Available online at www.sciencedirect.com

SciVerse ScienceDirect

Procedia Engineering 35 (2012) 165 – 175

Procedia Engineering

www.elsevier.com/locate/procedia

International Meeting of Electrical Engineering Research
ENIINVIE-2012

Usage of IP technologies to reduce noise levels in audio and recording cabins

Moumtadi Fatima[a], La Rotta Santos Pedro Felipe[a],
Delgado Hernández Julio Carlos[b] a*

[a]*Electronics Department, Engineering Faculty, Universidad Nacional Autonoma de Mexico*
04510, Ciudad Universitaria, Coyoacan
Mexico City
[b]*Health Sciences Faculty, Universidad Anahuac Mexico Norte*
46, Universidad Anahuac Avenue, Lomas Anahuac
52786, Huixquilucan, Estado de Mexico,
Mexico

Abstract

Nowadays, audio channels in musical concerts (as well as in cabins as in live concerts) start from analogical audio signals generated at microphones, continue with signal transmission to a central audio mixer through wire connections (a wire for each microphone) and end at its digital processing and posterior reproduction through audio system speakers. Algorithm proposed by the authors suggests the signal digitalization directly at microphones and the introduction of virtual audio mixers, which simplifies audio channels management, and reproduces equalization and volume control functions from physical ones, adding control mechanisms over output channels. Noise and attenuation levels are measured and compared with traditional systems, to demonstrate the advantages of using the proposed algorithm.

Keywords: Sound, noise levels, IP technologies.

* Corresponding author. Tel.: 52 55 56223083
E-mail address: fatima@fi-b.unam.mx.

1877-7058 © 2012 Published by Elsevier Ltd.
doi:10.1016/j.proeng.2012.04.177

1. Introduction

Modern live recording studio modes (stereo live recording, mixed live recording and mixed multitrack recording [1]) have a common structure: microphones-collectors of audio signals are wire connected to a central mixer that collects and processes all signals (figure 1). Live recording is an easy to configure and manage recording mode, which characterizes itself as cheap and quick, too; however, high frequency and amplitude fluctuating audio signals (rock music, for example) require to move musicians and singers to balance the recording.

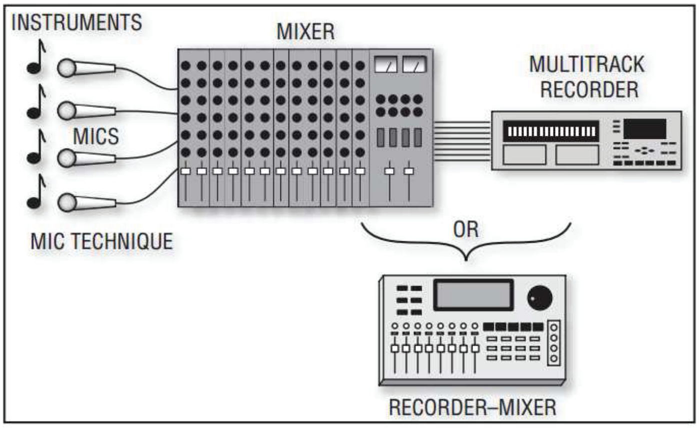


Figure 1. Connection scheme from microphones to audio mixer [1]

Audio mixers cover five functions, basically:

1. Input – preamplification and volume control over input signals;
2. Mixing – mix and balancing of more than two input signals
3. Quality control – sound characteristics managing;
4. Output – signal transmission to an output terminal
5. Monitoring – cue channel, a channel that allows heard audio signal before its recording or transmission [2].

Multitrack processing allows an independent management of input sound channels, mixing them on output terminals [3], a process that improves audio signal quality during recording and mixing through the use of several electronic devices as filters and amplifiers. Multitrack mixers recording capacities have been tested in recent years also by multimedia applications that require a central master to manage different tracks, functions that have been covered with complex software programs. However, as long as mixer covers more functions, more expensive and complex on its functioning and maintenance it becomes, remaining just for some large studios the capability to acquire it.

Actual commercial sound mixers (fixed, not mobiles) have a capability of 2 to 56 microphone inputs, from 2 to 24 line inputs, from 2 to 16 data buses and till 8 output channels [4], a bit rate of 96 000 bps and a pulse length of 24 bits. Some models include Ethernet control inputs and USB control and reproduction inputs, too.

2. Local Area Networks (LANs) and audio systems

LAN usage advantages in data transport have been widely documented. From its creation en 1980, Ethernet technologies have evolved from a wired, modular technology to a wide bandwidth one, with security, error and flow control mechanisms incorporated. Some modern Ethernet standards are complemented with mobility – wireless standards 802.11. In both cases, networks can be enlarged, adding a switch (or an access point in wireless variants) each 90 meters.

In order to calculate the bandwidth needed by the system to transport digitalized signal from the microphones to central mixer, some considerations have to be observed.

Each audio channel will be transmitted without any compression, sending 24 bits per pulse at a bit rate of 96000 bps, a total load generated of 2 304 000 bps per channel [5]. This scheme guarantees maximal audio quality.

Ethernet frames, in the other hand, have a load capability of 12000 data bits, and a header of 432 bits (when application managed is sent over a non-connection oriented protocol, as the real-time traffic [6]), which represents an increment of 3.6% to total load. To send audio signals without disruptions, 192 frames per second need to be sent, forming a total load of 2 386 994 bits, almost a quarter of total Ethernet channel capability. For microphone connections an Ethernet link can be used, but from switch to central mixer the link carries data from all individual channels. Technologies that can be added to the scheme are FastEthernet or Gigabit Ethernet. If the system includes 24 channels, load will reach 57.6 Mbps; if system is formed by 56 individual channels, then load to be carried reaches 134.4 Mbps. In both cases, FastEthernet or Gigabit Ethernet has the capability to transport generated data, even considering the 20% of extra bandwidth required by these technologies to carry it [7].

3. Microphones

Microphones must digitalize the information before sending it across the local network, capturing the signal generated by sources, sampling, quantifying and sending it towards the IP address of the computer that hostesses the virtual mixer application. Microphones must include 4 components: a network adapter, a microcontroller (that prepares the information prior to be transmitted and also manages a memory buffer) an analogical – digital converter and the microphone (figure 2).

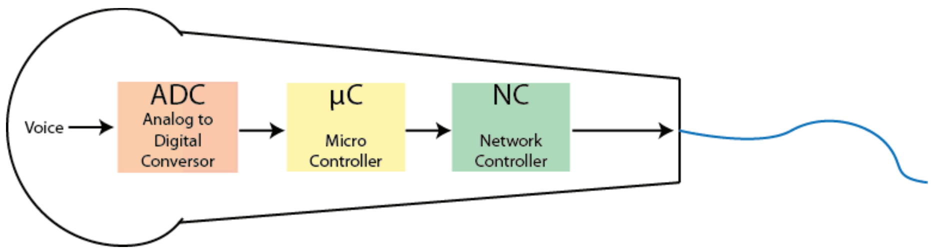


Figure 2. Microphone components

To realize the ADC conversion was picked an integrated circuit PCM1851APTJR from Texas Instruments, with a bit rate of 96000 bps and 24 bits per sample. The microcontroller, a MIPS32 M4K from MIPS products, covers a total area of 1.4 mm^2 and requires a maximal power consumption of 2.2 mW/MHz [8], characteristics that fully cover the microphone needs. Network controller used was a

RTL8019AS from Realtek, a full duplex network controller. All microphones are powered using Power over Ethernet standard (PoE), avoiding extra wiring to electrically supply them.

4. Virtual mixer

The software used to develop the virtual audio mixer has to have the capability to reproduce all analogical audio mixer components (volume control and equalization for input channels, centralized gain control, balance control that allows output channel selection for each signal, channel saturation indicators and on/off buttons for input channels), adding some improvements in output channels as volume control and signal level visual indicators, for example. Software has to be stable and be characterized for its facility of managing and its coupling with the operating system that hostess it. Programming language Objective C, developed specially for MAC OS X operating system from Apple computers covers all these requirements. This programming language is delivered together with the programming platform Xcode within the OS X operating system (figure 3).

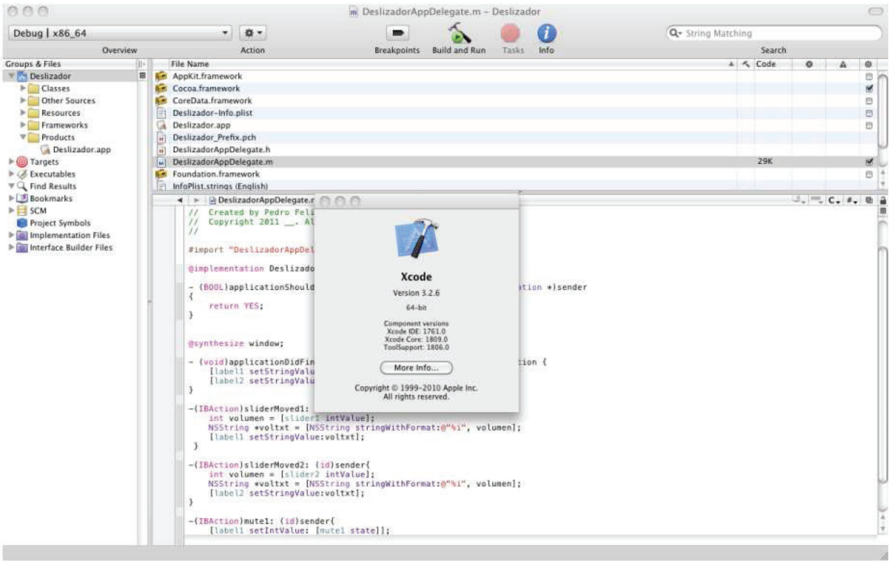


Figure 3. XCode programming interface

Programming code is divided into two parts. First one develops general audio mixer graphical interface: volume, equalization and balance controls (figure 4 shows a developed interface with 4 channels).

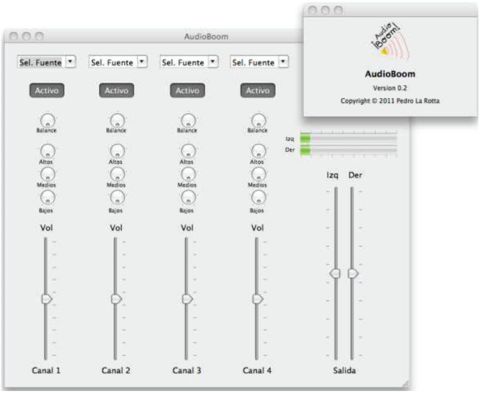


Figura 4. Virtual audio mixer interface

Second part of programming code is dedicated to digital signal processing, subdivided in 6 steps:

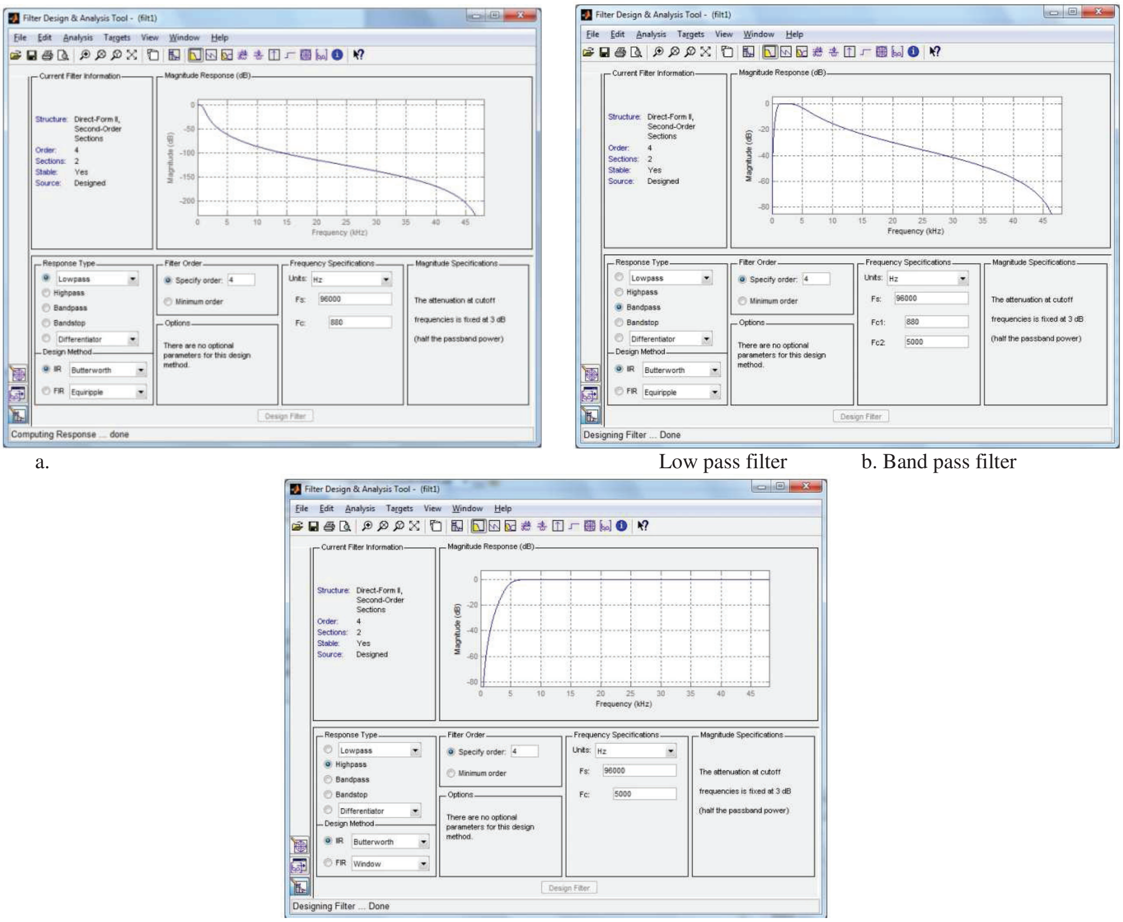
1. Transport of data from an input channel to general output mixer channel. The audio mixer must control the volume of signal sources (both microphones or tape reproduction units);
2. Amplification of input audio signals, in accordance with predefined by channel volume levels;
3. Signal processing using filters of equalization to improve audio quality;
4. Signal transmission (sending the signal through one channel or splitting it into two output channels, in dependence of mixer balance adjust);
5. Output signal amplification by system master volume level;
6. Signal forwarding to speakers.

System equalizers were programmed to have three filters per channel. Implemented infinite impulse response (IIR) filters were developed using FDATool, a tool embedded in MatLab from MathWorks. This tool calculates coefficient values for filters, merely indicating desired cut frequencies. Developers used the set of cut frequencies showed in table 1.

Table 1. Cut frequencies for IIR filters

Filter type	Cut frequencies
Low pass filter	880 Hz
Band pass filter	880 Hz – 5000 Hz
High pass filter	5000 Hz

All filters were implemented as 4^{th} grade, 2 levels Butterworth filters. These filters are characterized by the possibility they have to assure short codes and acceptable quality in cut frequencies values, where pass power has as much as half the value of transmission power. For digital recording, “0 dBFS” (0 decibels in full scale) is the maximal limit for signal without distortion; average noise level is generally at -20dBFS. Critical signal-to-noise (S/N) values for these filters are: 60 dB is average level, 70 dB is good level and 80 dB represents the best audio quality [1]. Filter design and response in frequency are showed in figure 5.



a. Low pass filter b. Band pass filter

c. High pass filter

Figure 5. Design and frequency response for low pass (a), band pass (b) and high pass (c) implemented filters.

Once transfer functions for all filters are defined by the software, general difference function in discrete time is defined (1) and programmed (figure 6).

$$Y(n) = A_{14} * (A_{11}\, x[n] + A_{12}\, x[n-1] + A_{13}\, x[n-2] + A_{15}\, y[n-1] + A_{16}\, y[n-2]) \quad (1)$$

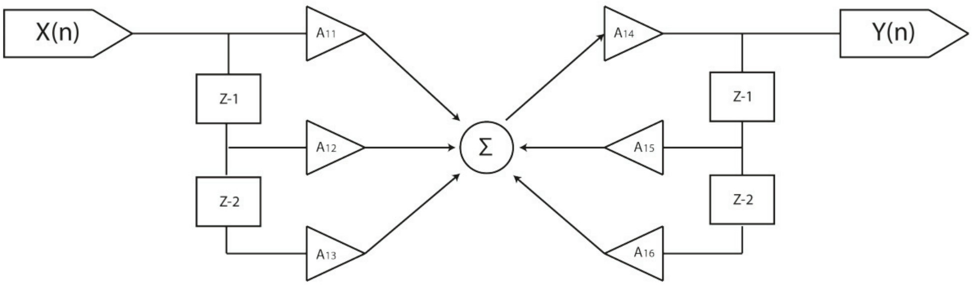


Figure 6. Equalizer design

Filter output signal is forwarded to the speakers through audio mixer output channel. General speakers electronic components diagram is showed in figure 7.

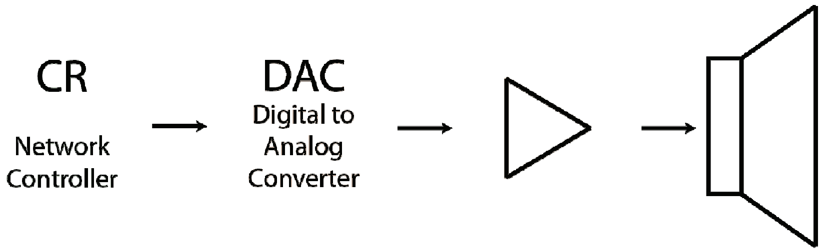


Figure 7. Speaker electronic components.

5. Signal losses

After it crosses the network controller and the DAC converter, signal is finally amplified using a low power consumption amplifier. Figure 8 shows an equivalent to final part in figure 7 diagram: a series circuit connection, where the speaker is located at a wire end, which acts itself, in the other hand, as a resistance.

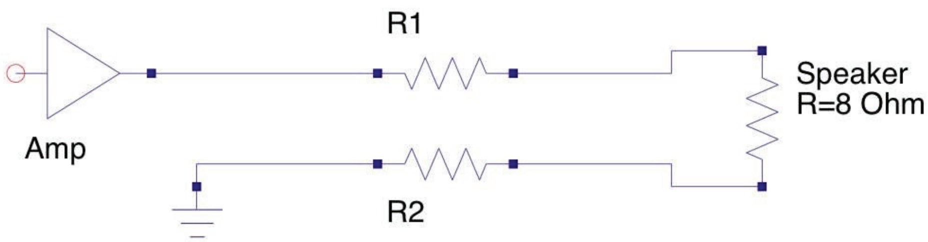


Figura 8. Wire – speaker series circuit connection

Value for resistances R1 and R2 depends directly of the length and wire diameter, but its values are equal ($R1 = R2$). Power, in the other side, takes its value from the product of the square of the current multiplied by the resistance, as can be seen in (2).

$$P = I^2 R \tag{2}$$

System power factor is defined as the ratio of speaker power between the total power consumed by the system, as described in (5):

$$P_{speaker} = I^2 R_{speaker} \tag{3}$$
$$P_{cable} = I^2 R_{cable} \tag{4}$$
$$\frac{I^2 R_{speaker}}{I^2 R_{speaker} + I^2 R_{cable}} = \frac{I^2 R_{speaker}}{I^2 (R_{speaker} + R_{cable})} = \frac{R_{speaker}}{(R_{speaker} + R_{cable})} \tag{5}$$

Total system losses, in the other hand, can be expressed as follows (6):

$$Losses = 20 \log \left(\frac{R_{speaker}}{R_{speaker} + R_{cable}} \right) \tag{6}$$

Resistance value induced by wires depends of the wire diameter, increasing (or decreasing) signal attenuation levels. Measurements realized with sound connections cables of Condumex [9], show the values of signal attenuation (for a maximum distance of 100 meters) in figure 9. The used cables were:

1. Cables Condumex CL2R for high potency speakers, AWG 14 diameter (code 65601635);
2. Cables Condumex CL2R for high potency speakers, AWG 16 diameter (code 65601835);
3. Cables Condumex with transparent PVC cover, AWG 18 diameter (code 720270);
4. Cables Condumex with transparent PVC cover, AWG 20 diameter (code 720824);
5. Cables Condumex with transparent PVC cover, AWG 22 diameter (code 720271).

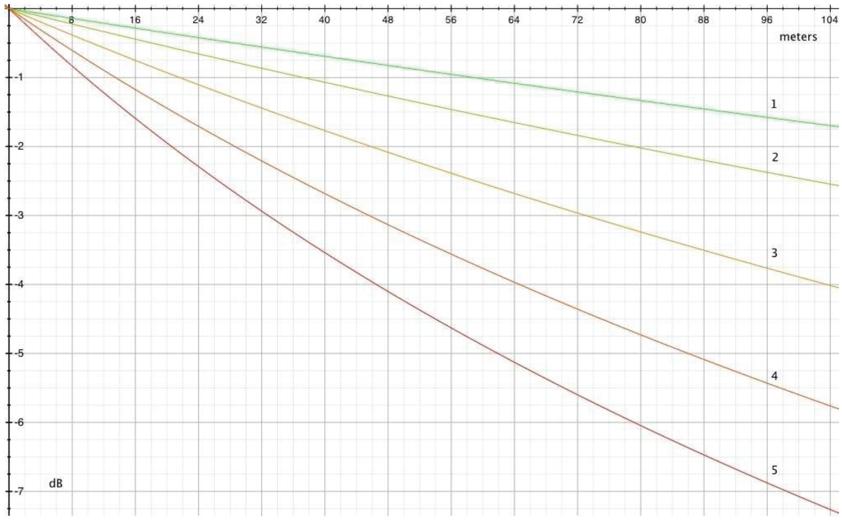


Figure 9. Signal losses levels for different audio cables

From figure 9 is easy to see that the usage of non-high potency cables reduces the length of critical transmission (with loss levels above 1 dB). For this reason the system design considered minimal distances between the amplifier and the speaker, reducing signal losses induced by cables. Digital signal crosses local area network till the DAC converter, located close to the amplifier (and to the speaker, too). This design, in addition, allows to place away the mixer from the speakers, giving major capability of coverage to the audio system without important signal losses.

Network cable has a 24 AWG diameter, and, according to IEEE standard 802.3 – 2008, load resistance in network cards has values in a range from 73Ω to 83Ω ± 1%, a power factor clearly higher than audio cables. Also, transmission voltage levels are lower than in audio signals (approximately 1315 mV at its peak value), that could originate less losses too [10]. In figure 10 are compared losses levels from three different cables:

1. UTP Cable Condumex, AWG 24 diameter (code 66445872) [11];
2. Cables Condumex CL2R for high potency speakers, AWG 14 diameter (code 65601635);
3. Cables Condumex with transparent PVC cover, AWG 18 diameter (code 720270).

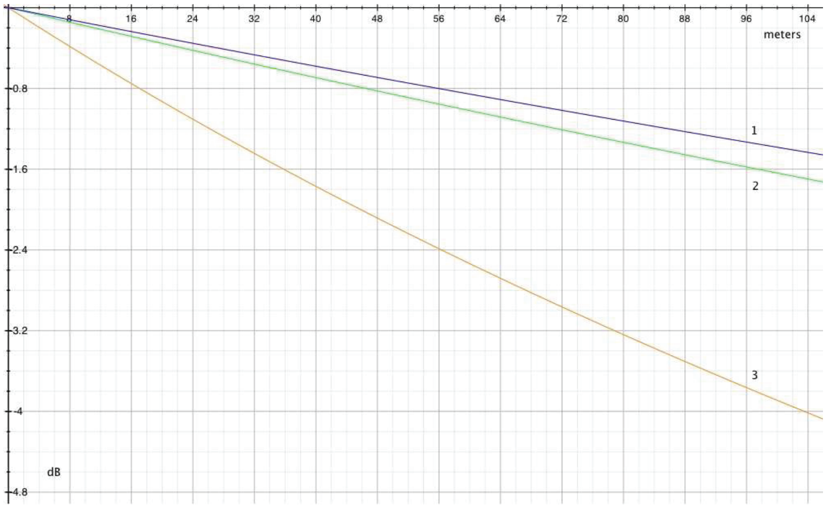


Figure 10. Comparison between audio and network cables attenuation.

Figure 10 shows that data transmission through local area networks, digitalizing sound signal directly at microphones and placing the amplifier as close as possible to the speaker minimizes cable induced losses, improving audio quality in output devices.

6. Noise levels

To calculate noise levels in the developed system is necessary to know higher voltage delivering level and the number of signal quantification levels too. As long as electrical supply to microphones will be realized using PoE standard (802.3af), a nominal supply of 48 volts is delivered [12]; for quantification, ADC converters implemented in the solution use 24 bits per pulse.

Output digitalized signal can be described as the real signal value plus / minus quantification noise (originated after rounding values to legal quantification signal levels) as is showed in (7):

$$x_q = x + e_q, \text{ where } e_q = x_q - x \tag{7}$$

Maximal quantification error level is $q/2$.

Signal to noise ratio induced by quantification could be calculated using relation (8):

$$S/N_q = \frac{\text{Signal power}}{\text{Quantification noise power}} \tag{8}$$

Signal power, with an amplitude level described as A, a peak value of $\frac{mq}{2}$ and an enegy level of $\frac{A^2T}{2}$ is calculated using relation (9):

$$P = 2^{2(m-1)} \times q^2 \tag{9}$$

Noise power, in the other hand, is calculated through its expected value $E\{e_q\}$ and its variance ($\tau^2\ e_q$). As long as e_q is a continuous random variable uniformly distributed ($-\frac{q}{2} \leq e_q \leq \frac{q}{2}$), can be concluded that its variance reaches its average power. Then, variance can be expressed as $\frac{q^2}{12}$, and S/N ratio, in a corresponding way, is defined as follows (10):

$$S/N_q = 10log\left[\frac{2^{2(m-1)} \cdot q^2}{\frac{q^2}{12}}\right] = 10log\left[3 \cdot 2^{2m}\right] = 10log3 + 20m\ log2$$

$$S/N_q = 4.7712 + 6.02m\ [dB] \tag{10}$$

And being $m = 24$ (number of bits per sample) S/N_q ratio for system is described in (11):

$$S/N_q = 4.7712 + (6.02 \cdot 24) = 149.2512\ [dB] \tag{11}$$

Commercial professional mixers actually work with a S/N ratio of 82 dB, but this value is not considering noise induced by wires from microphones to mixers and from mixers to speakers.

7. Conclusions

As was shown in the paper, the usage of LANs to transport audio signals in audio cabins, even when no compression algorithms are implemented, allows the improvement of sound quality, as long as line noises are changed for quantification noises, lower on their amplitude. Also was demonstrated that actual network capabilityty is enough to transport the audio stream.

Developed devices (as physical as logical) were tested versus actual commercial probed devices, and, as was showed through the paper, was demonstrated that the proposed solution bring improved results.

Several tests over the solution developed were done, and the most relevant and significant of them were showed.

References

[1] Bartlett B, Bartlett J. Practical recording techniques 5th ed. Editorial Focal press, USA, 2009.

[2] Zettl H. Manual de producción de televisión, 7a. ed. Ed Thomson, México, 2005.

[3] Hausman C, Benoit P, O'Donell L. Producción en la radio moderna 5ta ed. Ed Thomson, México, 2001.

[4] Catálogo TELETEC de audio consolas, fuentes y micrófonos. Available at: http://www.teletec.com.mx/images/catalogo/pdf/H-%20Audio%20Consolas,%20Fuentes%20y%20Microfonos.pdf (consulted in December, 2011)

[5] Bailey A. Network technology for digital audio. Ed. Focal Press, USA, 2001.

[6] Church S, Pizzi S. Audio over IP – building Pro AoIP systems with LimeWire. Ed. Focal Press, USA, 2010.

[7] Held G. Ethernet networks: design, implementation, organization and management, 4th ed. Ed. Wiley & sons Ltd, UK, 2003.
[8] http://www.mips.com/products/cores/32-64-bit-cores/mips32-4k/ (consulted in November, 2011).
[9] Fichas técnicas de cables de audio. Servicios CONDUMEX, S.A. de C.V. Mexico, 2011.
[10] IEEE standard 802.3 – 2008, part 3: Carrier sense multiple access with Collision Detection (CSMA/CD) Access Method and Physical Layer Specifications. IEEE, 2008.
[11] Fichas técnicas de cables para redes (LAN). Servicios CONDUMEX, S.A. de C.V. Mexico, 2011.
[12] 802.3af requirements. Available at http://www.ieee802.org/3/af/requirements.pdf (consulted in November 2011).