possible to achieve in stereo. Then, this phantasmagoric aspect can be enhanced by making the voice of the spectrum rotate spherically in the listener’s head, without the need for a visual referent. Whereas, the voice of the actor playing Hamlet is captured by his wireless microphone and remains centered in the stereo image. In other words, it is possible to combine both types of pickup to generate a single multidimensional mix that generates an effect on the listener that is not possible to replicate with this level of precision in conventional loudspeakers.

The binaural microphone can also be used to locate sound events spatially, again without the need for a visual reference on stage. For example, a character entering from the lower right corner of the stage through a door. The mere convention of hearing a door opening immediately makes us as spectators think that someone will enter through it. With the help of a binaural microphone and headphones, it is also possible to pinpoint from where that door sound is produced, inviting the viewer to look towards the place where the source is produced, even if that sound was never produced on stage. To achieve these effects, it is necessary to remember that binaural sound replicates human hearing, therefore, the location of the sounds in binaural audio must be done in correspondence to where the audience is looking. Rather, the left, right, above and below the viewer also correspond to that of the binaural microphone.

The stereo sound, although conventional, will allow to keep in mind the voices of the actors who now benefit from interpreting their lines in a more intimate and natural way, cinematographic if you will, since the need to project the voice to the last seat is no longer necessary. The music benefits from stereo sound since it is not referential. In other words, it is not so important to know from where or who the sound of the piano is produced, but what it produces emotionally. Sound effects can be pre-produced in both stereo and binaural formats.

Finally, the average volume of the show should be decided in advance to ensure a uniform sound for the duration of the event, as well as to preserve its dynamic range. By dynamic range we mean that the sound will rise and fall in volume according to the events and happenings presented in the play. In the quiet passages it is to be expected that the volume will be low to medium, while at the climax of the work the volume will increase almost to its maximum limit. Determining this volume and its dynamic range is, in itself, a widely researched object of study, which we will not address in detail here, but we will mention some elementary aspects.

As the Fletcher-Munson studies demonstrated almost a century ago, human hearing is not linear and its perception of frequencies varies according to the volume at which they are reproduced, the band between 500 and 4,000 Hz being the most stable. Coincidentally, the human voice tends to be located mostly in this frequency range, so that, anatomically speaking, the human ear is designed to be sensitive to the human voice and, in rebound, to these frequencies. Frequencies outside this sensitive human range are much more affected by the intensity, or volume at which they sound. When Fletcher and Munson did their experiments, they noticed that, at very low volumes, the human ear is inefficient at hearing the low and high frequencies compared to the mid frequencies, even though the volume of all frequencies was the same.

As the intensity or volume of all frequencies was increased, the scientists began to notice that the subjects listened better to the low and high frequencies until they reached a point where it can be said that most of the subjects listened to low, medium and high frequency bands at the same volume, when, in fact, all these bands were arranged at the same level. These measurements showed that humans stabilize their perception of frequencies at around 80 dB SPL, even though lower intensities are those that experience a clipping of the low and high bands, and higher intensities are those that experience an increase even though the increase in volume is even for all frequencies. As a conclusion, the scientists found that hearing is subjective and that, in order to achieve a stabilization of the whole spectrum, a specific volume had to be reached. This level was then taken as a reference to measure how long a person can endure continuous sound without becoming fatigued or experiencing pain and, consequently, deafness. It was determined that 85 dB SPL is a bearable level for up to 8 continuous hours and that exposures greater than 85 dB SPL are only possible within an hour or a few minutes.

So how does this information affect the performance of our design? Both the sound designer and the mixing engineer will be responsible for sending an exact copy of the sound they produce at the FOH to all the headphones in the audience. In order to determine the correct volume balance between the actors' dialogue, music, effects and sound environments, it is necessary to calibrate all headphones to the same level, which will be 80 dB SPL. This level refers to the continuous volume and not to the momentary one, better known as "peak", since the ear needs a certain margin of time to determine a sound as "low" or "high". To determine this volume in the headphones, the receivers must receive a constant signal from the mixing desk (through the transmitter) at a level of -18 dB Full Scale (FS), since we will be working with digital devices.

Contrary to sound or mechanical acoustic waves, which start at 0 dB SPL or threshold of hearing up to 144 dB SPL or threshold of pain, digital audio works on a scale where the lowest audible representation of loudness is represented by a -∞, while its maximum is 0 dB FS. Since we will be passing sound from one medium to another[13] it is necessary to ensure that its level remains constant across them. Thus, the sound constant on the mixing console should read –18 dB FS. This level, in turn, should read the same on both the wireless transmitter and receiver. Finally, the volume knob on the receiver should be turned until the sound emitted in the headphones reaches 80 dB SPL. To avoid confusion, we should remember that dB SPL measures the loudness of sound waves in the air. While dB FS indicates the volume levels in a digital medium.

This practice allows us to have a dynamic ceiling. By placing the volume average at –18 dB FS on the table, the headphones at 80 dB SPL, leaving us a margin or ceiling of 18 dB on both sides, to increase the volume without reaching distortion. This means that, if the dialogues of the actors enter the table at -18 dB FS and the headphones sound at 80 dB SPL, when we add music and other sound elements, the added volume levels could reach up to –8 dB FS, which translates into 90 dB SPL in the headphones. We would still have 8 dB ceiling, to increase the volume. These volume increases we can reserve for short moments in our show, such as explosions or the grand finale of the play where there will be music, dialogue and sound effects. Thus, the highest possible volume in our show is 98 dB SPL (80 dB+18 dB dynamic ceiling), the average of 80 dB SPL and, of course, lower levels than this, with the advantage that they will be perceived by the audience without any problem.

11. Conclusions

It is evident that it is possible to create a theatrical show in which sound is transmitted wirelessly to headphones for more than 96 spectators in Costa Rica. But it is also evident that the investment for such a show requires a considerable amount of money. Although the document uses 96 spectators to establish its design, it does not suggest that this implementation is not possible for smaller capacity theaters, such as the Sala Vargas Calvo, while still having the same benefits or results. Hence the value of presenting a design proposal and how the devices communicate with each other.

The wireless transmission means a simple and cable-free installation inside the building, which facilitates the free transit of the spectators, in case such a need arises as part of the proposal. It is also stated that the possibility of exploiting the advantages of acoustic isolation offered by headphones combined with the capture and reproduction of sounds, in binaural format mixed together with the traditional stereo sound, expand the sound possibilities of artistic creation that are not possible to replicate in systems based on loudspeakers, such as those in our Costa Rican theaters.

As mentioned earlier, the use of microphones allows the actors to deliver a performance that is more focused on the interpretation of the text and not on the projection of the voice. This advantage is further evidenced by listening to these speeches on headphones which, in turn, allow the pickup volume of the microphones to be raised by eliminating feedback, or feedback, which is difficult to control when using loudspeakers. Another important aspect that this paper concludes is that it does not restrict the mix format to be delivered to the audience, but rather invites a combination of both (binaural or stereo). The safe zone is to make a show based on traditional stereo and, if possible, add a binaural microphone, either to pre-record sounds and then reproduce them in

[13]The means are: Acoustic with headphones, electronic by converting acoustic waves to electrical impulses, digital because the table converts these sources into digital audio, electronic by IEM transmitters and receivers, and acoustic again with headphones.

the show, or use it as a generator of three-dimensional sound effects live.

Finally, it is the designer's task to determine the average volume of the show so that it is delivered through the hearing aids in a pleasant manner. Although IEM receivers offer the option of adjusting the volume of each receiver, determining the average volume not only ensures consistency between the broadcast and reception of the audio material to be transmitted, but also helps the audience stay focused on the show and not get distracted by adjusting the volume of their receivers. This is either because the volume is too high and stuns or because it is so low that it loses detail of fundamental elements such as the acting dialogue.

Conflict of interest

The author declares no conflict of interest.

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