PRIMERAS JORNADAS REGIONALES DE ACÚSTICA

AdAA 2009

19 y 20 de Noviembre de 2009, Rosario, Argentina

PROGRAMA
CONFERENCIAS
ACTAS

Compiladores:

Federico Miyara
Ernesto Accolti
Fernando A. Marengo Rodriguez
Susana Cabanellas
Marta Yanitelli
Vivian Pasch
Pablo Miechi

- Tendiendo puentes entre la acústica y la sociedad -

Primeras Jornadas Regionales de Acústica

AdAA 2009

Programa

Conferencias

ACTAS

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Laboratorio de Acústica y Electroacústica

NOTA DE BIENVENIDA

Estimados colegas:

Hace unos pocos meses apenas, en la tradicional reunión mensual de Comisión Directiva de la Asociación de Acústicos Argentinos (AdAA), se propuso organizar unas Jornadas de Acústica similares a las que se venían realizando con cierta periodicidad bajo designaciones tales como Reuniones de Ruido Urbano o Congresos de Acústica del Nuevo Milenio. La novedad era que, a fin de dar mayor difusión a las actividades de la AdAA hacia el resto del país, se realizarían no en Buenos Aires sino en alguna ciudad representativa del interior. Se barajaron diversas posibilidades y finalmente el honor y la responsabilidad recayeron en la ciudad de Rosario.

Capitalizando la experiencia ganada en la organización del Congreso FIA 2008, un grupo entusiasta de investigadores sobre ruido de la Universidad Nacional de Rosario decidimos dar lo mejor de nosotros para ofrecer unas Jornadas de gran nivel organizativo. Así, hemos incorporado la revisión por pares de los trabajos presentados, lo cual garantiza la calidad y solidez técnica y científica de las ponencias. Hemos convocado a especialistas nacionales y extranjeros para dictar conferencias distinguidas y para integrar una mesa redonda sobre Ruido Urbano. Hemos organizado también una videoconferencia en directo desde Estados Unidos a cargo de Leo L. Beranek, uno de los más célebres acústicos del siglo XX y del actual.

Quisimos darle un enfoque fuertemente social a este encuentro, y por ello hemos adoptado el lema "Tendiendo puentes entre la acústica y la sociedad", que se ve reflejado en la imagen que acompaña a toda la gráfica. Es por ello, también, que hemos invitado a un encantador grupo de niños músicos, integrantes de la Escuela Orquesta de Barrio Ludueña, a deleitarnos en un emotivo agasajo musical. Es un hermoso proyecto de integración social que sirve de ejemplo cuando las fuerzas flaquean.

Les damos, entonces, nuestra más cordial bienvenida y esperamos que disfruten plenamente de estas Jornadas.

Federico Miyara
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imfia
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Universidad Nacional de Tres de Febrero

PROGRAMA

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<tr><td>11:00
12:15</td><td colspan="2">Videoconferencia
Leo Beranek
Some important contributions to Acoustics</td></tr>
<tr><td colspan="3"></td></tr>
<tr><td>12:30
13:00</td><td colspan="2">Conferencia
Samir Gerges
European Aircraft Noise Research Network</td></tr>
<tr><td>13:00
15:00</td><td colspan="2">Tiempo libre para almuerzo</td></tr>
<tr><td></td><td>Sesión A1 - Acústica de salas</td><td>Sesión B1 - Ruido I</td></tr>
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<tr><td>16:30
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<tr><td>17:00
17:30</td><td colspan="2">Conferencia
Samir Gerges
Protectores auditivos: Atenuación y confort</td></tr>
<tr><td colspan="3"></td></tr>
<tr><td></td><td>Sesión A2 - Acústica musical</td><td>Sesión B2 - Acústica Arquitectónica</td></tr>
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<tr><td colspan="3"></td></tr>
<tr><td>19:15
20:00</td><td colspan="2">Agasajo Musical
Escuela Orquesta de Barrio Ludueña</td></tr>
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	Sesión A3 - Ruido II	Sesión B3 - Audio
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10:30 11:00	Café	
11:00 11:30	Conferencia Jorge P. Arenas Control de ruido mediante barreras acústicas	
	Sesión A4 - Temas misceláneos	Sesión B4 - Acústica Computacional
11:40	A-002	A-032
12:00	A-030	A-051
12:20	A-023	A-057
12:40	A-061	A-035
13:00 14:30	Tiempo libre para almuerzo	
	Sesión A5 - Salud y protección auditiva	Sesión B5 - Psicoacústica
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16:30 17:00	Café	
17:00 18:30	Mesa redonda Ruido urbano	Conferencia Gabriel González Ferreira Música y Software libre
18:45 19:15	Acto de cierre	
19:30 21:00	Cocktail de despedida	

CONFERENCIAS PLENARIAS

C01
Some important contributions to acoustics

Leo L. Beranek
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beranekleo@ieee.org

Abstract: This lecture reviews the principal contributions that I have made to the acoustics of noise reduction, the design of small loudspeakers, and the acoustics of concert halls and opera houses. In 1949 I discovered how to line a duct with an acoustical material in such a way that high attenuations of sound down the duct would result. I used this discovery to reduce the noise of an enormous jet engine test facility which resulted in the world's largest muffler. This same technique was used to reduce the exhaust noise of large aircraft reciprocating engines on the last propeller-driven airplanes before the jet age. It has led to the packaged mufflers used today in air-conditioning systems for buildings. In my classes at MIT I developed a new method of analyzing the behavior of loudspeakers which led to the small library-shelf-sized loudspeakers that are commonly found everywhere today. Finally, over an extended period of time I developed an understanding of the acoustical characteristics of concert halls and opera houses. I advised architects on the design of several of the world's celebrated concert halls and opera houses. The talk will show how these same techniques and findings are still important to acousticians.

1 Cleveland-NASA Supersonic Wind Tunnel

It was in 1950 that I received a telephone call from the National Aeronautics and Space Administration's Lewis Flight Propulsion Laboratory in Cleveland, Ohio (NASA Cleveland). The Director was frantic. At about midnight, two days earlier, the laboratory had put into operation a new jet engine in a supersonic wind tunnel. The noise produced was so intense, like a series of explosions, which could be heard as far as 16 km away. I was told that the noise was so serious, the City of Cleveland refused to let them operate the facility again, except for tests on how to eliminate the noise. Members of my company, Bolt Beranek and Newman, Inc., and I made measurements and determined that the noise had to be reduced about 30 dB between 5 Hz and 300 Hz and about 20 dB at frequencies above. BBN was asked to solve the problem. How could so many decibels at these frequencies be accomplished? I immediately thought of an experiment that I had run the year before.

It was at the Massachusetts Institute of Technology where I was a Professor. I was setting up a means for calibrating microphones by comparing them with a laboratory standard. I built a 0.3 m square duct, about 3 m long with an anechoic wedge at one end and a loudspeaker at the other end. I lined this duct with a 2.5 cm thick porous acoustical material. The method was to first put the standard microphone in the duct about one meter away from the loudspeaker and read its output and then the put other at exactly the same position and read its output.

First, I measured the strength of the sound wave produced by the loudspeaker at various positions along the center of the duct. Imagine my surprise when I discovered that at about 1500 Hz the attenuation of the sound wave traveling along the duct was about 25 dB/m. This was more than twice as much as the expected attenuation of about 10 dB/m. What was the explanation?

Some years before I had studied a paper by Philip Morse (JASA Vol. 11, pp 205-210) that gave a new way of calculating the attenuation of sound traveling in a duct, using the normal acoustical impedance of the material lining the duct as the starting point. He assumed that the acoustical material would be on only one side of the duct. In 1940 (JASA Vol. 12, pp 228-231), I expanded the Morse theory to deal with a duct lined on four sides. I shall call this the Morse/Beranek theory.

This theory required the acoustic impedance of materials and I knew how to determine this quantity (Published later, JASA Vol.13, pp 248 to 260). Let me tell you how I applied this information to quieting the terrible noise source at NASA.

As you know, the normal acoustical impedance Z at the surface of a material is equal to the ratio of the sound pressure p divided by the velocity u of the air particles at that surface i.e., $Z = p/u$.

I could calculate Z by using electrical circuit theory where p is analogous to voltage e and u is analogous to current i. (See Fig. 1)

The sizes of the elements can be determined from a knowledge of the flow resistance of the material R_1, the density of the air in the material ρ (about 1.18 kg/m^3), the speed of sound in the air inside the material c_1 (about 300 m/s), the porosity of the material P (about 0.9), the atmospheric pressure p_0 (10^5), and the dimensions d and ℓ. With no airspace, $L = \rho d/3$, $C = 0.9d/(\rho c_1^2)$ and $R = R_1 d/3$. With airspace, the added elements are $R_2 = 0.28R_1\ell$, $L_2 = 0.28\ell$, $C_2 = \ell/10^5$.

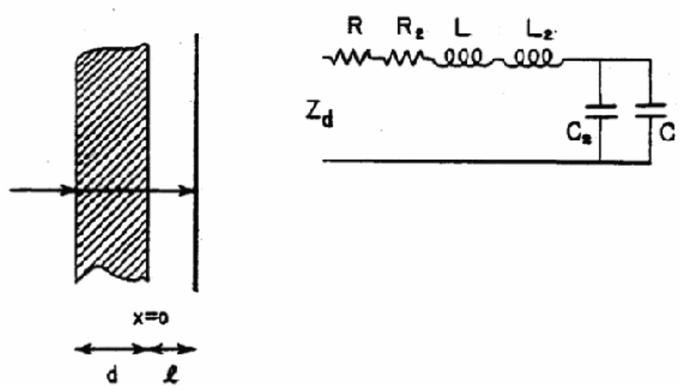


Figure 1. Electrical circuit equivalent for the structure shown on the left. The acoustic impedance at the surface of the material is $Z = p/u$ where p is the sound pressure measured at the surface and u is the particle velocity measured at the surface.

The acoustic impedance $Z/\rho c = R/\rho c + jX/\rho c$, calculated for a material 2.65 cm thick with and without an airspace 3.3 cm deep behind is given in Fig. 2 (rigid wall backing).

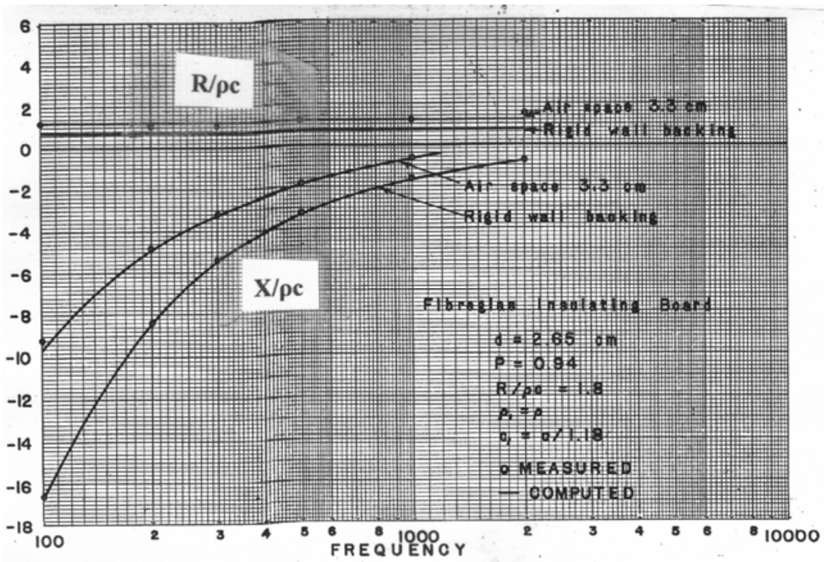
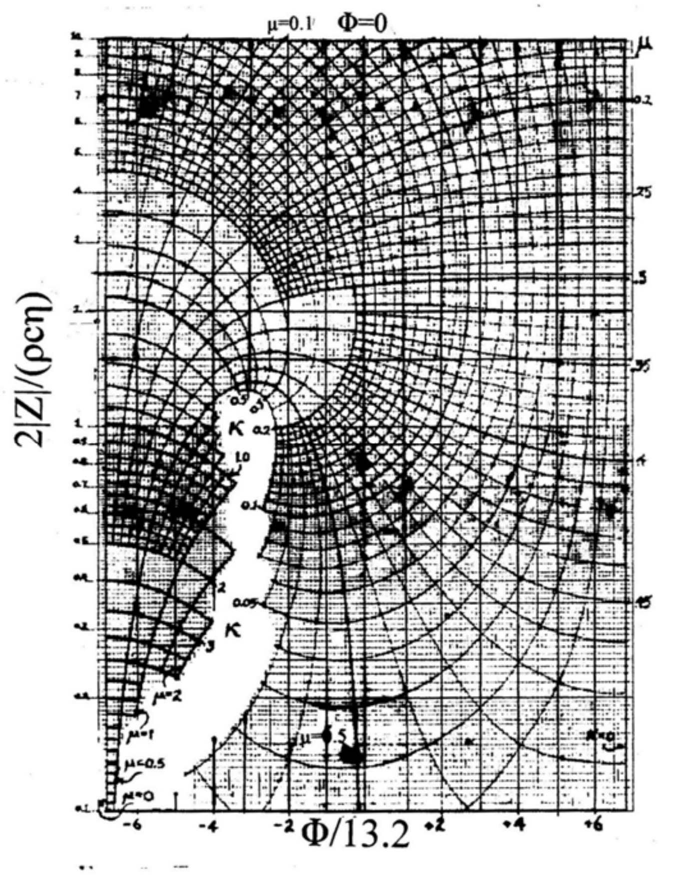


Figure 2. Normal acoustic impedance of a typical material for lining a duct. No airspace and an airspace of 3.3 cm are shown. The method of calculating impedances is given in a paper by me (JASA Vol.13, pp 248 to 260).

The next step was to calculate the attenuation of the sound wave as it traveled down the duct. First, we must calculate three quantities: $|Z|/\rho c = \sqrt{(R/\rho c)^2 + (X/\rho c)^2}$, $\Phi = \tan^{-1}X/R$ (that is, $\tan\Phi = X/R$), and $\eta = L/(\lambda/2)$. The quantity λ is the wavelength of the sound and equals the speed of sound in air divided by the frequency of the sound wave, i.e., c/f.

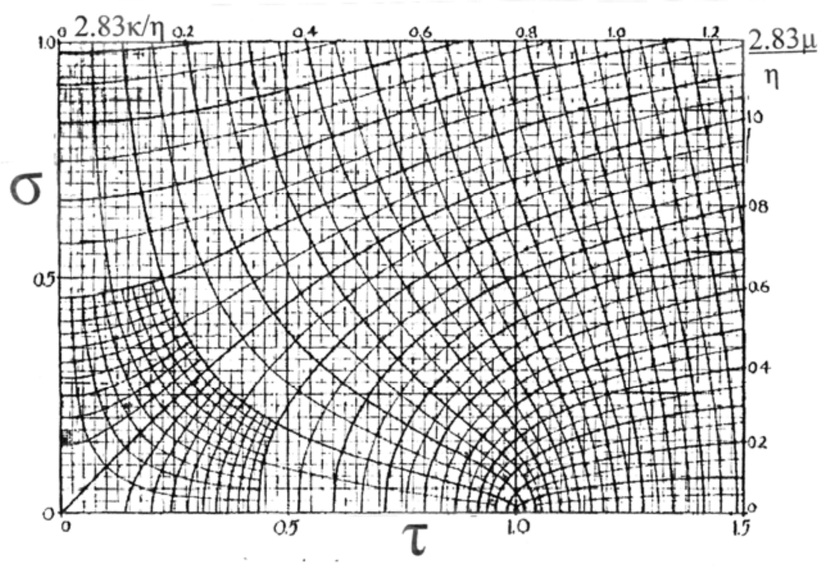
Morse's first chart, corrected by Beranek to 4 sides of duct treated, uses the two quantities $2|Z|/(\rho c\eta)$ and $\Phi/13.2$ to get two quantities κ and μ.



Morse's Chart 1.For converting the *Z* of the material into two intermediate quantities, κ and μ.

For example, at 1500 Hz a duct with L = 0.3 m wide, c = 344 m/s, $|Z|/\rho c = 3.68$, and $\Phi = -25.1^{o}$. We have $\eta = 2.62$. So, from the chart, for $2|Z|/\rho c\eta = 2.66$ and $\Phi/13.2 = -1.27$, we get $\mu = 0.270$ and $\kappa = 0.245$.

Morse's Chart 2 gives σ if μ and κ are known. The quantity σ determines the attenuation of the wave as it travels down the tube. (c/τ) is the phase velocity of the wave, where c is the velocity in free space. The value of τ is usually close to 1.0. For μ = 0.270 and κ = 0.245, we



Morse's Chart 2. This chart gives σ and τ when μ and κ are known .

have, $2.83\mu/\eta = 0.292$ and $2.83\kappa/\eta = 0.265$. From the chart we get σ approximately equal to 0.10. This must be converted to dB/m.

The attenuation of the sound wave as it travels down the duct is given by:

$$A_{dB/m} = 54.6\ (\sigma f/c).$$

In our example at 1500 Hz, the $A_{dB/m} = 54.6\ (0.1\times1500/344) = 24$ dB/m. This explains the large attenuation that I obtained in the MIT calibration duct.

Other useful information:

For $2|Z|/\rho c\eta$ greater than 10, use

$$\mu = \cos(45^o - \Phi/2)\times[\eta/(\pi|Z|/\rho c)]^{1/2}$$
$$\kappa = \sin(45^o - \Phi/2)\times[\eta/(\pi|Z|/\rho c)]^{1/2}$$

For σ smaller than 10, use

$$\sigma = (2.83\mu/\eta)\times(2.83\kappa/\eta)/\tau$$
$$\tau = [1 + (2.83\kappa/\eta)^2 - (2.83\mu/\eta)^2]^{1/2}$$

To complete the picture, I have calculated the dB/m for a 0.3 m square duct lined with a material that has the $Z/\rho c$ as a function of frequency from 100 Hz to 3000 Hz as is given in Fig. 2 (assuming no airspace behind). The results are plotted in Fig. 3. The amazing value of 24 dB/m at 1500 Hz which I had measured at MIT is also calculated.

This finding was applied to the Cleveland-NASA muffler.

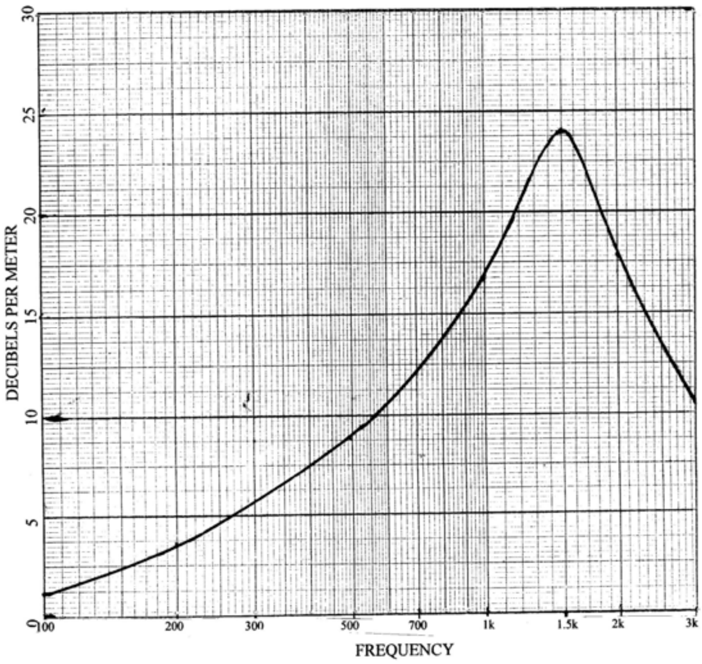


Figure 3: Attenuation in a 0.3 m square duct lined on four sides with 2.5 cm thick material having acoustic impedance shown in Fig. 2. This curve is calculated by the Morse/Beranek procedure and the measured data were almost exactly the same.

1.1 Designing the Cleveland-NASA Muffler

The Cleveland-NASA muffler as built is shown in Fig. 4. The dimensions of the wind tunnel (which is not shown at the lower left) was 2.44 m times 1.83 m, which equals 4.46 m^2. This was also the area of the small end of the cylindrical tube that is named “Trumpet Muffler”. At the large end its area was 54 m^2. This tube was already in place and the C-scale noise measured at 25 ft (7.62 m) from the opening was about 125 dB. NASA insisted that any attachment had to result in negligible back pressure. This meant than any added muffler had to have an open area of 54 m^2. We measured the background noise in the neighborhoods and this showed us that the noise had to be reduced about 30 dB between 5 Hz and 300 Hz and about 20 dB at frequencies above if complaints were to be eliminated.

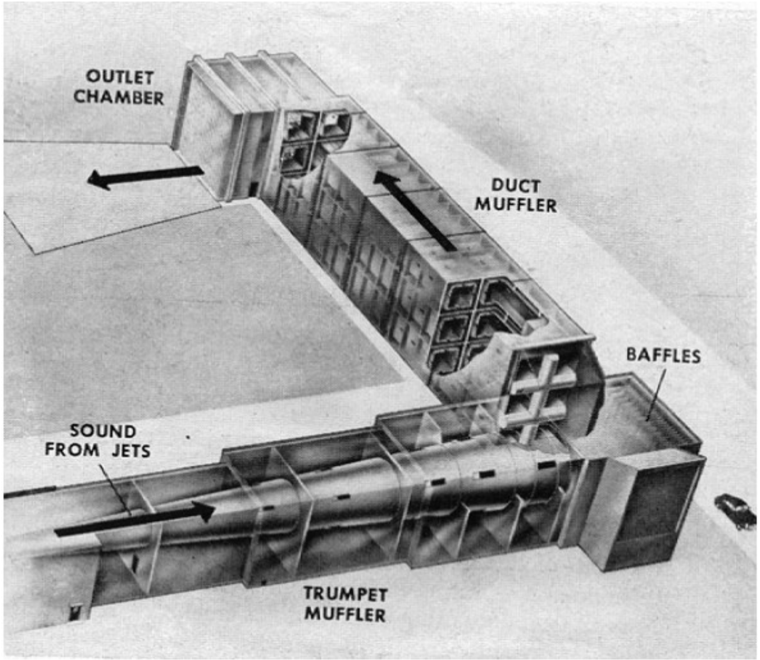


Figure 4: Design of the completed Cleveland-NASA muffler, showing three principal components, (1) Trumpet Muffler, (2) Duct Muffler, and (3) Outlet Chamber.

The trumpet muffler was enclosed in a concrete shell about 30 cm thick. The noise between 5 Hz and 20 Hz was reduced using Helmholtz resonators (rectangular holes that led into surrounding airspaces that were separated from each other by the subdividers shown). The average attenuation measured on completion was about 25 dB in this frequency region. An additional 5 dB was achieved by the first 90^o bend.

Now, I wish to explain the design of the "Duct Muffler". Go to Fig. 5.

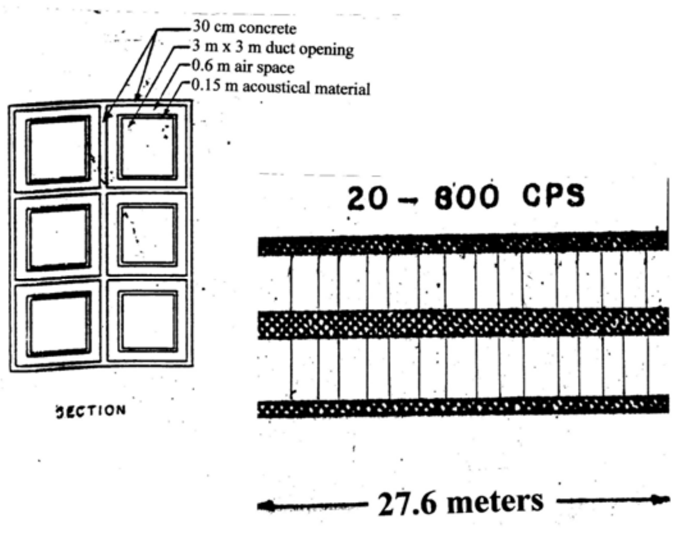


Figure 5: The duct muffler with a section 10 m × 15 m and a length 27.7 m. The thin lines indicate that the 0.6 m airspace had to be subdivided by thin partitions to prevent the sound from traveling along the duct in that airspace.

The Morse/Beranek theory was used. By trial and error it was found that the 54 m^2 area should be divided into six ducts as shown in Fig. 5.It was assumed in the calculations that the maximum attenuation should occur at 100 Hz and that a 1.6 dB/m attenuation should be achieved between about 400 Hz and 2000 Hz .

The next question was whether to fill the 0.75 m space in each duct with acoustical material or to have a layer of material with an open space behind. Because the material must not erode from the high speed air flow, a perforated metal covering was necessary. I learned that commercial panels having a thickness of 0.15 m, with a perforated metal surface on each side were readily available. They were chosen and a Fiberglas material was chosen to go inside.

In the 0.6 m space that would be behind the steel panels, air waves would be free to travel and with them it would be impossible to obtain a meaningful normal acoustical impedance Z/ρc. To reduce this travel, blocking partitions were introduced about every 6 m. The charts showed that the acoustical lining should have a flow resistance of about 11.5 rayl/cm and this could be achieved with a Fiberglas blanket with a density of 88 kg/m^3.

The final calculated curve from the Morse/Beranek theory is shown by the upper solid curve in Fig. 6. (Note that the frequency scale is converted to that of the final muffler.) The vertical scale should be multiplied by 3.28 to get the attenuation in dB/m. It was assumed that there were no traveling waves in the space behind the panels.

The results of measurements made in a 2.5:1 scale model of the duct are also shown in Fig. 6. Hence, the maximum full-scale dB/m calculated value is 1.2 times 3.28, i.e., ca 4 dB/m. This full value was not measured because of the 0.6 m open airspace even though thin partitions were introduced every 2.5 m (shown by the dashed curve) to eliminate traveling waves. When the partitions were randomly spaced at 1.8 ft, 2.5 ft and 2.8 ft, the better solid measured curve was obtained.

These model tests showed that the attenuation in the completed duct would be about 1.6 dB/m between 30 Hz and 280 Hz. Actually, in the completed duct, the attenuation measured 1.6 dB/m between 20 Hz and 600 Hz, giving an overall attenuation for 27.6 m of 44 dB.

Because, our goal was to achieve 25 dB of reduction in the 20 Hz to 600 Hz frequency region, this was a significant over-design . This fact was brought to the attention of the NASA people. They said that because the predictions were based on measurements in a 2.5 scale model, they wanted no chance of error when the full scale muffler was built, so the overdesign was approved and built.

The 20 dB of attenuation that was needed above 600 Hz was achieved with parallel baffles located in the outlet chamber section of Fig. 3, 10 cm thick spaced 40 cm on centers and 3.7 m in length .

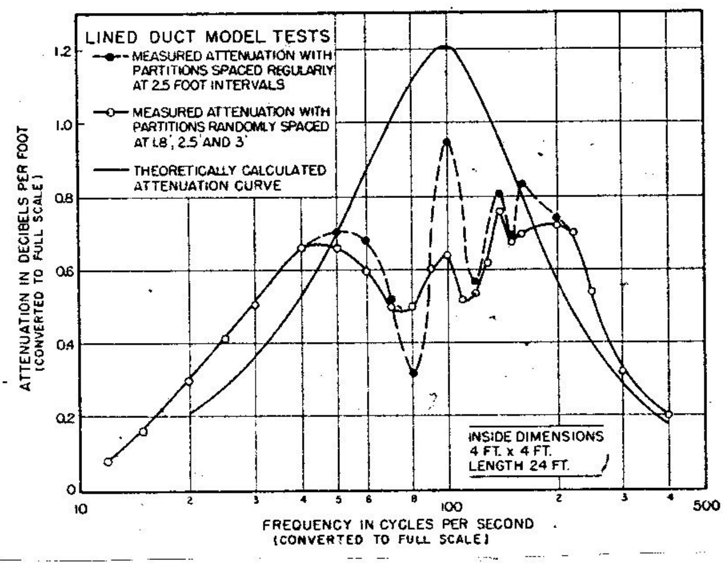


Figure 6: Measurements of duct attenuation in dB/ft in a 2.5:1 scale model.

2 Contributions to Loudspeaker Design

In my acoustics class at MIT I put together a new way to determine the performance of loudspeakers. I first presented it in my lectures at the Instituto Radiotécnico de Universidad de Buenos Aires in 1947. My book ACOUSTICS (Acoustical Society of America, 1954) was translated into Spanish. The material below is from ACOUSTICS.

I was the first person to develop an equivalent circuit that had all three domains in it, electrical, mechanical and acoustical elements. Prior to that, only the mechanical and acoustical elements were shown in the equivalent circuit. My reasoning started with the (a) representation in Fig. 7 below.

A direct-radiator loudspeaker has a cone, which has a mass M_{MD}, a suspension with a compliance $C_{MS,}$ and a mechanical resistance $R_{MS} = 1/r_{MS}$. The air into which it radiates puts a load on the cone —actually on both sides of the cone. The top sketch in Fig. 7 shows that one side of a mass is always at ground. The compliance looks like a spring and the resistance looks like a dashpot. All components vibrate at the velocity of the cone u_c. What we see is that u_c is “across” each element (like voltage) and this means that there is a force “through” each element (like current). As shown in (b) of Fig. 7. We can replace the mass element with a capacitance, i.e., $u = f/j\omega M_{md}$. The compliance element is replaced by an inductance, $u = f \times j\omega C_{MS}$. And the mechanical resistance is r_{MS}. . The air load on each side of the loudspeaker is expressed in the circuit by the impedance z_{MR} .

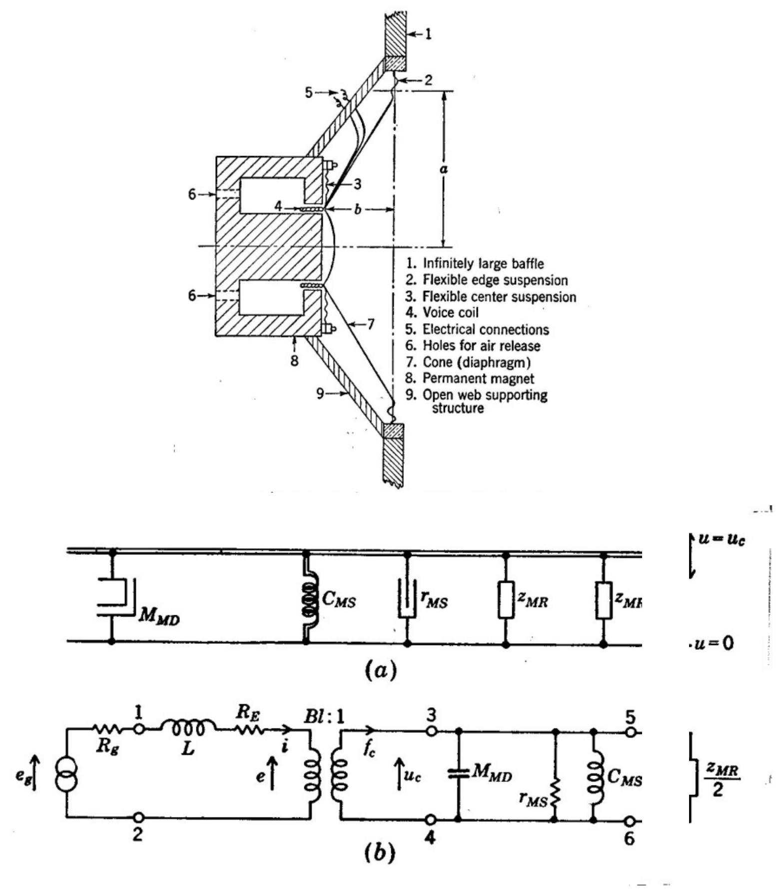
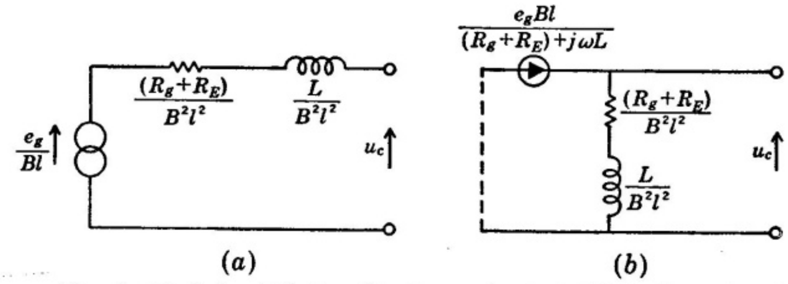


Figure 7. Director radiator loudspeaker. (a) Mechanical appearing diagram. (b) Equivalent electrical circuit of the mobility type.

The electrical side is brought into the circuit by means of a transformer that obeys the law $e = B\ell u$, where B is the magnetic flux density, ℓ is the length of wire on the voice coil, and e is the drop across the input side and u the drop across the output side of the transformer , also shown in (b) of Fig. 7.

The transformer can be eliminated by using either of the circuits (a) or (b) in Fig. 8. But, most engineers are uncomfortable with a circuit element that equates mass to a condenser, compliance to an inductance. Also desired is a circuit that shows the elements in series instead of in parallel.



The electrical circuit (referred to the mechanical side) is shown here in two equivalent forms. The circuits are of the mobility type.

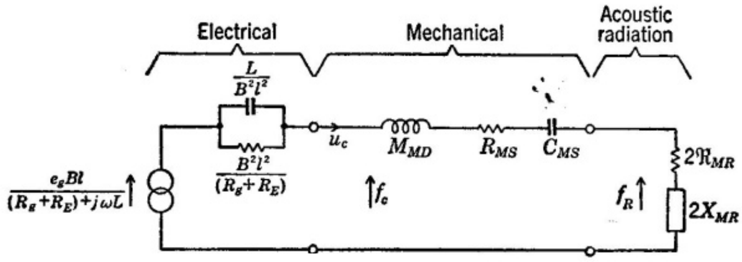


Figure 8. The circuit of Fig. 7 converted to its dual: velocity through the elements.

I innovated a means for making this conversion by taking the "dual", using the technique that follows:

A ground point (6) is established and it connects through the generator to (1), through the transformer to (2), through the mass of the cone to (3) through the resistance of the cone to (4) through the compliance of the suspension to (5) and through the radiation load on the two sides of the cone through (6). By following these numbers and changing inductances to capacitances and capacitances to inductances in the dual we get the circuit in Fig. 8 above.

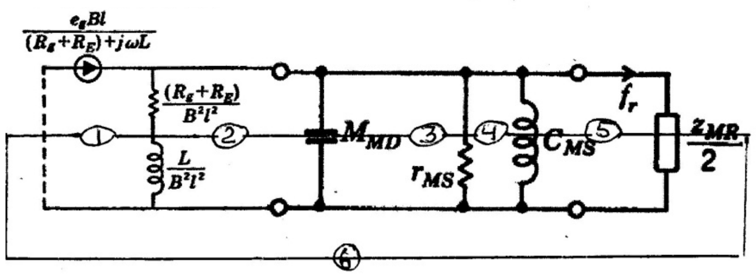


Figure 9: Method for taking the dual of a circuit.

When the loudspeaker is mounted in a closed box, the box looks like a compliance C_{AB} and it appears in series with the compliance C_{AS} of the cone, as shown below.

Two of my students were Edgar Vilchur and Henry Kloss. They looked at this circuit and said, "Let us mount the voice coil on a very flexible suspension, thus allowing it to "float", that is to say, to have a very large compliance C_{AS}. Even a relatively small box, say 0.25 m^3 will have a very large compliance C_{AB}. With both of these compliances large in magnitude, the resonant frequency of the loudspeaker will be very much lower than it was for prior speakers. This will give better bass response—the only negative feature is that the radiation efficiency will be lower and this will require a more powerful amplifier ". Their innovation means that today we have loudspeakers in shelf-sized boxes with acceptable bass response.

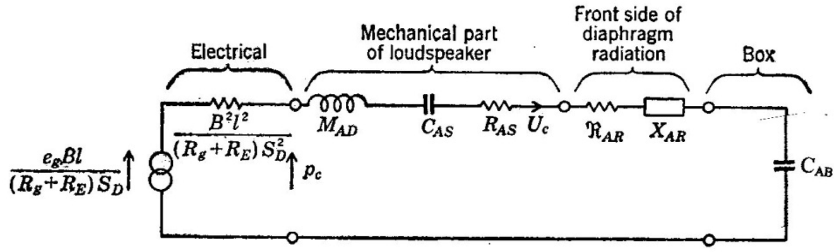


Figure 10. The final equivalent circuit for a loudspeaker in a closed box.

3 Concert Halls and Opera Houses

Concert hall and opera house acoustics have been my principle interest through most of my professional life. In 1955 an unusual opportunity arose for me to make a detailed study of the acoustics of concert halls. I traveled and attended concerts in England (five cities), Austria (two cities), Germany (six cities), Denmark, France, Finland (two cities), the Netherlands, Sweden, Switzerland (three cities), and the USA (14 cities). I interviewed twenty-three outstanding conductors and performers and 21 music critics in 17 cities. Technical data were obtained from acoustical measurements made in all of the halls from acousticians around the world.

These studies revealed that the best liked halls are shaped like shoeboxes, are not wide (less than 26 m), and have reverberation times (fully occupied) in excess of 1.8 s. Fan shaped halls are least well liked, and halls in which the audience surrounds the orchestra are mixed. In the surround halls, usually there are a number of sets with excellent acoustics, but there are locations in the hall where some sections of the orchestra are emphasized over others, other locations where the music seems to come from a side surface , and others where the sound is not natural.

Throughout my studies I have had the privilege of working with three acoustical engineers who have traveled to most of the halls that I have studied. They have obtained acoustical data which give guidance to all readers of my books on how to produce halls that make attendance at concerts a pleasure. These engineers are Dr. Takayuki Hidaka, Dr. (Ms) Noriko Nishihara and Dr. Toshiyuki Okano of the Takenaka Research and Development Institute of Chiba, Japan.

The principal measures of acoustical quality that are generally agreed to and which they measure are:

Reverberation time—the time it takes for a loud sound to die to inaudibility—sometimes called fullness of tone or envelopment; **Strength**—loudness; **Warmth**—strength of the bass sounds; **Binaural Quality Index** (a very important characteristic)—a measure of spaciousness, meaning that the sound seems to come from a wider source than is actually creating it; **Initial-Time-Delay Gap**—the time between when a direct sound arrives and when the first reflections arrive at a listeners position—this measure is closely allied to the width of a hall. **Surface diffusivity**—irregularities on the walls and ceiling that are large enough to give the reverberation a mellow tone. Let us now look at some well liked halls of the world.

3.1 Concert Halls

The **Vienna Grosser Musikvereinssaal**_opened in 1870 and is the home of the Vienna Philharmonic, a venerable orchestra of Europe. Everywhere are gilt, ornamentation, and statuettes (See Fig. 11.). The superior acoustics of the hall are due to its shoebox shape, its relatively small size (1,680 seats),its irregular interior surfaces, and its relatively long reverberation time (2.0 s). The interior is plaster or plaster on concrete block which makes for a strong bass. Its acoustics are similar to those in Boston Symphony Hall, except louder because of the smaller sized. The "Binaural Quality Index" equals 0.64.

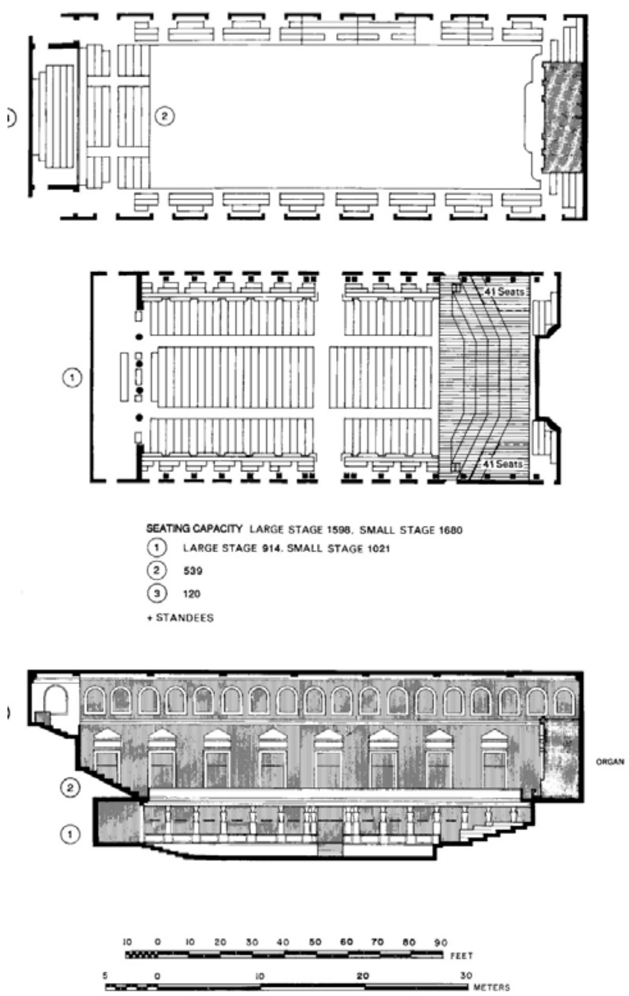


Figure 11. Grosser Musikvereinssaal in Vienna, Austria.

Boston Symphony Hall, built in 1900, is shoebox in shape, with a high, horizontal, coffered ceiling and two wraparound balconies. It resembles Vienna's Grosser Musikvereinssaal but it is different, primarily because it seats 2,625 people compared to 1,680 for Vienna. (See Fig. 12.) The sound is clear, live, warm, brilliant and loud, without being overly loud. The sound is remarkably uniform in quality throughout the hall. The interior surfaces are irregular which gives the reverberation a rich quality and the high open space above the top balcony allows the reverberation to reach its maximum effect—not possible in halls where the seating on one of the surfaces extends nearly to the ceiling. The reverberation time (occupied) is 1.9 sec. and the "Binaural Quality Index" is 0.61.

The **Tokyo Opera City Concert Hall** has a unique interior shape for a concert hall. It embodies a pyramidal ceiling with its apex reaching 28 m above the main floor. Seating 1636, this hall opened 1997. (See Fig. 13.) Beneath the lighting trough which is above the top balcony, the hall is shoebox in shape, a property that assures uniform sound throughout the listening spaces. The acoustical measurements confirm that the hall has optimum occupied reverberation time (1.96 sec), and excellent clarity, intimacy, spaciousness, warmth and strength of sound. The "Binaural Quality Index" equals 0.72. I was the Design Acoustical Consultant, working with the Takenaka Research and Development Institute.

3.2 Opera Houses

Next, I would like to report on a study of 23 opera houses in Europe, Japan and the Americas. The seating numbers in these houses vary from 1125 to 3816, while the volume and reverberation times (occupied) range from 7000 m^3 to 24 724 m^3 and 1.1 s to 2.0 s, respectively.

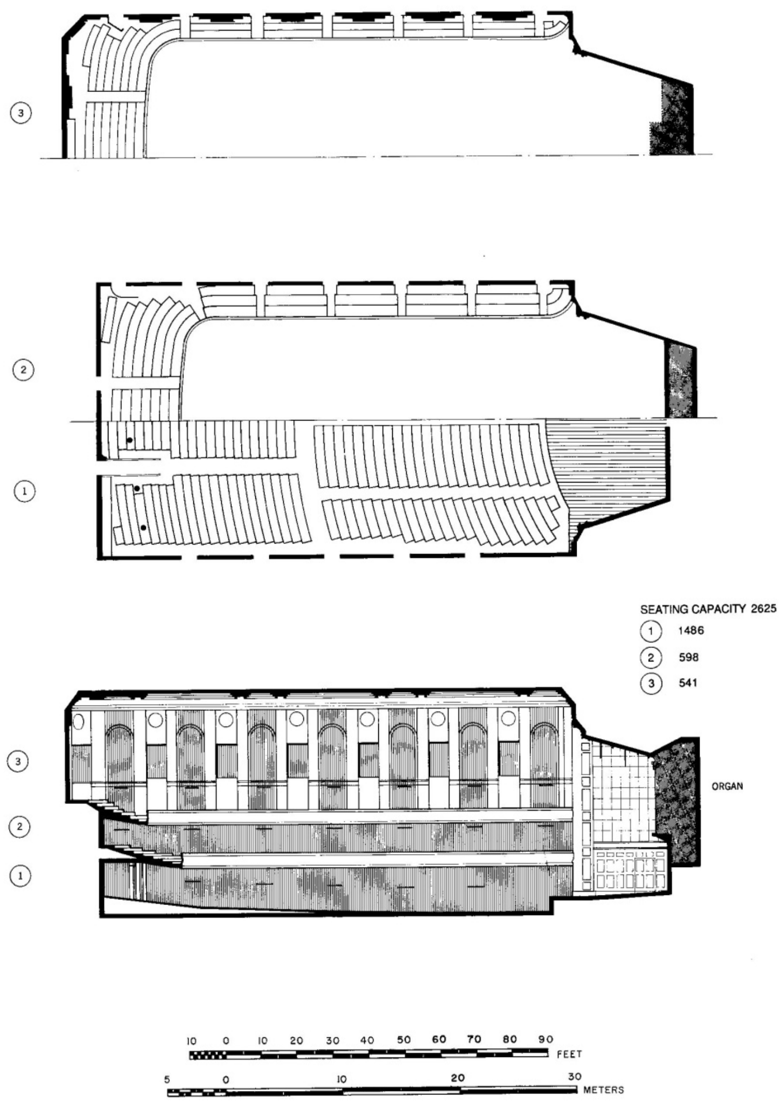


Figure 12. Symphony Hall in Boston, Massachusetts, USA.

Questionnaires were mailed to 67 important opera conductors and 22 responded, with one response not usable. The questionnaire asked for acoustical quality ratings of 24 halls on a scale from “Poor” to “One of the Best”, with intermediate steps “Passable”, “Good”, and “Very Good”. They were asked to rate the sound in the audience and the pit separately.

The results are given in Fig. 14. It came as a surprise that the highest rated opera house in the world is the Teatro Colón in Buenos Aires. It was no surprise that the venerable Milan, La Scala and Dresden, Semperoper came in next. Unfortunately, seven houses were rated less than good. In all the houses, except three, the ratings in the pit and the hall were the same. In those three, the pit sound was rated above that in the hall.

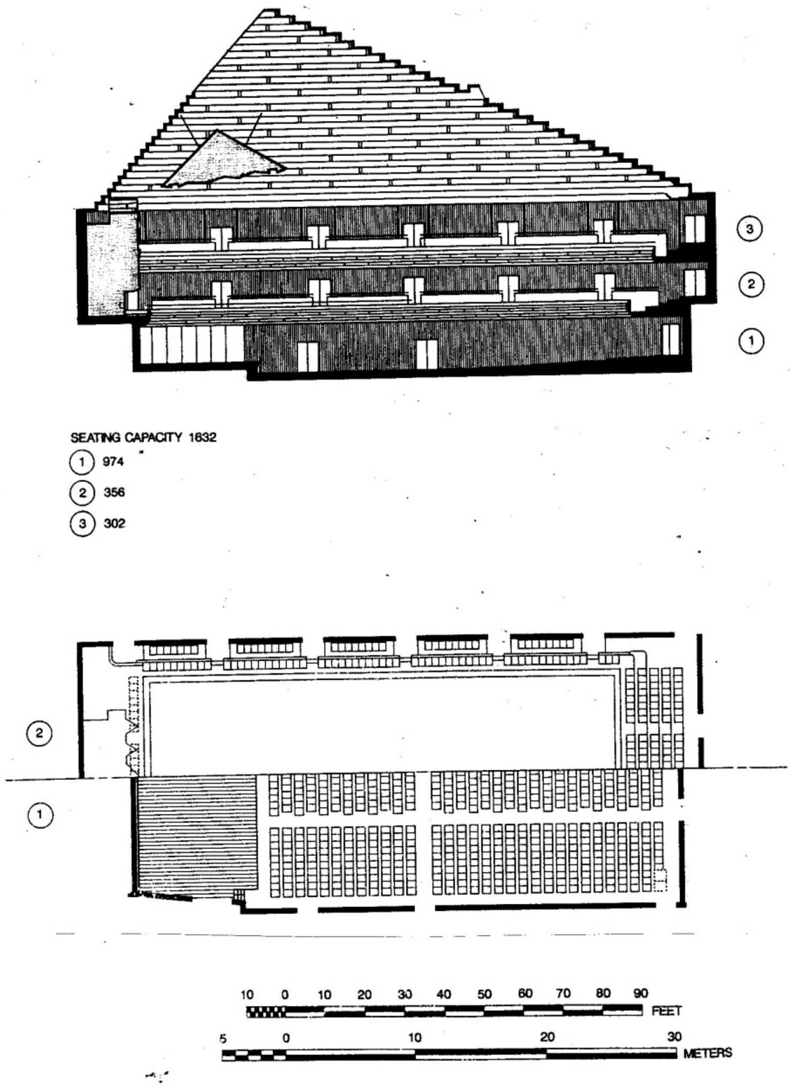


Figure 13. Tokyo Opera City Concert Hall, Japan.

The **Teatro Colón in Buenos Aires** is larger than most opera houses in Europe, seating 2,487 plus standees. Besides the main floor there are six rings plus a standees area at the top. The sound is remarkably uniform, although some prefer the top two rings because of augmented sound from the main ceiling. Performers say that the house responds easily to their singing and that the orchestra does not drown out the singers. The reverberation time (occupied) is 1.6 sec, about 0.2 sec above the average for other houses. The acoustic measurement called "Binaural Quality Index" is 0.65, larger than that for all opera houses for which data are readily available. A concert shell is erected on the stage for symphonic concerts and the hall is surprisingly good for the standard symphonic repertoire.

The **Staatsoper in Vienna** is a world-famous opera house. It is relatively small, seating 1,709. Compared to the New York Metropolitan Opera House's 3,816 seats and San Francisco's 3,252 seats, it is a miniature. It was rated slightly lower than La Scala in Milan, but its small size makes for intimate productions and enables the casting of singers with less than tremendous voices. Its reverberation time (occupied) is 1.4 s. The "Binaural Quality Index" is an excellent 0.60. This house is not used for symphonic concerts.

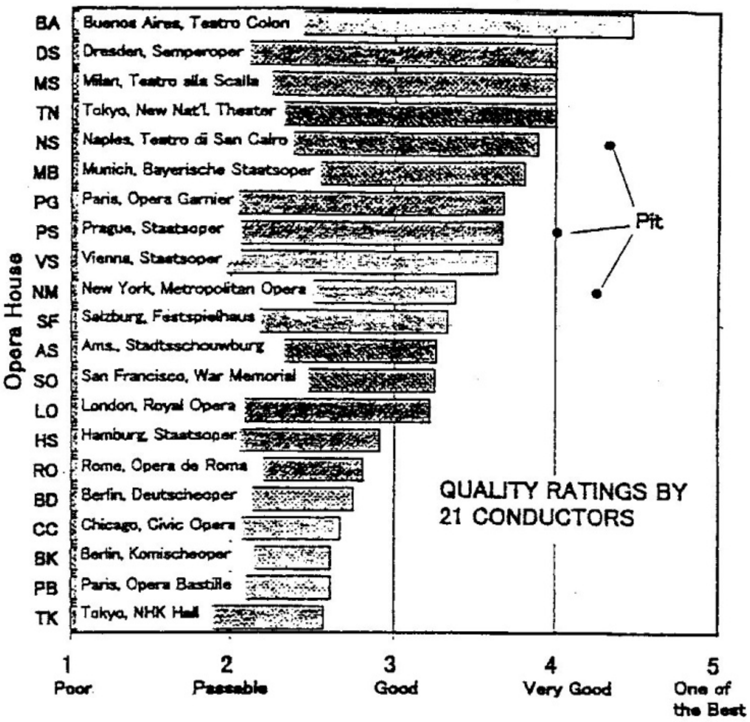


Figure 14. Acoustical quality ratings in the audience areas of 21 opera houses by 21 conductors. Only halls that received six or more ratings are included in this figure.

The **New National Theatre, Opera House, in Tokyo, Japan,** one of the recently built opera houses. It also is relatively small, seating 1,810. Its acoustics were rated equal to the Semperoper in Dresden, and La Scala in Milan. The architect, Takahiko Yanigasawa states, "The design facilitates the exchange of energies between those on stage and those attending the performance. The people in the audience are positioned so that they surround the stage as much as possible, and the balconies encourage empathy amongst them as they face each other across the hall." Its reverberation time (occupied) is 1.5 s. The "Binaural Quality Index" is an excellent 0.65. This house is not used for symphonic concerts.

I was the Design Acoustical Consultant, working with the Takenaka Research and Development Institute.

APPENDIX: I have written books at three different times on the acoustics of concert halls and opera houses: All findings as well as drawings (to the same scale) and photographs are included: In 1962, 54 halls were analyzed (*Music, Acoustics & Architecture*, Wiley). This was followed in 1996 by 76 halls (*Concert and Opera Halls, How They Sound*, Acoustical Society of America) and in 2004 by 100 Halls (*Concert Halls and Opera Houses, Music, Acoustics and Architecture*, Springer). I published a paper in 2008, that updated the conclusions of the 2004 book (*Concert Hall Acoustics—2008*, J. Audio Eng. Soc. Vol. 56, 532-544).

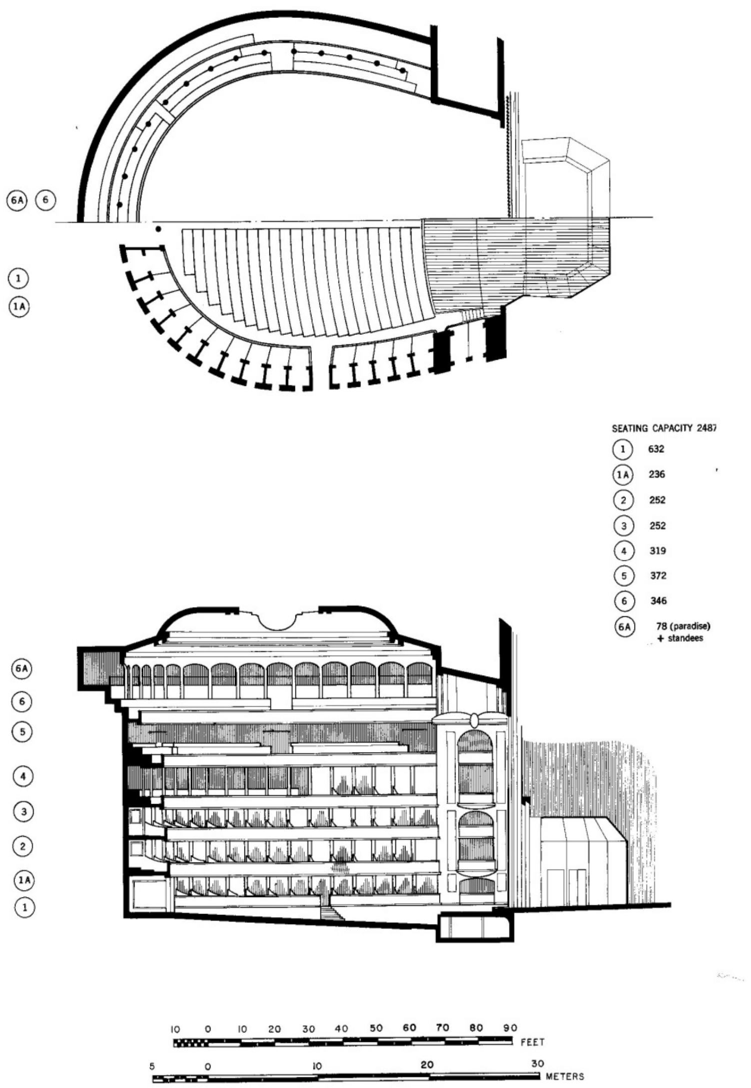


Figure 15. Drawings for the Teatro Colón opera house in Buenos Aires, Argentina.

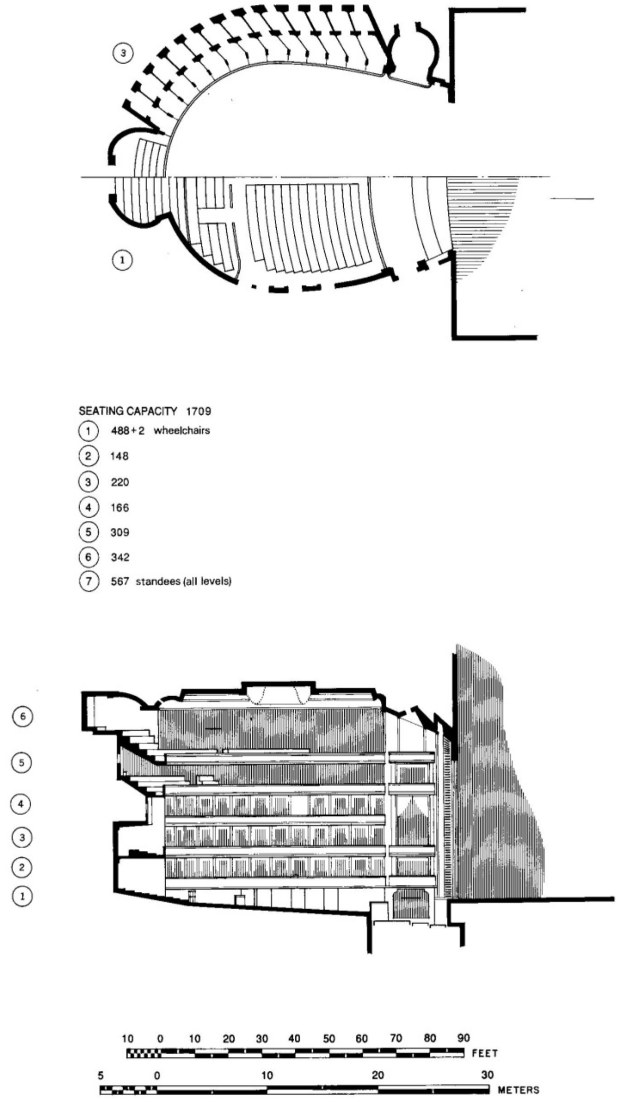


Figure 16. Drawings for the Staatsoper opera house in Vienna, Austria.

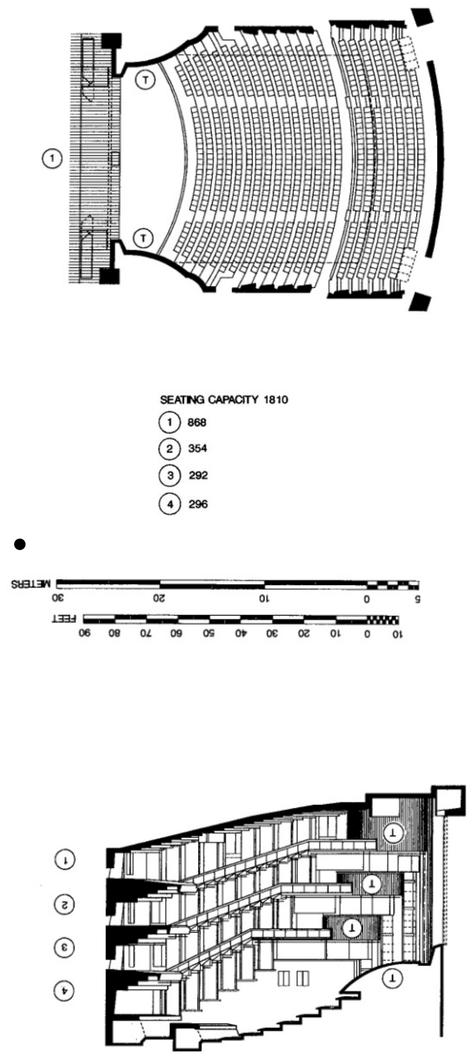


Figure 17: Drawings of the New National Theatre, Opera House, in Tokyo, Japan.

C02
European Aircraft Noise Research Network

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Abstract: The X3-NOISE Coordination Action, through its network structure and comprehensive workplan involving expert groups, scientific workshops, stakeholder seminars and a common information system, addresses the aircraft noise challenges set by the ACARE 2020 Vision. To this end, X3-Noise undertakes the elaboration and coordination of research activities, the dissemination of results and the integration of European research activity in the field of air transport related to noise. The objectives of the project can be summarised as follows: 1) Evaluation of EC-funded projects and assessment of their contribution to the state-of-the-art. 2) Formulation of priorities and key topics for future technology projects through development of common strategies, identification of required infrastructures, benchmarking against competing countries and complementarity with national activities. 3) Formulation of research strategy for the development of tools / instruments aimed at supporting the management of airport environmental capacity within the frame of ICAO Balanced Approach and EC Environmental Noise Directive. 4) Identification of potential reinforcement of future project partnerships through expertise mapping to foster new collaborations and promote novel ideas. 5) Optimum exploitation of anticipated breakthroughs in Aircraft Noise research by the scientific community, SME's interested in aeronautics applications and other noise engineering areas. 6) Appropriate communication and feedback strategy to inform on progress made in aircraft noise research, including the development of technical information aimed at Regulatory Bodies and Policymaking Agencies on the basis of state-of-the-art, demonstrated technology and operational capabilities of future aircraft. 7) Improved integration of the European Aircraft Noise Research Community through a network of National Focal Points (NFPs), including the structured development of local networks in new EU member states in order to foster participation in future projects. Over 4 years, the project involves 32 partners from 20 countries (FR, UK, ES, NL, BE, DE, SE, IT, PL, HE, HU, CZ, LT, IE, PT, RO, CH, UA, EG, BR), combining the complementary skills and expertise of industry partners, SMEs, university and research establishments to cover the whole field of interest. The international co-operation aspects of the research agenda to be developed through the project activity are further reinforced by the participation of 3 partners from Ukraine, Egypt and Brazil acting as Focal Points at Regional level.

1 Introduction

The X3-NOISE Coordination Action, through its network structure and comprehensive work-plan involving expert groups, scientific workshops, stakeholder seminars and a common information system, addresses the aircraft noise challenges set by the ACARE 2020 Vision.

To this end, X3-Noise undertakes the elaboration and coordination of research activities, the dissemination of results and the integration of European research activity in the field of air transport related to noise.

The objectives of the project can be summarized as follows:

- Evaluation of EC-funded projects and assessment of their contribution to the state-of-the-art.
- Formulation of priorities and key topics for future technology projects through development of common strategies, identification of required infrastructures, benchmarking against competing countries and complementarity with national activities.
- Formulation of research strategy for the development of tools / instruments aimed at supporting the management of airport environmental capacity within the frame of ICAO Balanced Approach and EC Environmental Noise Directive.
- Identification of potential reinforcement of future project partnerships through expertise mapping to foster new collaborations and promote novel ideas.

- Optimum exploitation of anticipated breakthroughs in Aircraft Noise research by the scientific community, SME's interested in aeronautics applications and other noise engineering areas.
- Appropriate communication and feedback strategy to inform on progress made in aircraft noise research, including the development of technical information aimed at Regulatory Bodies and Policymaking Agencies on the basis of state-of-the-art, demonstrated technology and operational capabilities of future aircraft.
- Improved integration of the European Aircraft Noise Research Community through a network of National Focal Points (NFPs) , including the structured development of local networks in new EU member states in order to foster participation in future projects.

Over 4 years, the project involves 32 partners from 20 countries (FR, UK, ES, NL, BE, DE, SE, IT, PL, HE, HU, CZ, LT, IE, PT, RO, CH, UA, EG, BR), combining the complementary skills and expertise of industry partners, SMEs, university and research establishments to cover the whole field of interest. The international co-operation aspects of the research agenda to be developed through the project activity are further reinforced by the participation of 3 partners from Ukraine, Egypt and Brazil acting as Focal Points at Regional level.

2 DUCAT – Basic research on duct acoustics and radiation

2.1 General Information

Since fan noise will be a major contributor to theexterior noise of Very High By-pass Ratio (VHBR) and Ultra High By-pass Ratio (UHBR) turbofans, aerospace industry is planning to introduce new nacelle noise reduction technologies as adaptive and active liners (actuators). Optimization of these reduction means requires a thorough understanding and accurate description of the sound propagation in ducts. Therefore, the main goal of DUCAT is to develop, extend and validate computational methods for the propagation and radiation of fan noise, including the effects of acoustic liners. A number of relevant aspects of this topic are not covered by the computation models existing today. Duct acoustics design tools have to be reliable, accurate, fast and versatile. According to aerospace industrial needs, these models should ideally be able to handle: realistic nacelle geometries and non-uniform flow (in intake and by-pass duct), - non-uniform acoustic liners and duct wall mounted actuators, radiation into the far field, realistic frequencies and Sound Pressure Levels.

Within short terms, it is not expected that all aspects can be addressed with a single model. Therefore in DUCAT a small number of numerical models (Finite Element (FEM), Boundary Element (BEM), coupled FEM/BEM, a non-linear propagation model and a ray-acoustics model) will be developed covering the whole frequency range of interest for fan noise (kRmax = 100). Focal points for the various models will be: for the BEM-model: acoustic radiation and the inclusion of non-uniform (potential) flow, for the 3D-FEM-model: acoustic radiation in sheared exhaust flow and 3-D nacelle geometry, for the coupled 3-D FEM/BEM model: influence of boundary layer flow on the effectiveness of liners, for the non-linear model: effect of liners on propagation just upstream of the fan, for the ray-acoustics model: high dimensionless frequencies (kR > 40). These models are partially complementary and partially overlapping, which offers the possibility to find the best modeling for each aspect of duct acoustics.

The models will be validated by various experiments in European anechoic wind tunnels. A main validation experiment will be carried out using a model turbofan in the German Dutch Wind Tunnel (DNvV). The experimental data will constitute a database for the validation of the codes developed in this project and for future applications. Also data from the previously CEC sponsored FANPAC-project will be used. After validation, the range of applications of the models and the restrictions for the use as industrial design tools for nacelle acoustic optimization will be established. Furthermore, as a case study, a liner design exercise on the nacelle of a generic VHBR turbofan will be performed.

Figure 1. DUCAT.

The final result of DUCAT will be an assessment of the applicability of the various computational models for duct acoustics problems and liner optimization. With the improved and validated models, the engine and aircraft industry will have the possibility to develop adequate design tools for both passive and active liner optimization. Further some spin-off to other industries is foreseen, since fluid machines as pumps, fans and internal combustion engines are major noise sources in modern society. The work in this project will clearly make progress beyond the state of the art by developing and extending computational models for duct acoustics and validating those by a small number of precise experiments.

Start Date: 1998-01-01
End Date: 2000-12-31
Duration: 36 months
Project Status: Completed
Programme Type: 4th FWP (Fourth Framework Programme)

Prime Contractor
Organisation : Nationaal Lucht- en Ruimtevaart Laboratorium (NLR)
Department : Fluid Dynamics Division Aeroacoustics Department
Address : 31 Voorsterweg PO Box 153
Postcode : 8316 PR
City : Marknesse
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Country : NETHERLANDS
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Project Partners
Aerospatiale Matra Airbus (FR), Rolls Royce (GB), BMW Rolls Royce (DE), TURBOMECA (FR), ONERA (FR), KTH (SE), ISVR Southampton (GB), University of Galway (IE), Université de Technologie de Compiègne (FR), Technical University of Denmark (DK)

3 RESOUND – Reduction of engine source noise through understanding and novel design

3.1 General Information

The principal aircraft and engine manufacturers in Europe are facing increasing pressure to reduce aircraft noise levels. This arises both from the community expectations of improved quality of life and from the need to compensate for the expected growth in air traffic. Consistent with an EC policy statement:"a coordinated strategic approach at European level is essential and major efforts need to be devoted to techniques for further reduction of exterior aircraft noise to overcome today's technology barrier. Hence the objective for R&D is to enable a breakthrough in noise control technology".

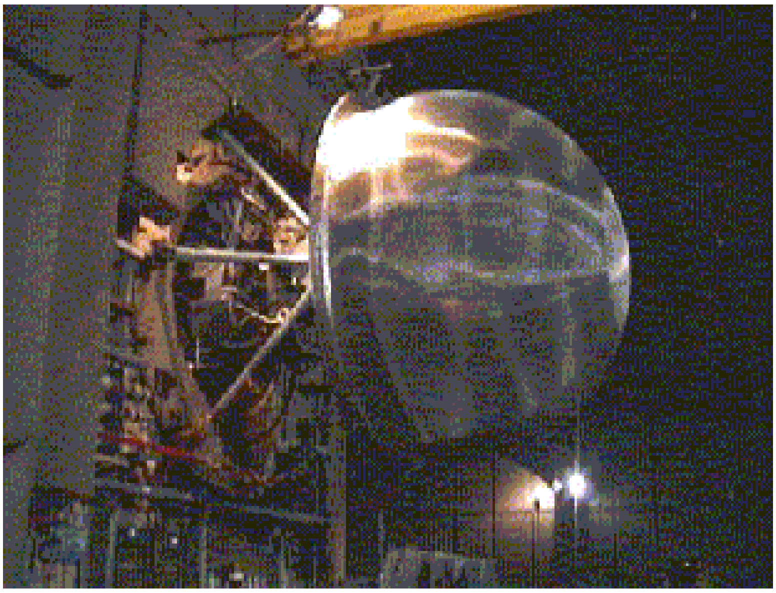

Figure 2. RESOUND.

The objective of RESOUND is to acquire the technology necessary to support the design of derivative and new aero-engines with noise levels that are 4 dB quieter than those of aircraft currently entering service. This will provide the foundation for the achievement of a mid-term (8 years) objective of reducing aircraft noise levels by at least 6 dB, and allow European industry to compete on an equal footing with the US.

RESOUND addresses the challenge of reducing the noise at source, in particular turbo-machinery noise, through (1) engine component aeroacoustic design and (2) through novel noise controlling devices that can be integrated within the engine structure. Innovative technologies to be evaluated, with the aid of theoretical techniques and experiments at model and full scale, include: fan noise reduction through reduced tip speed and pressure ratio optimization noise reduction with fan and stator axial sweep and circumferential lean fan noise reduction with variable by-pass nozzle and passive fan tip treatments combustion noise reduction through improved and validated generation and propagation model assessment of potential noise hazards of low NOx combustors LP turbine noise reduction through exit guide vane design turbo-machinery noise reduction through active stator design turbo-machinery noise reduction by means of auxiliary aeroacoustic control devices. Based on the technology acquired, RESOUND will deliver a full assessment of the community noise benefits of controlling engine noise at source, through design and with novel active/passive devices.

The reduction of aircraft noise through improved nacelle technology and airframe design is being addressed by complementary proposals (RANNTAC and RAIN respectively), supported by one Type 2 project (DUCAT), all of which will be coordinated through the X-NOISE thematic network that has been formed as a result of the Environmentally Friendly

Aircraft study (TEFA). Such a combined effort is necessary to meet the challenge of the US industry, which is backed by a fully funded program (200 MUSD over 7 years).

Start Date : 1998-01-01
End Date : 2000-12-31
Duration : 36 months
Project Status : Completed
Programme Type : 4th FWP (Fourth Framework Programme)

Prime Contractor :
Organisation : Rolls Royce plc
Department : Aerospace Group
Address : Moor Lane PO Box 31
Postcode : DE24 8BJ
City : Derby
Region : EAST MIDLANDS DERBYSHIRE, NOTTINGHAMSHIRE
Country : UNITED KINGDOM
Contact Person : Andrew KEMPTON
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Project Partners :
Aerospatiale Matra Airbus (FR), SNECMA (FR), MTU (DE), BMW Rolls Royce (DE), Dassault (FR), Dornier (DE), Daimler Chrysler (DE), NLR (NL), DERA (GB), ONERA (FR), DLR (DE), ISVR Southampton (GB), University of Galway (IE), University of Cambridge (GB), Ecole Centrale de Lyon (FR), Bertin (FR), Metravib (FR)

4 RANNTAC – Reduction of aircraft noise by nacelle treatment and active control

4.1 General Information

Community reaction to aircraft noise recognized as one of the most important constraints limiting future growth of air transport industry unless all future aircraft noise levels decrease in such a manner to offset the effect of air traffic growth on noise exposure. The current limitations of the state of the art aircraft noise reduction technology and the benefits that the US industry is starting to get in from the tremendous R & D effort in subsonic transport aircraft noise reduction coordinated by FAA and NASA (launched in 1994), urge the European industry to find now the appropriate answer.

The RANNTAC programme, together with the RESOUND programme on engine noise source reduction technology and the RAIN programme on the reduction of airframe and installation noise are the three main and complementary pieces of the proposed R & D effort coordinated at European level by the X-NOISE thematic network on external noise. This effort is needed to meet the requirements expressed in the EU consultation paper of November 96 for reduction in aircraft noise levels and accept the US implicit challenge a complying with much more stringent noise rules in the early 2000's.

The objective of the combined programme is to deliver within 8 year, aircraft environmental noise abatement of 6 decibels both at departure from airports and at arrival relative to current technology demonstrated by aeroplane industry. The objective of RANNTAC is to acquire the technology necessary to support the development and manufacturing of turbofan engine nacelles featuring noise reduction devices and designs enabling to achieve up to 4 dB attenuation on engine internal noise sources in addition to that achieved by currently produced acoustic liners. Half this noise reduction will be demonstrated at the end of the programme while the means to achieve the remainder will be identified and defined for further large-scale demonstration.

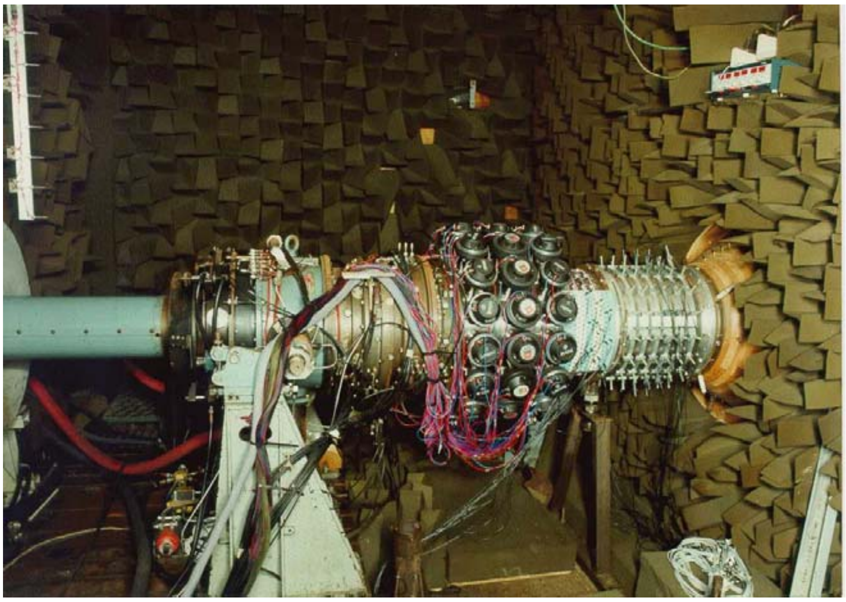

Figure 3. RANNTAC.

The practical results of the proposed research is to secure the design capabilities of developing in future transport aircraft nacelles, a large range of novel sound absorbing liners, air intake shapes allowing to control the fan noise radiation and in duct active noise control systems. The work programme will provide theoretical and experimental acoustic evaluation of all proposed systems and concepts in the context of real nacelle duct environment and assessment of the expected benefits on future aircraft noise together with the chances of industrial application and foreseen impact on aircraft economics.

The programme's structure features three tasks: Specifications, assessments, Exploitation and Management, Development of novel nacelle treatment concepts, Development of active Noise Control technology in nacelles. The work will be essentially carried out in parallel in tasks 2 an 3 each of them developing and providing in the end a panel of noise reduction solutions of different types, every type being evaluated on appropriate common test facilities.

Start Date : 1998-01-01
End Date: 2000-12-31
Duration: 36 months
Project Status: Completed
Programme Type: 4th FWP (Fourth Framework Programme)

Prime Contractor :
Organisation : Aerospatiale Matra Airbus
Department : Etude Générales - Performances - Acoustique
Address : 316 Route de Bayonne
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City : Toulouse
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Project partners:
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de Lyon (FR), Bertin (FR), Metravib (FR), Cambridge Concept (GB), Ferroperm (DK), CTTM (FR), EPF Lausanne (CH)

5 RAIN – Reduction of airframe and installation noise

5.1 General Information

The airframe itself can strongly influence the noise radiation from aircraft. Not only can the airframe through diffraction, refraction and reflection modify the noise radiated by the engines but it also acts as a strong source of additional noise radiation either directly from its components (for example the undercarriage and wing with high lift devices deployed) or indirectly by modifying or coupling with certain engine flow features. To date such sources of noise whilst important have had limited impact on the overall optimization of the aircraft. However, the trend towards very high by-pass ratio engines with their differing source breakdown coupled with the trend to aircraft of larger size offers the prospect that future new aircraft designs may well be limited in realizing their full operational efficiency and hence competitiveness by such acoustic effects. Therefore, there is a clear industrial need to develop an improved design capability which enables the effects of the airframe on the overall aircraft noise radiation to be addressed in the early stages of design. Such a capability is essential if airframe noise and installation effects are to feature in the aircraft optimization process.

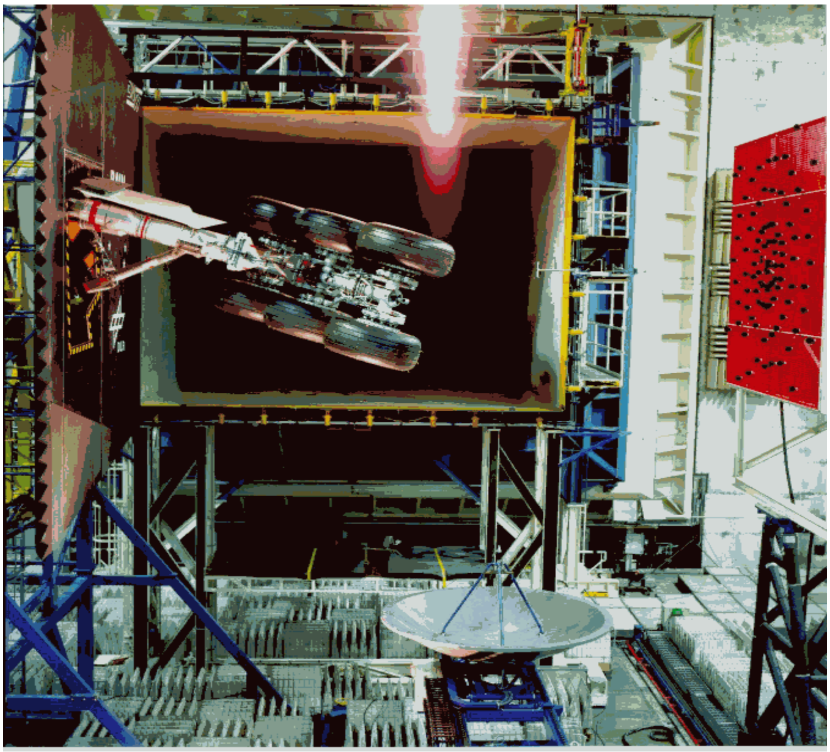

Figure 4. RAIN.

On approach the noise radiated from the airframe itself will be comparable with that of the engines and will limit the potential benefit of future engine noise reductions. The noise contribution from each of the components of the airframe need to be accurately predicted. As the nature and magnitude of airframe installation affects the noise radiation from the various engine sources in differing ways, the need is to accurately predict the installation effects for the main or dominant components - jet, fan turbine and combustion.

The current proposal aims to make a major contribution towards establishing such a competitive design capability. Advanced analysis tools will be developed based on sound theoretical approach at component level. New improved model scale experimental noise databases will be established and used in the development and calibration of these new analysis tools. Full-scale flight test data will be used to establish scale effects and for confirmation of the overall prediction capability. In parallel to the development of the analysis tools practical approaches to noise reduction will be identified and evaluated to provide confidence that such analysis tools are capable of implementation within design optimization studies.

The project is expected to realize at component level reductions in noise of the order of 5 to 10 dB. Full impact will be realized when the new airframe related technology developed here is integrated with that of the engine on an overall aircraft noise level basis when this is expected to result to a net improvement of less than 5 EPNL dependent on the engine noise contribution. This proposal is submitted within the framework Technical Area 3A.4 (environmental technologies) of the Brite-Euram Programme. As such it is one of the X - projects which form a cluster to address community noise reduction. It is believed that only a coordinated initiative at the European level, building on EC and National programs, expanding the scope of research topics to counter the US effort and providing the "demonstrator" capability that National programs cannot afford alone, will deliver the expected environmental and competitive benefit. The cost and scale of the project is further justified because of the number of tests required (wind tunnel occupation and the provision of a flight test aircraft costs are very significant) to obtain databases, so that the improved prediction tools of airframe noise and installation effects on engine (fan, jet and core) noise can be validated and design rules formulated. The noise reduction design will also be conducted and assessed (both in full scale and scaled model) in the wind tunnels. Together with the other noise cluster programs (RESOUND, RANNTAC) noise reduction achievement, the total noise reduction is expected to be 6 dB. The proposed work is very important as the airframe and installation noise is already affecting the existing aircraft, and need to be fully controlled in the design development of a new generation of aircraft (airframe noise may be a limiting factor in the development of the future large European aircraft) if community noise requirements are to be fully satisfied.

Start Date: 1998-01-01
End Date: 2000-12-31
Duration: 36 months
Project Status: Completed
Programme Type: 4th FWP (Fourth Framework Programme)

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Project Partners :
Aerospatiale Matra Airbus (FR), Daimler Chrysler Airbus (DE), Dassault (FR), Alenia (IT), Dornier (DE), Daimler Chrysler (DE), Messier Dowty (GB), Rolls Royce (GB), SNECMA (FR), MTU (DE), NLR (NL), DERA (GB), ONERA (FR), DLR (DE), ISVR Southampton (GB), University of Galway (IE).

6 SOURDINE – Study of optimization procedures for decreasing the impact of noise around airport

6.1 General Information

A critical side effect of the dominance of jet aircraft in Europe's increasingly congested skies is intrusive, damaging noise levels at airports and surrounding communities. Several projects funded by Directorate General XII are investigating the technology breakthroughs required to reduce aircraft noise emissions, while the DGVII project OURDINE, the French word f or "mute", is addressing the promising new area of possible changes in approach and take-off procedures.

The 13-month SOURDINE project is the first stage of a long-term programme aimed at defining new approach and take-off procedures for all European airports and at developing supporting tools. It will also define the ATM (air traffic management) activities to be carried out in the area of noise reduction under the 5th Framework Programme.

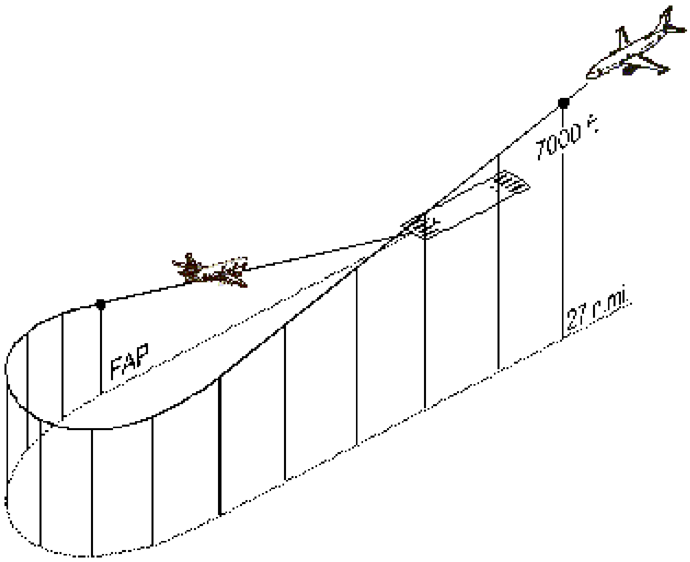

Figure 5. Take-off procedure.

SOURDINE will study and propose alternatives to reduce noise levels around airports by:

i. laying out general rules for most existing aircraft to update current approach and take off procedures;
ii. identifying short-term improvements in such measures as reduced flaps, delayed landing gear lowering and a higher descent speed;
iii. investigating and applying new procedures at selected airports - Schiphol, Madrid and Naples - to assess their feasibility. They will be tested in a simulated environment to measure their contribution to reducing noise without decreasing airport capacity or flight safety. In the next phase, candidate arrival and departure procedures will be tested in a real environment using defined measurement tools and methods;
iv. specifying and developing automation tools to support air traffic controllers and pilots in the application of new procedures. Operational global validation with the end users will take place to verify the impact and feasibility of these procedures.

The objective of this work is not only to reduce the overall noise encountered by people living near an airport, but also to ensure that airport capacity is not eroded. This ambitious project began on December 1, 1998 under the 4th Framework Programme and involves 11 partners in five member states. The development of the simulation tool and its calibration and the operational validation and the development of the automation tools for end users will be carried out in the 5th Framework Programme.

Duration: 12 months
Project Status: Completed
Programme Type: 4th FWP (Fourth Framework Programme)

Prime Contractor :
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Project Partners :
AENA (ES), INECO (ES), NLR (NL), RLD (NL), SICTA (IT), Air Support (IT), Aerospatiale Matra Airbus (FR), Air France (FR), DERA (GB), SerDB (FR)

7 SOURDINE II – Study of optimization procedures for decreasing the impact of noise around airport II

7.1 General Information

The objectives of Sourdine II are:- development and validation of new advanced innovative environmental friendly approach and departure procedures that have a positive impact on safety, capacity, environment and financial aspects.- development of an implementation plan to provide guidance for the migration from the current operational environment to the new procedures.- development of enabling technology to achieve the successful introduction of the selected departure and approach procedures, such as ATC controller tools, automated aircraft-ATC interaction tools and cockpit monitoring tools (e.g. safety nets)- development of a policy tool, enabling the policy makers to visualize and assess, for each new procedures, the relationship between safety, capacity, environment (noise and emissions) and the financial aspects.

Start Date: 2001-11-13
End Date: 2004-11-12
Duration: 36 months
Project Status: Execution
Programme Type: 5th FWP (Fifth Framework Programme)

Prime Contractor :
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Project Partners :
Ingenieria de Sistemas para la Defensa de España, S.A. (ES), Entidad Publica Empresarial Aeropuertos Españoles y Navegacion y Aera (ES), Ingenieria y Economia del Transporte S.A. (IT), Sistemi Innovativi per il Controllo del Traffico Aereo (IT), EADS Airbus S.A. (FR), European Organisation for the Safety or Air Navigation (FR)

8 TurboNoiseCFD – Turbo-machinery noise source CFD models for low aircraft engine designs

8.1 General Information

The European aircraft engine manufacturing industry is facing increasing pressure to reduce engine noise levels. The community expectations of improved quality of life through reduced noise levels and the current growth in air traffic are major socio-economic problems. A long-term solution is proposed here to create a new method for designing low noise turbo-machinery components through the exploitation of existing Computational Fluid Dynamics (CFD) software. This project would enable CFD software to be used for the prediction of noise, if the technical challenges such as dispersion and excessive memory and computational times can be overcome. If successful, the results of this project could be commercially exploited in the same way as the CFD codes are for turbo-machinery aerodynamics. The aim of this project is to contribute to the achievement of the Work Programme RTD objective of 10 dB reduction in 10 years in aircraft external perceived noise.

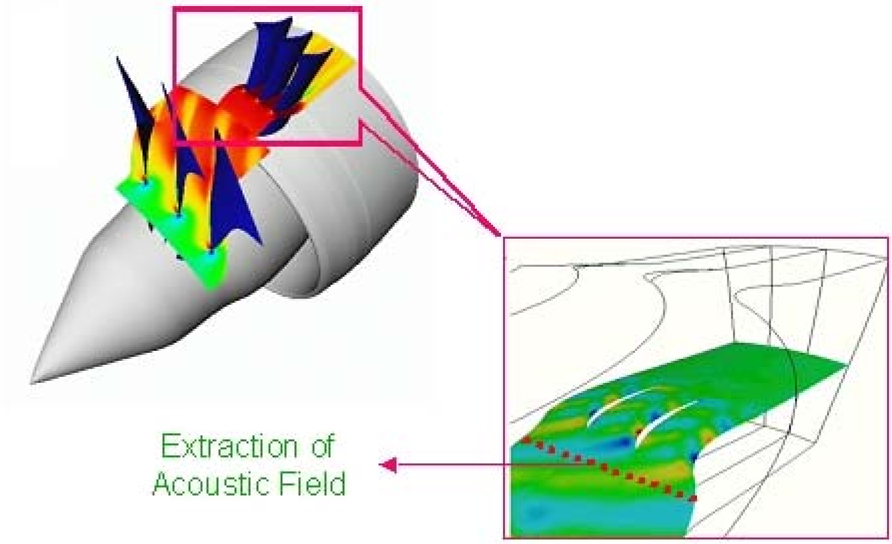

Figure 6. TurboNoiseCFD.

9 JEAN – Jet exhaust aerodynamics and noise

9.1 General Information

The JEAN proposal will address the prediction of noise by jet flows including the effects of mixing enhancement and co-axial configurations CFD techniques will be applied and validated to predict the turbulence characteristics of jets. These will be coupled to noise source generation and propagation models to estimate the near and far field noise. The result will be critically evaluated against data obtained from a series of carefully designed experiments.

The validated noise prediction procedures will be applied to mixing enhancement and to co-axial jets and the results compared with existing high quality data for these configurations. The work will recommend the use of a particular suite of techniques, which will have been validated for specific applications. These will then provide the basis for the development valuation tools for new concepts in low noise design of jet engines.

Start Date : 2000-02-01
End Date : 2003-02-01
Duration : 36 months
Project Status : Completed
Programme Type : 5th FWP (Fifth Framework Programme)

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Project Partners :
University of Cambridge (UK), University of Southampton (UK), Ecole Centrale de Lyon (FR), FFA - The Aeronautical Research Institute of Sweden (SE), Turbomeca SA (FR), Stichting Nationaal Lucht - En Ruimtevaart Laboratorium (NL), German Aerospace Centre (DE), Qinetiq Limited (UK), Snecma Moteurs SA (FR), Eindhoven University of Technology (NL), Office National d'Etudes et de Recherches Aérospatiales (FR), MTU Aéro Engines GMBH (DE), Technische Universitaet Berlin (DE), Université Pierre et Marie-Curie VI (FR), Industria de Turbo Propulsores SA (ES)

10 SILENCE(R) – Significantly lower community exposure to aircraft noise

10.1 General Information

SILENCE(R) addresses the issue of aircraft noise, a major cause of concern around European airports, through three major objectives:

- Large scale validation of noise reduction technologies whose development was initiated by EU and National projects in 1998;
- Assessment of the applicability of these technologies to current and future European products with minimum cost, weight or performance penalty;
- Determination of the associated achievable noise reduction, Novel concepts to be validated include low-noise fans, combustors and LP turbines, scared intakes, novel intake, bypass and hot-stream liners, nozzle jet noise suppressors, active control techniques and airframe noise reduction technologies.

Unless this technology can be developed and validated to reduce aircraft noise, traffic is likely to be limited by noise restrictions, affecting indirectly general economic growth. Expected results are consistent with the EC work program goal of 5 dB reduction. The total partnership involves over 50 organizations from 14 EU countries, one candidate country and one associated state. Aircraft, aero-engine, nacelle manufacturers, as end-users, are the main partners in SILENCE(R).

Figure 7. Silence(R).

Start Date : 2001-04-01
End Date : 2005-03-31
Duration : 48 months
Project Status : Execution
Programme Type : 5th FWP (Fifth Framework Programme)

Prime Contractor :
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Project Partners :
University of Southampton (UK), Centre National de la Recherche Scientifique (FR), Instituto Superior Tecnico (PT), Centre National de la Recherche Scientifique (FR), Trinity College Dublin (IE), Bruel & Kjaer Sound & Vibration Measurement A/S (DK), Mertravib Recherche Developpement Service SA (FR), German Aerospace Centre (DE), Institut National des Sciences Appliquées de Lyon (FR), Integreted Aerospace Sciences Corporation O.E. (HE), Sonaca SA (BE), Swiss Federal Institute of Technology Lausanne (CH), Industria de Turbo Propulsares SA (ES), ATECA - Application des Technologies Avancées S.A (FR), A4 Ingenieros Consultores SL (ES), Fokker Aerostructures BV (NL), Messier-Dowty SA (FR), Technical Research Centre of Finland (FN), Alenia Aerospazio - Un'Azienda Finmeccanica SpA (IT), Centre de Transfert de Technologie du Mans - Association pour les Transferts de Technologies du Mans (FR), Turbomeca SA (FR), Plansee AG (AT), Anotec Consulting, S.L. (ES), Hurel-Hispano le Havre (FR), Qinetiq Limited (UK), Aircelle SAS (FR), Dassault Aviation S.A. (FR), Airbus France SAS (FR), Dornier GMBH (DE), Rolls Royce PLC (UK), Walcher Elektronik GMBH (DE), Airbus Deuschland GMBH (DE), The National Research & Developement Institute for Turboengines Comoti R.A. (RO), Stichting Nationaal Lucht - En Ruimtevaaart Laboratorium (NL), Sener Ingenieria y Sistemas SA (ES), Short Brothers PLC (UK), Airbus UK Limited (UK), Rolls-Royce Deutschland LTD&CO KG (DE), Office National d'Etudes et de Recherches Aérospatiales (FR), EADS Deutschland GMBH (DE), Saab AB (SE), Fundacion Inasmet - Asociacion de Investigation Metalurgica del Pais Vasco (ES), Aermacchi SPA (IT), Fundacion Centro de Tecnologias Aeronauticas (ES), Subcontratacion de Proyectos Aeronauticos S.A. (ES), Siegel S.A. (ES), Inbis Technology LTD (UK), MTU Aero Engines GMBH (DE), Vibratec S.A. (FR)

11 FANPAC – Aeroacoustics Methods for fan-noise prediction and control

11.1 General Information

The objective of this programme is to provide European air-frame and nacelle manufacturers with the technology required to control fan tone noise on future civil transport aircraft equipped with advanced high-bypass-ratio (HBR, typically 6:1), very-high-bypass-ratio (VHBR, typically 9:1) or ultra-high-bypass-ratio (UHBR, typically 12:1-15:1), turbofans. Fan noise is predicted to be one of the most important noise sources on HBR, VHBR and UHBR engines and must be controlled if aircraft are to meet future community noise regulations.

Research in this field is required to keep European aircraft, aero-engine and nacelle manufacturers competitive with regard to the US. The objectives of the work are:

i. To establish the physical mechanisms responsible for the generation of fan tones, and to validate aero-acoustic models for predicting fan tone levels,
ii. To explore novel methods for controlling fan tone generation at source (in particular buzz-saw tones),
iii. To develop codes to predict noise attenuations by non-locally-reacting liners and non-axisymmetric liners,
iv. To develop a semi-empirical/theoretical model of wake propagation,
v. To design and test novel 500 Hz – 5000 Hz acoustic liners for fan noise control, and to validate models for predicting liner performance,
vi. To perform tests using a model fan rig to validate community and cabin noise prediction methods,
vii. To improve community and cabin fan tone noise prediction methods and to assess various techniques to control fan noise, aiming to reduce community noise levels by typically 4 dB and cabin buzz-saw noise levels by typically 5 dB – 10 dB.

11.2 Achievements

The FANPAC programme has addressed the opportunities for noise reduction at source by first developing an improved understanding of the noise source generation mechanisms. A considered and focused approach to noise reduction follows from this understanding. Similarly the optimization of noise reduction by conventional and novel acoustic liners follows from both a microscopic understanding of the internal liner behavior and a macroscopic understanding of the propagation of sound over complex impedance structures.

The development of understanding of the noise generation mechanisms in this programme has resulted principally from the testing of the model fan at a range of conditions in several different configurations and equipped with advanced acoustic and aerodynamic instrumentation.

To assess the potential benefit from novel, non-locally reacting, acoustic liners, a two step approach was taken by first analytically modeling the liner behavior in response to an applied sound field to determine impedances or characteristic properties and validating the model with laboratory impedance tests and second by developing attenuation prediction models for lined ducts.

Considerable progress has been made over the last 3 1/2 years towards the targets originally set. Studies into the human sensitivity to buzz saw noise have helped identify the problems for controlling cabin noise. There is potential to reduce fan noise by up to 5 dB at some conditions through source noise control and the use of acoustic liners. Other technologies such as active noise control may offer further benefits.

Start Date : 1993-01-01
End Date : 1996-06-30
Duration : 42 months
Project Status : Completed
Programme Type : 3rd FWP (Third Framework Programme)

Prime Contractor :
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Country : UNITED KINGDOM
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Project Partners :
University of Southampton (UK), University of Salford (UK), SNECMA (FR), Aérospatiale (FR), University College Galway (IRL), Danmarks Tekniske Universitet (DK), Nationaal Lucht- en Ruimtevaart Laboratorium (NL), Alenia (IT), Deutsche Aerospace Airbus GmbH (DE), Université du Maine (FR), Short Brothers (UK).

12 SNAAP – Study of noise and aerodynamics of advanced propellers

12.1 General Information

Recent progress in aerodynamics, aeroelasticity, materials and structures has enabled the design of innovative propeller configurations which operate at the same cruise speeds as jet-propelled aircraft. The inherently high propulsive efficiency of these advanced propellers and propfans allows fuel savings with a corresponding reduction in exhaust emissions. The major drawback is the level of noise emitted by propellers operating at high rotational speeds. This is partly due to a lack of detailed information on the understanding of the physical phenomena involved so that commensurate progress in the aeroacoustics of advanced propellers is now essential.

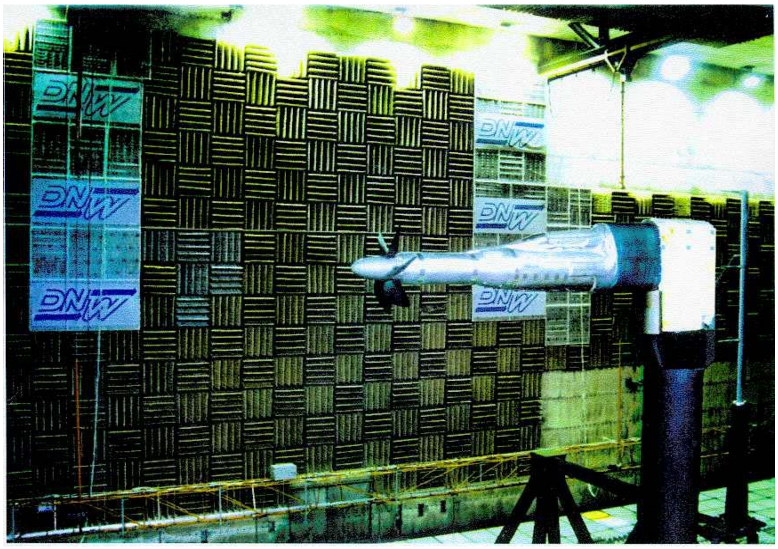

Figure 8. SNAAP.

12.2 Achievements

The project SNAAP has provided a complete data base from propeller wind tunnel basic test, i.e. without installation effect. Sophisticated techniques like blade fitted and inflow sensors adapted to acoustic measurements have been used. In addition aerodynamic and acoustic prediction codes have been developed and their validity investigated. The work program performed was addressed to: the definition of the parameters and the conditions to be tested in the wind tunnel, to the design and manufacturing of the propeller blade models, to the preparation and realization of the tests, to the preparation and validation of the theoretical aero-acoustic prediction tools.

Two types of aircraft flight conditions have been simulated to define the condition at which the propellers have been tested: take-off (low Mach number, high angle of attack) and cruise (high Mach number, low angle of attack). The design and manufacturing of the composite instrumented blades was a really challenge. Manufacturing of such instrumented propeller blade was new in Europe. All the activity was completed in 10 months, two sets of 8 blades delivered for wind tunnel tests. The two instrumented models are two 6-bladed propellers. The first one a Low Speed Propeller (LSP) is an advanced subsonic propeller for a cruise Mach number of 0.7 and helical tip Mach number of 0.88; the second one an High Speed Propeller (HSP) is a transonic one with a cruise Mach number of 0.78 and helical tip Mach number of 1.1. Both the propeller models have a diameter of 90 cm. The propeller assemblies have been completed and tested with success.

Although actual work started with one year delay, the wind tunnel testing activity met its targets beyond expectations. Two wind tunnel facilities have been used: ARA transonic wind tunnel, to simulate the cruise condition of the aircraft (high flight Mach numbers), and DNW low speed wind tunnel to simulate take-off, climb and approach conditions (low Mach numbers). Both wind tunnel tests have been safely conducted to a successful end. The amount of data accumulated is unique on European scene and meets the 99% of the project objectives, which is a very impressive result for such a challenging experimental exercise. Those data have been used to validate computer codes developed within the project.

An aeroacoustic code is now available to predict the aerodynamics and the acoustics of such advanced propellers. This is the most important result of the research programme. The code consists of several computer programs which are linked together by a simple procedure. The aerodynamic module of this code works within limits clearly investigated along the project: it cannot include non-axial inflow angles, and it is unable to converge below Mach 0.2.Two acoustic modules developed and validated within the project, are included in the code. Those modules use two different approach one in the time domain, the other in the frequency domain, both use the input from the unique aerodynamic module.

Also the acoustic modules work within some limits: when “classical noise terms” are computed (thickness, loading ...) the aero input does not influence so much the noise output. So that in take-off condition the predictions are very satisfactory but only with no in-flow angle of attack. In cruise conditions, the prediction tool can provide very helpful results for a large range of “not-too-high” flight Mach number. While when higher Mach number causes helical tip speed to be significantly supersonic, some numerical problems may arise in the loading and quadruple noise terms calculations.

Furthermore other acoustic tools have been developed within the project and are now available: one to predict the installation effect, due to the aircraft fuselage; another to identify the noise source using fluctuating pressure records. In addition some empirical methods or simplified models to account the scattering effect of the fuselage and of the angle of attack on the noise, have been developed.

Start Date : 1993-01-01
End Date : 1996-06-30
Duration : 36 + 6 months
Project Status : Completed
Programme Type : 3th FWP (Third Framework Programme),

Prime Contractor :
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Project partners:
Aerospatiale (FR), Dornier (DE), Fokker (NL), NLR (NL), Dowty (UK), Ratier-Figeac (FR), ONERA (FR), University of Galway (IE), CIRA (IT), University of Galway (IE), IBK (Ingenieurburo Dr.Kretzschmar), (DE), TCD (Trinity College Dublino), (IE), IST (Istituto Superiore Tecnico), (P).

13 APIAN – Advanced propulsion integration aerodynamics

13.1 General Information

The need to cope with rapidly evolving requirements for economically viable and environmentally acceptable propulsion systems has forced airframers to explore revolutionary systems such as ultra-high by-pass ratio engines especially open rotors. The development (and exploitation) of these novel power-plants requires technology to integrate then with the airframe to produce efficient performance and reduced noise for operational configurations. The ultimate goal is a competitive new generation of commuter aircraft with the same operational capacity and comfort as regional jets yet with reduced emission and noise. It is first proposed to investigate in wind tunnels advanced propeller driven aircraft equipped with high-speed propellers (Mach 0.7-0.8). The test rig will primarily consist of an advanced powered wind tunnel model developed in IMT3"GEMINI II" programme. This includes an internal six components balance, a set of two air driven high-power turbines (290 SHP Engine Simulators), a set of typical high-speed propellers (Mach 0.78) with rotating balances and instrumentation such as dynamic pressure sensors or microphones.

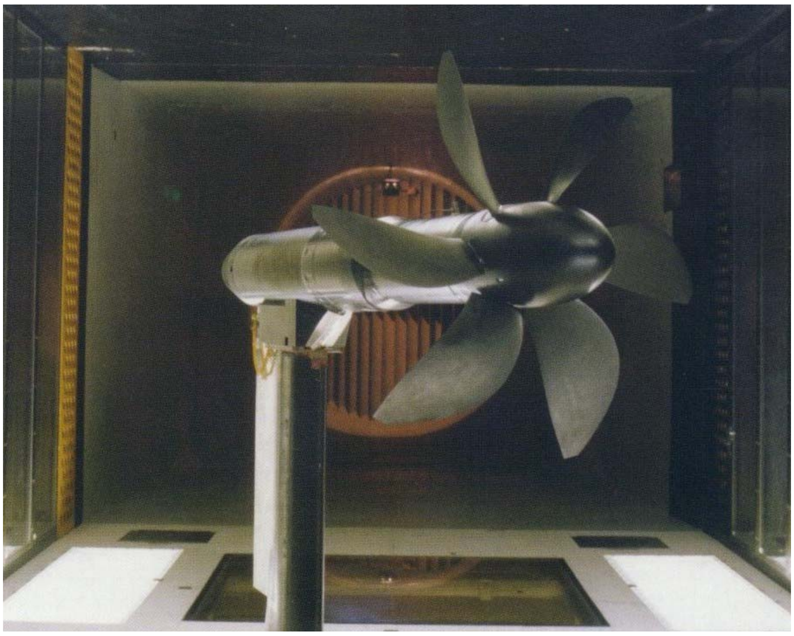

Figure 8. APIAN.

Wings, nacelles and propellers are original design from GEMINI and SNAAP CEC funded research programmes. The powered model will be tested in ONERA S1 transonic wind tunnel and in DNW low speed tunnel to investigate aerodynamics and acoustics of "propeller-to-airframe interactions". The same propellers will be tested on an isolated rig in the same conditions to identify the effect of the airframe. In addition, this isolated rig will be adapted to NLR-HST for transonic regime (up to 2 bars generative pressure) to identify Reynolds effects on the propellers (scale effect). Forces, moments, pressures, radiated noise will be measured together with flow field velocities by mean of techniques such as Particle Image Velocimetry (P.I.V.). Second goal is to develop the capability of propeller noise prediction in transonic regime when scattered by an airframe (boundary layer, wing, fuselage effects), SMEs, Universities and Research Centers will cooperate to generate such a computational tool. This will be a development and a first application of the codes initiated under the IMT3 "SNAAP" CEC funded research programme. Predictions will allow an optimization of the test matrix and experimental data will later on be compared to the theoretical ones to ensure quality of the industrial exploitation policy.

Start Date : 1996-06-01
End Date : 2001-01-31
Duration : 56 months
Project Status : Completed
Programme Type : 4th FWP (Fourth Framework Programme)

Prime Contractor :
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Project partners :
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14 AEROCERT – Aircraft Environmental Impacts and Certification Criteria

14.1 General Information

AEROCERT describes the research items to provide options for improvement of the certification standards, and recommendations for preferred options related to aircraft environmental impact.

The key objectives and principal tasks are:

To identify necessary revisions and/or extensions of the emission certification procedures.

a. To identify the known and possible impacts of aircraft emissions on the environment and to identify the data needed to quantify these impacts.
b. To define possible indices showing the actual impact of aircraft emissions on the environment or usable to quantify the actual impact on the environment.
c. To identify whether the existing certification procedures reflect the impact on the environment, considering different influences such as flight procedures.
d. To define options for possible improvements by changing standards and procedures or by extending them, considering the technical feasibility, effectiveness and economic impact of the proposed improvements.

To identify the effect of operational and maintenance procedures on the certified emission levels.

a. To identify the influence of the deterioration in the emission levels of engine and aircraft and to characterize the deterioration in noise and emission levels.
b. To make recommendations on operational and maintenance procedures to keep near to the certification levels.

Start Date : 1997-07-01
End Date : 2000-06-30
Duration : 36 months
Project Status : Completed
Programme Type : 4th FWP (Fourth Framework Programme)

Prime Contractor :
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Project partners :
Loughborough University of Technology (UK), Secretery of State for Defence, Acting Through Defence Evaluation and Reserarch Agency (UK), The Aeronautical Research Institute of Sweden (SE), Deutsche Forschungsanstal Für Luft - Und Raumfarhrt E.V. (DE).

15 ROSAS – Research on silent aircraft concepts

15.1 General Information

As a complement to the on-going research and development efforts deployed by the European aeronautical community towards the reduction of civil aircraft noise, efforts which focus on reducing noise emitted at the source by the engine and the airframe like in RAIN and SILENCE(R) programmes, the ROSAS project aims atdeveloping the necessary capabilities for the evaluation and selection of innovative silent aircraft concepts, characterized by the shielding of the engine noise sources by the airframe components (wing/fuselage/ empennage).

Figure 9. ROSAS.

ROSAS tackles the main critical issues of alternative installations of advanced turbofan engines through acoustic experimental and theoretical investigations, including a wind tunnel test campaign, and CFD-based aerodynamic work to identify the key phenomena and related risks and achieve an efficient shape design in the power plant area. This will be completed with a multi-disciplinary evaluation of the innovative concepts in order to achieve a fair comparison with the conventional under-wing engine installation.

Start Date : 2002-01-01
End Date : 22004-12-31
Duration : 36 months
Project Status : Execution
Programme Type : 5th FWP (Fifth Framework Programme)

Prime Contractor :
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16 AROMA – Acoustic radiation of small turbo-machines

16.1 General Information

Predicting the noise produced by turbo-machines requires three essential modeling components:

(1) an accurate description of the flow through the turbo-machine (WP1);
(2) a model predicting the amplitude of the acoustic source from the results of the CFD calculation (WP2);
(3) a prediction of the propagation of the source in a lined duct taking into account the flow field in the duct (WP3);

The different components must then be seamlessly integrated and connected to an optimization tool (WP4).

Finally each component of the loop and the integrated system must be validated against experimental results (WP5).

In order to be successful the project must be managed and its results disseminated and exploited (WP6).

The proposal is a result of the SCRATCH initiative and gives therefore a leading role to SME. The focus is also on the development of a product / methodology that can be exploited by the SMEs at the end of the project.

Start Date : 2001-12-01
End Date : 2004-05-31
Duration : 30 months
Project Status : Execution
Programme Type : 5th FWP (Fifth Framework Programme)

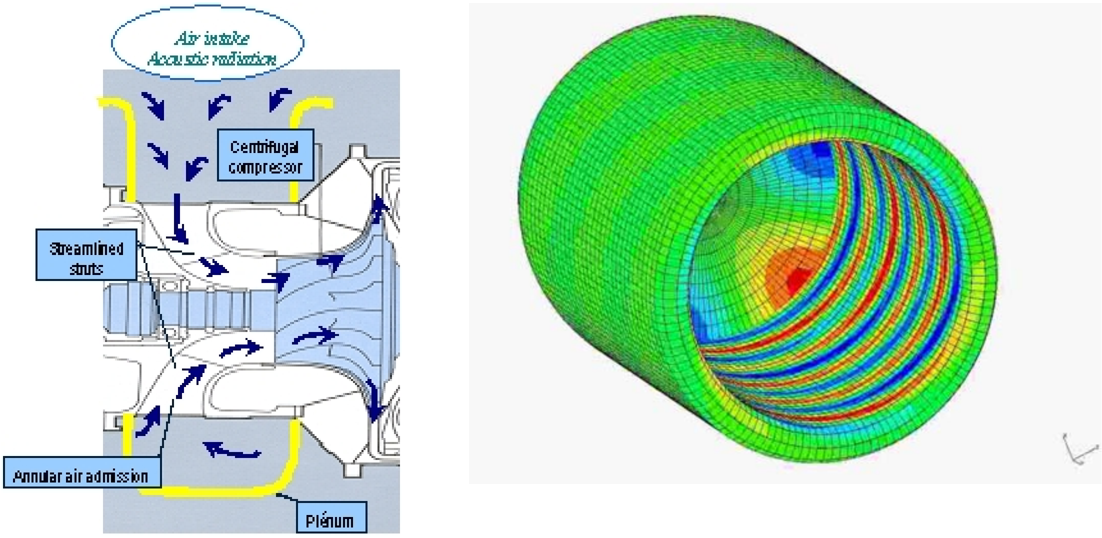

Figure 10. AROMA.

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Numerical Mechanics Applications International S.A. (BE), Liebherr Aerospace Toulouse SA (FR), Aerodisa (ES), Turbomeca SA (FR)

C03
Protectores auditivos: Atenuación de ruido y confort

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Abstract: This paper presents different hearing protectors types and the mechanisms of functioning including the possible ways of noise transmission paths. Noise attenuation measurement techniques are discussed and recommendations for their selection based on comfort and noise attenuation are presented.

Resumen: Este documento presenta diferentes tipos de protectores auditivos y los mecanismos de funcionamiento como las posibles formas de transmisión de ruido. Se discutirán las técnicas de medición de atenuación de ruido y serán presentadas recomendaciones para la selección basada en el confort y en la atenuación del ruido.

1 Introducción

Cuando las técnicas de control de ruido no están disponibles de inmediato, o hasta que se tomen acciones para reducir el ruido a niveles permisibles, el protector auditivo de uso personal es uno de los métodos más comunes y prácticos para reducir la dosis de ruido. Este tipo de solución no debe ser considerada como definitiva, debido a las características intrínsecas de los protectores, tales como: poco confort, dificultad en la comunicación verbal, etc. En este documento se discutirán, a modo informativo, los mecanismos por los cuales los protectores actúan en la atenuación del ruido y los posibles caminos de transmisión. Se presentan los tipos de protectores y las técnicas de ensayo de atenuación, las recomendaciones para su selección, uso y mantenimiento, y los posibles problemas que pueden encontrarse en su utilización.

2 Funcionamiento del protector

Como los daños a la audición ocurren normalmente en el oído interno, el protector auditivo actúa como una barrera acústica que protege tal parte del oído.

El funcionamiento de un protector auditivo depende de sus características propias y de las características fisiológicas y anatómicas del usuario. En el caso de un individuo con protector auditivo, la energía sonora puede llegar al oído interno por cuatro caminos diferentes (ver figura 1):

1. Transmisión Vía Ósea y Vía Tejido.
2. Vibraciones del Protector.
3. Transmisión a través del Material del Protector.
4. Filtración a través del Contacto.

Cualquiera de los cuatro factores mencionados anteriormente puede limitar la atenuación de ruido del protector.

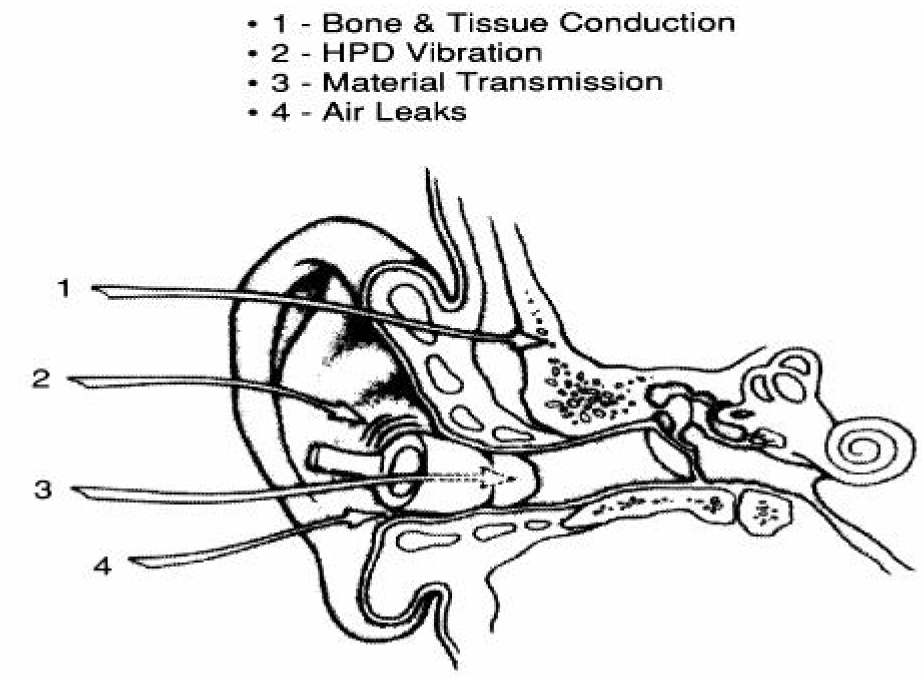

Figura 1. La energía sonora puede llegar al oído interno por cuatro caminos diferentes.

3 Tipos de protectores auditivos

Existen varios tipos, marcas y modelos de protectores auditivos, con alrededor de 1000 modelos distintos a nivel internacional. La figura 2 presenta algunos de los diversos modelos existentes en el mercado. Protector de inserción auto-moldeable, protector de inserción del tipo pre-moldeado, protector de inserción del tipo personalizado, protector del tipo orejera y tipos especiales de protectores auditivos.

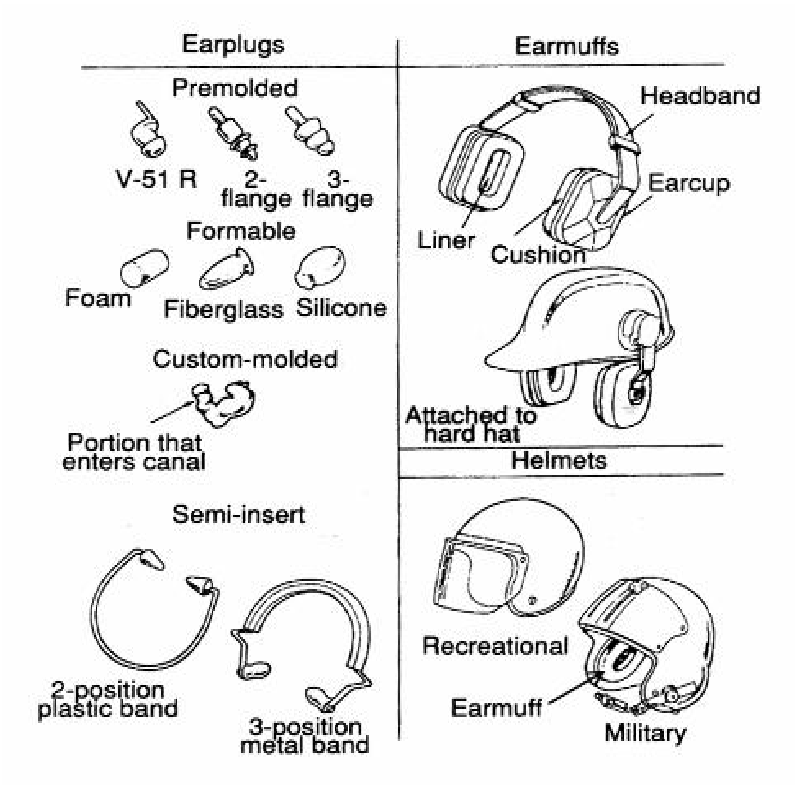

Figura 2. Tipos de protectores auditivos.

4 Reducción del ruido

El objetivo principal de los protectores auditivos es reducir los niveles excesivos de ruido a los cuales el usuario está expuesto, a un nivel aceptable. Para comparar la reducción del ruido debida a los distintos tipos de protectores, los ensayos de atenuación de ruido deben ser realizados de acuerdo a normativas nacionales e internacionales. Los fabricantes deben señalar las técnicas de ensayo utilizadas y los laboratorios donde los ensayos fueron realizados.

4.1 Ensayos de atenuación del ruido

La primera normativa sobre los procedimientos de ensayo de atenuación de ruido en laboratorios es la ANSI Z 24.22/1957 [ANSI, 1957]. Esta norma fue revisada y se publicó la ANSI S3.19/1974 (y ASA STD 1/1975), que a su vez fue revisada nuevamente con la publicación de la ANSI S12.6-1984, similar a las normas ASA 55-1984 e ISO 4869-1990 y recientemente la ANSI S12.6-1997 [ANSI, 1974; ANSI 1984; ANSI 1997].

La tabla 1 muestra la comparación entre las cuatro normativas para el método subjetivo. La tabla 2 muestra las diferencias básicas entre las normativas ANSI e ISO para el método objetivo.

Tabla 1. Comparación entre las cuatro normativas para el método subjetivo.

Especificação	ANSI Z 24.22	ANSI S 3.19-1974	ANSI S 12.6-1984	ANSI 12.6-1997(B)
Excitação da sala	Tons puros	bandas 1/3	bandas 1/3	bandas 1/3
Tipo de sala	anecóica	reverberante	T <1,6 (seg.)	T ≤1,6 (seg.)
Número de auto-falantes	um	3 mínimo	não especificado	não especificado
Ruído de fundo 0,125/0,25/0,50/, 02,0/4,0/8,0/ kHz	34/25/16 12/10/08/22	24/18/16 16/14/9/30	28/18/14 14/8/9/20	28/18,5/14,5/ 14/8,5/9/20,5
Número de pessoas	10	10	10	20 para plug 10 para concha
Repetição	3	3	3	2
Critério de colocação	colocado pelo indivíduo ensaiado para melhor atenuação	colocado pelo indivíduo ensaiado ou executor de ensaio	colocado pelo indivíduo ensaiado e verificado por exec. de ensaio	colocado pelo indivíduo ensaiado sem interferência
Movimento da cabeça após colocação e antes do teste	sim	não	não	não
Seleção e exclusão dos indivíduos	seleção aleatória	colocação não adequada deve ser relatada, mas não incluída na avaliação		colocação não adequada deve ser incluída

Tabla 2. Diferencias básicas entre las normativas ANSI e ISO para el método objetivo.

Especificação	ANSI S3.19	ISO 4869-3
Nível de pressão sonora mínima NPS (dB)	85 dB	75 dB: 63-250 Hz 90 dB: 315-4 KHz 85 dB: 5-8 KHz
Dureza da pele artificial	dureza shoore 00 20 + 5	dureza 30-85 IRHD
Microfone de medições	microfone de pressão	microfone de pressão
Características da sala de teste	campo sonoro com incidência aleatória	campo sonoro com incidência aleatória
Isolamento acústico	60 dB por cada banda	50 dB: 63-250 Hz 65 dB: 315-4 KHz 55 dB: 5 - 8 KHz
Microfone para avaliação do campo acústico	microfone direcional	microfone direcional ou microfone omnidireccional
Ruído de fundo RF	NPS - RF ≥ 60	NPS - RF ≥ 10 NPS e RF medidos com protetor
Tipo de campo acústico	ruído de banda larga filtrado em 1/3 de oitava	ruido rosa filtrado em 1/3 de oitava

La selección del método de ensayo de la normativa ANSI S12.6-1997, ya sea el método A o B, está basada en la aplicación deseada.

El procedimiento descrito por el método A, corresponde, aproximadamente, a los procedimientos usados en las normativas ANSI S12.6-1984, ANSI S3.19-1974 e ISO 4869-1. Tales resultados son útiles para diseñar protectores auditivos, para lograr un conocimiento teórico de las limitaciones de su desempeño y de las pruebas rutinarias de calidad.

El procedimiento descrito por el método B tiene por objetivo conseguir una aproximación de los límites máximos de atenuación en el mundo real, que pueden ser esperables para grupos de usuarios expuestos a ruido ocupacional.

Los individuos debidamente entrenados y motivados, pueden obtener mayor atenuación que con el método B, acercándose a los resultados obtenidos con el método A, especialmente para los protectores tipo orejeras. Se ha demostrado que los valores de atenuación obtenidos por el método B se aproximan más al desempeño real de los protectores en terreno.

Independientemente del método de prueba usado, ya sea con la colocación bajo supervisión (A) o por el auditor (B), los valores de atenuación serán aplicables, en general, en la medida que:

(1) Los protectores sean, en la práctica, colocados de la misma forma que durante la prueba de laboratorio.
(2) Se realice correctamente el mantenimiento de los protectores.
(3) Las características anatómicas de los individuos involucrados en los ensayos de laboratorio corresponda, aproximadamente, a las de la población de los auditores en el ensayo.

La norma ANSI S3.19-74, en su contexto, describe un método objetivo usando una cabeza artificial, eliminando la necesidad del uso de personas para la realización de los ensayos y entregando resultados más repetibles. Los resultados son comparativos y también son usados para el diseño del protector y el control de calidad. En el método subjetivo, las atenuaciones medidas llegan más cerca de los valores reales que en el método objetivo, pues toman en consideración el camino de transmisión vía ósea. Sin embargo, es un método lento, caro y requiere de una cámara especial con $T < 1,6$ s. La normativa ISO 4869-3 usa el método objetivo y es similar a la norma ANSI S3.19 (método objetivo). La figura 3 muestra las dimensiones normalizadas de la cabeza artificial, como se especifica en la ANSI S3.19.

Desde el desarrollo de la ANSI Z24.22-1957, los métodos de atenuación del oído real en el umbral de audición (*Real-Ear Attenuation at Threshold* – REAT), para la medición de la atenuación de ruido de los protectores auditivos, han sido ampliamente discutidos en la literatura. Además, se han desarrollado y probado muchos procedimientos adicionales, en algunos casos normalizados. Sin embargo, el método de atenuación de un oído real es el más aceptado y utilizado en la práctica. Generalmente, se acepta que los datos de REAT son los que mejor representan la medida de la reducción del ruido de los dispositivos convencionales, con independencia de los niveles usados (protectores lineales) y para una condición de prueba específica, esto es, para los individuos que fueron ensayados y para la forma en que los protectores fueron colocados. Los valores de las incertezas en el ensayo de protectores auditivos de acuerdo a las normas.

5 Números simplificados (únicos) para la atenuación de los protectores

Una transformación de los datos de atenuación media a un número único (baremo), posibilita una manera cuantitativa simple, eficiente y rápida para comparar y seleccionar los protectores.

Los fabricantes e importadores de los protectores auditivos son los responsables de suministrar tres importantes informaciones para los compradores y usuarios:

(1) La atenuación media del ruido en dB (en este caso es el mismo en dB(A)) del protector auditivo, en función de la frecuencia en las bandas de 1/1 octava entre 125 Hz y 8 kHz (siete valores);

(2) La desviación estándar para las mismas frecuencias del punto (1);

(3) Un número simple de la atenuación global, tal como NRR, NRRsf, SNR o HML (un valor para NRR o NRRsf o SNR o tres valores para HML).

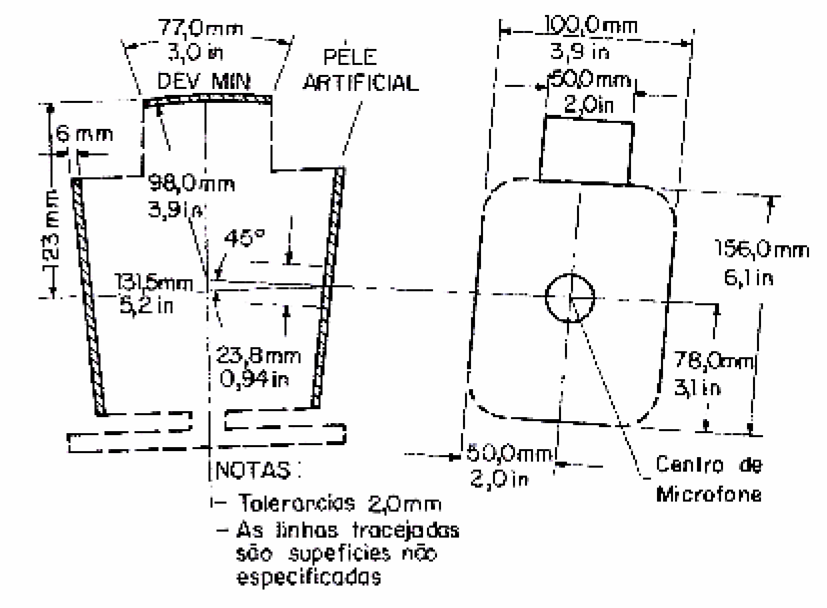

Figura 3. Las dimensiones normalizadas de la cabeza artificial.

Todos los índices mencionados (NRR, NRRsf, SNR y HML), corresponden a los datos reducidos de las informaciones de atenuación media y la desviación estándar en siete bandas de frecuencia (14 valores).

Los cálculos de los índices se basan en un espectro de ruido rosa como ambiente patrón, que no es el caso para todos los usuarios. Por lo tanto, estos índices no pueden ser usados para el cálculo preciso del nivel de presión sonora en el oído con protector, en otro ambiente que no sea el de ruido rosa.

6 Procedimiento para valorar la eficiencia de los protectores auditivos en los ambientes laborales

El nivel de presión sonora en el oído protegido (esto es, con la colocación del protector auditivo), puede ser valorado por varios procedimientos. Existen dos métodos principales: el método largo y los métodos simplificados. Siempre que sea posible, debe usarse el método largo.

6.1 Método largo

Este método permite la obtención de los niveles de presión sonora en dBA en bandas de frecuencias entre los 125 Hz y los 8 kHz, el nivel total en el oído protegido y la atenuación total suministrada por el protector en este ambiente laboral.

El nivel de presión sonora esperado para el oído protegido, podrá determinarse confrontando los datos del protector auditivo con los niveles del ruido para cada banda de frecuencia. El ejemplo que se mostrará a continuación, ilustra la metodología de cálculo para la determinación de los niveles de presión. Aquí se consideran bandas de 1/1 octava, cuyas frecuencias centrales, en Hz, son mostradas en la tabla 3. Los niveles de presión sonora, en dBA (para cada una de las bandas de 1/1 octava), a los cuales está sometido el usuario sin el uso de protector, se muestran en la primera fila de la tabla 3. La suma de estos valores es de 109 dB(A) y corresponde al nivel total al cual el individuo está expuesto, antes de colocarse el protector.

Tabla 3. La metodología de cálculo para la determinación de los niveles de presión para el oído protegido.

Frecuencia (Hz)	125	250	500	1000	2000	4000	8000
1- NPS Ambiente	83,9	93,4	101,8	106	102,2	97	88,9
2- Atenuación media	14	19	31	36	37	48	40
3- Desviación estándar (σ)	5	6	6	7	7	7	8
4- Atenuación - 2σ	4	7	19	22	23	34	24
5- NPS con protector (1-4)	79,9	86,4	82,8	84	79,2	63,0	64
Todos los niveles son en dB(A).							

Se considera para las bandas de 1/1 octava adoptadas, las atenuaciones medias, en dB, del protector utilizado y las respectivas desviaciones estándar (datos entregados por el fabricante y obtenidos por medio de un ensayo normalizado en un laboratorio acreditado), como se muestra en las filas 2 y 3 de la tabla 3. Se determina los límites inferiores de atenuación del protector, restando dos desviaciones estándar de la atenuación media, de acuerdo a lo descrito anteriormente, para una confiabilidad del 98 %.

Restando la fila 4 de la fila 1 en la tabla 3, para cada banda de 1/1 octava, se obtienen los niveles de presión sonora a las que el individuo estará sometido con el uso del protector, para cada banda de frecuencia (ver fila 5). La suma logarítmica de estos valores es 90,3 dBA y corresponde al nivel de presión sonora total al cual el individuo estará sometido después de la colocación del protector. La diferencia entre los niveles totales obtenidos antes y después de la colocación del protector es 109 dB – 90,3 dB = 18,7 dB y corresponde, para esta situación específica, a la atenuación esperada del protector utilizado, con una confianza del 98 %.

6.2 Métodos simplificados

Existen varios números únicos (baremos) para representar los datos de atenuación del protector auditivo. A continuación, se presentará el cálculo del nivel de presión sonora en el oído protegido de acuerdo con los principales números usados a nivel internacional.

6.2.1 Nivel de Reducción de Ruido (NRR)

El NRR reducido se puede usar para el cálculo del nivel de presión sonora protegido NPSc (dBA), con el uso del protector auditivo, sometido a un NPSs (dBC) o NPSs (dBA) en el ambiente, de acuerdo a las siguientes ecuaciones (1) y (2):

$$NPS_c\,(\text{dBA}) = NPS_s\,(\text{dBC}) - NRR \text{ , o} \qquad (1)$$

$$NPS_c\,(\text{dBA}) = NPS_s\,(\text{dBC}) - (NRR - 7) \text{ ,} \qquad (2)$$

El factor de 7 dB en la ecuación anterior corresponde a la diferencia entre la escala en dBA y dBC, además de otros factores.

6.2.2 Nivel de Reducción de Ruido (*NRRsf*)

El NRRsf es usado para el cálculo del nivel de presión sonora protegido NPSc (dBA), con el uso del protector auditivo, sometido a un NPSs (dBC) o NPSs(dBA) en el ambiente, de acuerdo a las siguientes ecuaciones (3) y (4):

$$NPS_c\,(\text{dBA}) = NPS_s\,(\text{dBA}) - NRRsf \text{ , o} \qquad (3)$$

$$NPS_c\,(\text{dBA}) = NPS_s\,(\text{dBC}) - (NRRsf + 5) \text{ ,} \qquad (4)$$

6.2.3 Número Único de Valoración (*SNR*)

El SNR puede ser usado para el cálculo del nivel de presión sonora protegido, por medio de la ecuación (5):

$$NPS_{\text{protegido}}\ (\text{dBA}) \ = \ NPS_{\text{en el ambiente}}\ (\text{dBC}) \ - \ SNR_{84} \text{ ,} \qquad (5)$$

El nivel protegido calculado por la ecuación anterior es menor que el nivel protegido para el 84% de los casos.

7 Ensayos mecánicos

La normativa europea EN 352-1/1993 explicita los ensayos mecánicos para los protectores tipo orejera, tales como: fuerza de contacto del arco, ensayo de caída libre, ensayo de baja temperatura (opcional), ensayo de flexión del arco, ensayo de baño de agua caliente, ensayo objetivo de atenuación (ANSI S3.19-74 y ISO 4869-3 y BS-EN 24-869-3).

8 El confort del usuario

Hasta el momento, el confort de los protectores auditivos no puede ser dado por una característica cuantitativa absoluta (como lo sería un índice de confort). Los parámetros tales como la masa, materiales, forma, presión de las almohadillas, fuerza del arco, ajuste, facilidad de colocación y remoción, pueden ser relevantes. Los requisitos para tales parámetros están definidos en las normativas europeas EN 352-1, EN 352-2 y EN 352-3, o en las normativas locales equivalentes [EN, 1993]. Siempre que sea posible, se debe permitir al usuario escoger el protector auditivo. Los compradores, supervisores, profesionales de las áreas de seguridad, salud e higiene ocupacional deben asegurar que se elija sólo entre los tipos de protectores adecuados.

Los parámetros más importantes para el confort del protector tipo orejera son:

(1) La distribución de presión en contacto entre protector y cabeza.
(2) Fuerza total.
(3) Centro de gravedad.
(4) Presión.
(5) Contacto en relación al área total de la orejera (almohada) del protector.
(6) Masa del protector.

9 Desempeño de los protectores en los ambientes industriales

En los últimos años, varias investigaciones han sido realizadas acerca de la atenuación de los protectores en ambientes industriales (mundo real), relacionando los valores reales con los obtenidos en condiciones ideales de laboratorio.

La atenuación medida en laboratorio es generalmente mayor que la atenuación lograda en terreno. Las razones de esto son:

a) Los auditores en el laboratorio se colocan los protectores para lograr la máxima atenuación, mientras que los trabajadores en terreno se colocan los protectores para lograr el máximo confort;

b) El auditor en el laboratorio está sentado cómodamente por poco tiempo, mientras que el trabajador en terreno ejecuta actividades físicas y movimientos durante la jornada laboral de, por ejemplo, 8 horas diarias;

c) El auditor está bien entrenado para el uso del protector, mientras que el trabajador, por lo general, no lo está;

d) El auditor escoge el tamaño adecuado de protector, colocándoselo y ajustándolo correctamente, mientras el trabajador ajusta el protector incorrectamente.

La atenuación, la desviación estándar y el NRR son válidos para el 98% de los usuarios (usando dos veces la desviación estándar), considerando las mismas condiciones de uso del protector (colocación, ajustes, etc.) en el laboratorio y en el terreno. La colocación y ajuste del protector en terreno es, generalmente, inadecuada y por lo tanto, se recomienda que los valores de NRR sean reducidos. El documento más reciente de la NIOSH, recomienda que los valores de NRR suministrados por los fabricantes (medidos en laboratorio con el método REAT, de acuerdo a la normativa ANSI S3.19-1974, que implica la colocación por el supervisor del ensayo), deban ser reducidos de acuerdo a los siguientes factores:

- Protector tipo orejeras: multiplicar el valor de NRR por 0,75 (75 %);

– Protector tipo inserción con materiales expandidos: multiplicar el valor de NRR por 0,5 (50 %);
– Otros protectores de inserción: multiplicar el valor de NRR por 0,3 (30 %).

Las reducciones anteriores no deben ser aplicadas en caso que el NRR esté basado en el ensayo con el método REAT siguiendo la nueva normativa ANSI S12.6-1997, en donde la colocación del protector es realizada por los auditores de la prueba. Por consiguiente, se recomienda abandonar el uso del NRR y cambiarlo por el NRRsf, que se basa en la normativa ANSI S12.6-1997.

10 Protector de inserción y orejera usados simultáneamente

Cuando la atenuación de un protector simple no es suficiente para la reducción de los altos niveles de ruido, el uso simultáneo de los protectores, de inserción y orejera, puede representar una solución que permita mayor atenuación.

La atenuación entregada con el uso simultáneo de dos protectores no puede ser obtenida a partir de la suma algebraica de las atenuaciones de cada uno, ya que se produce una interacción entre los dos tipos, a través del acoplamiento del aire en el canal y el acoplamiento mecánico a través del tejido del oído. La atenuación total será mayor que la mayor atenuación de ambos (por ejemplo: una pared doble entrega, como máximo, 6 dB de pérdida de transmisión sobre la suma de las pérdidas de cada una). No existe un método teórico o empírico para predecir la atenuación entregada por dos protectores usados simultáneamente.

11 Efecto del porcentaje de tiempo de uso

El uso constante del protector auditivo durante toda la jornada de trabajo es muy importante. La pérdida de audición está directamente relacionada al nivel equivalente L_{eq} en dB(A). Para un incremento de 3 dB en el nivel equivalente de exposición, el trabajador debe reducir el tiempo de exposición a la mitad (ver capítulo 2). La figura 15.23 muestra la atenuación real entregada por los protectores, en función del porcentaje del tiempo de uso. Por ejemplo, un trabajador usando constantemente un protector que entrega 20 dBA de atenuación, presentará una pérdida de atenuación de 3 dBA si este protector es usado sólo el 50% del tiempo. Este hecho puede ser mostrado a través del siguiente ejemplo: un trabajador está expuesto a un nivel de 100 dBA en una banda de frecuencia. Si usa un protector con atenuación de 20 dBA, cuando lo utiliza el tiempo completo, estará sometido a un nivel equivalente de 80 dBA. Por otro lado, si usa el protector sólo el 50 % del tiempo, el nivel equivalente estará dado por la ecuación (6):

$$L_{\mathrm{eq}} = 10\log\frac{10^{8}+10^{10}}{2} = 97\ \mathrm{dBA}, \qquad (6)$$

o, el protector entrega, en este caso, una atenuación de apenas

$$100\ \mathrm{dBA} - 97\ \mathrm{dBA} = 3\ \mathrm{dBA},$$

12 Problemas de utilización de los protectores auditivos

En la mayoría de las situaciones industriales, el confort y la durabilidad son factores más importantes que algunos decibeles adicionales de atenuación, considerándose que la atenuación ya alcanzada es razonable. Los diseños e instrucciones de uso de los protectores deben considerar los siguientes factores: higiene, falta de confort, efectos en la comunicación verbal, efecto en la localización direccional, señales de alarma, seguridad y costos.

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C04
Noise control by means of sound barriers

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Abstract: Sound barriers have been widely used in controlling the traffic noise produced by highways and other transportation noise sources. In general, a sound barrier is a solid object that interrupts either the direct or reflected sound path between a source and a receiver. Although several mitigation alternatives are available to reduce traffic noise, the use of sound barriers has become the most important mitigation measure in Environmental Impact Studies. In this conference, the fundamentals of noise control in highways and design aspects involved in the use of sound barriers will be presented. In particular, types of barriers, edge treatments for improving its insertion loss, prediction models, main standards, atmospheric effects, and some application examples will be reported. In addition, some details on the effect of buildings used as sound barriers will be presented. Finally, some results on the adverse effects and potential environmental impacts of sound barriers on both users and adjacent communities will also be discussed.

Resumen: Las barreras acústicas han sido ampliamente utilizadas para controlar el ruido de tráfico producido por las carreteras y por otros medios de transporte. En general, una barrera acústica es un objeto sólido que interrumpe el camino de transmisión del sonido directo o reflejado, entre una fuente y un observador. Aunque existen varias alternativas de mitigación para reducir el ruido de tráfico, el uso de las barreras acústicas se ha convertido en la medida de mitigación más importante en los estudios de impacto ambiental. Esta conferencia presentará los principios fundamentales del control de ruido en carreteras y los principales aspectos del diseño de las barreras acústicas. En particular, se presentarán los tipos de barreras, los tratamientos de sus cumbreras para mejorar su aislamiento, los modelos de predicción, las principales normativas en uso, los efectos atmosféricos y algunos ejemplos de aplicación. Además, se presentará algunos detalles sobre el efecto de los edificios usados como barreras acústicas. Finalmente, se discutirá también algunos resultados sobre los efectos adversos de las barreras y sus potenciales impactos ambientales, tanto en los usuarios como en las comunidades adyacentes.

1 Introduction[1]

A barrier is a device designed to reflect most of the sound energy incident back towards the source of sound. Today, the use of barriers to control noise problems is an example of a practical application of a complicated physical theory: the theory of diffraction, a physical phenomenon that corresponds to the non-specular reflection or scattering of sound waves by an object or boundary. Most of the theories of diffraction were originally invented for optics, but they find many applications in acoustics. In particular, noise barriers are a commonly used measure to reduce the high levels of environmental noise produced by the traffic on highways. For their proper use, aspects of design, economics, materials, construction details, aesthetic, and durability must be considered, in order to ensure good performance.

2 Attenuation of sound by barriers

As in the diffraction of light waves, when the sound reaches a listener by an indirect path over a barrier, there is a shadow zone and a bright zone, as shown in Fig. 1. However, the diffracted wave coming from the top edge of the barrier affects a small transition region close to the shadow zone by interfering with the direct wave (Kurze and Beranek, 1971).

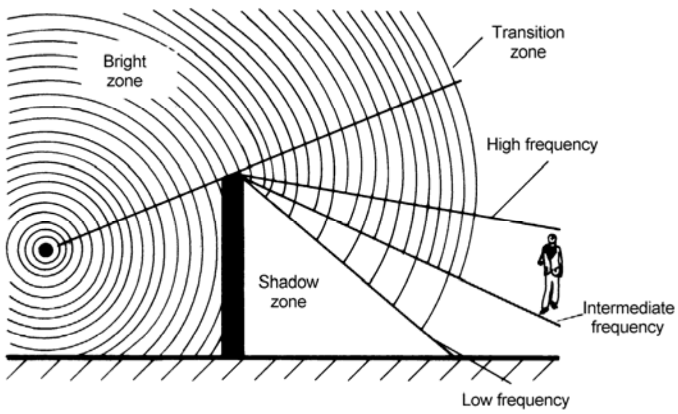

Figure 1. Diffraction of sound waves by a rigid barrier.

In a seminal work Keller (1957) proposed the geometric theory of diffraction (GTD) for barriers, which has been employed in the formulation of many different physical problems. Basically, he stated that from the set of diffracted sound rays from the barrier edge, the ray that reaches the reception point corresponds to the ray that satisfies Fermat's principle.

This geometrical theory of diffraction leads to relatively simple formulas, which combine the practicability of Kirchhoff's approximations with the greater accuracy of the Sommerfeld-type solutions and can be generalized to treat diffraction by three-dimensional objects of any smooth shape (Kurze, 1940). The geometrical situation is sketched in figure 2. For an infinite extended and very thin semi-plane, and assuming no reflections on the ground, the diffracted sound pressure amplitude is

$$\Psi = \frac{-1}{2\sqrt{2\pi k}} \frac{Q(\phi,\phi_S)}{\sqrt{r r_S (r + r_S)}} \exp\{i(k[r + r_S] + \pi / 4)\}, \qquad (1)$$

where r_s is the distance between the source and the top of the plane, r is the distance between the top of the plane and the reception point, k is the free-field wave number, and

$$Q(\phi,\phi_S) = \frac{1}{\cos\frac{1}{2}(\phi + \phi_S)} + \frac{1}{\cos\frac{1}{2}(\phi - \phi_S)}. \qquad (2)$$

[1] This article is based on previously published work by the author (Arenas, 2006; 2007a).

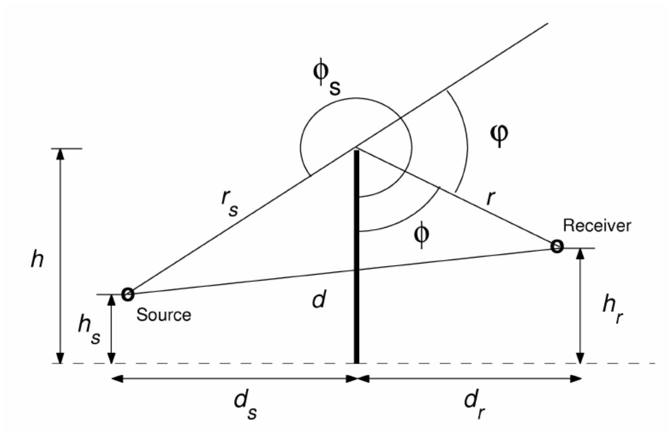

Figure 2. Geometry of a thin screen used in the diffraction theories.

The function $Q(\phi, \phi_s)$ implies that the edge of the plane radiates sound as a directional sound source. The sum in Eq. (2) corresponds to contributions from the source and its image due to reflection at the barrier. In addition, from equation (1) it is observed that for a fixed value of r_s, that satisfies the condition $kr_s > 1$, the amplitude of the sound wave is proportional to $1/\sqrt{r}$, for points located close to the diffracting edge. This implies cylindrical divergence, and thus a decay of 3 dB per doubling of distance. On the other hand, for points far away from the edge, the amplitude will have spherical divergence. Therefore the sound pressure level will decay 6 dB per doubling the distance. This important fact can also be obtained from the asymptotic analysis of the physical solution obtained by Macdonald (1915).

Certainly, from a practical point of view, most of the applications of the physical and geometrical theory had been difficult to use due to the complexity of the analysis, which does not permit fast calculation for design purposes. Because of this, several algorithms, charts and plots have been developed from time to time. One of the most well-known simplifications was a design chart proposed by Redfearn (1940). Since a rigorous solution of the diffraction problems involve several parameters in its formulation, it is clear that the approximations using the Redfearn's chart could involve large errors.

Kurze and Anderson (1971) reported a seminal study that presented one algorithm widely used today. This algorithm was obtained by comparing the experimental results of Rathe (1969) and Redfearn (1940) and the geometric theory of diffraction. In fact, their final equation can be derived from the Redfearn's parameter. In addition, Maekawa (1968) presented a chart based on the physical theory of diffraction and also numerous experimental results. His chart gave values of attenuation versus the dimensionless Fresnel number defined as

$$N = \pm\frac{2}{\lambda}\delta = \pm\frac{2}{\lambda}(r_S + r - d), \qquad (3)$$

where δ is called the path length difference. The ± is used to indicate the corresponding zone, such that N is positive in the shadow zone and negative in the bright zone.

For values of $N > 1$, Maekawa's result for insertion loss can be approximated by $13 + 10 \log N$. When $\varphi << 1$ and $d \to r + r_s$, the insertion loss can be approximated by $5 + 10\log 4\pi N$.

However, in order to find a more reliable expression for attenuation, Kurze and Anderson modified the results and obtained the analytical-empirical equation

$$IL_{KA} = 5 + 20\log\frac{\sqrt{2\pi N}}{\tanh\sqrt{2\pi N}}. \qquad (4)$$

Equation (4) gives good results in practice for N >0 and it shows good agreement with the experimental results obtained by Maekawa, for values of attenuation up to 24 dB. Equation (4) has been the starting point to define most of the barrier design algorithms used today to mitigate the impact of noise from highways.

2.1 Sound transmission loss of barriers

Barriers are a form of partial enclosure (they do not completely enclose the source or receiver) to reduce the direct sound field radiated in one direction only. The barrier edges diffract the sound waves but some waves can pass through the barrier according to the sound transmission laws. All the theories of diffraction have been developed assuming that the transmission loss of the barrier material is sufficiently large that transmission through the barrier can be ignored. Obviously, the heavier the barrier material, or the higher the frequency, the greater the transmission loss for sound going through the barrier.

A generally applicable acoustic requirement for a barrier material is to limit the component of sound passing through it to 10 dB less than the predicted noise level due to sound diffracted over the barrier. Evidently, this is not a governing criterion for concrete or masonry, but can be important for light aluminum, timber and for glazing panels. In addition, this may be an important consideration when designing “windows” in very tall barriers.

In a study on barriers used indoors, Warnock (1974) compared the transmitted sound through a barrier with the diffracted sound over the barrier. He found that the transmitted sound is negligible if the surface density of a single screen satisfies the criteria $\rho_S = 3\sqrt{\delta}$ kg/m^2. The minimum acceptable value of ρ_s corresponds to the transmission loss at 1000 Hz being 6 dB higher than the theoretical diffraction loss.

As a general rule, when the barrier surface density ρ_s exceeds 20 kg/m^2, the transmitted sound through the barrier can be ignored and then the diffraction sets the limit on the noise reduction that may be achieved.

According to the discussion above, when butting or overlapping components assembles a noise barrier, it is important that the joints are well sealed to prevent leakage. As an indication, it is common for timber barriers to be manufactured from 19 mm thick material. As indicated by the mass law, this provides a sound reduction index of 20 dB if joints are tight which is quite sufficient for barriers designed to provide an attenuation of 10 dB. In some countries, the legislation requires a sample of barrier to be tested in accordance with the local standard for sound insulation of partitions in buildings (Arenas, 2007a).

3 Barriers used outdoors

The use of barriers outdoors to control the noise from highways is surely the most well-known application of barriers. While noise barriers do not eliminate all highway traffic noise, they do reduce it substantially and improve the quality of life for people who live adjacent to busy highways. Noise barriers include walls, fences, earth berms, dense plantings, buildings, or combinations of them that interrupt the line-of-sight between source and observer. It appears that construction of barriers is the main alternative used for the reduction of noise, although quiet road surfaces, insulation of properties or use of tunnels have also been used for this purpose.

The theory discussed so far has been established for point or coherent line sources. However, the sound radiated from a highway is composed for several incoherent moving sources (vehicles of different types). It has been shown (Kurze and Beranek, 1971) that when a noise source approximates to an incoherent line source (stream of traffic) then the insertion loss is about 5 dB lower than the one calculated for a point source. From field results it has also been observed that earth berms (mounds of earth) produce about 3 dB more attenuation than walls of the same height. Then, predicted barrier attenuation values will always be approximations (Arenas, 2007b).

Attenuation other than resulting from wave divergence is called excess attenuation. Noise reduction due to a barrier is considered as a reduction to be added to other reductions due to such effects as spherical spreading, attenuation by absorption in the air, wind and temperature gradients, presence of grass and trees, etc. Therefore, it is common to refer to the excess attenuation by a barrier instead of insertion loss of barriers.

3.1 Finite extended barriers

If a barrier is finite in length (as the barriers used indoors) flanking (noise traveling around the ends of the barrier) will reduce the attenuation. In highway applications, it is recommended that the minimum angle of view that should be screened to avoid flanking is 160°. This means that to effectively reduce the noise coming around its ends, a barrier should be at least eight times as long as the distance from the home or receiver to the barrier.

For a barrier finite in length, parallel to a highway and located between the highway and the observer, as shown in figure 3, the approximate A-weighted attenuation in dB is given by

$$A = 10\log\left(\frac{1}{\beta_2-\beta_1}\int_{\beta_1}^{\beta_2} 10^{A(\beta)/10}\,d\beta\right), \tag{5}$$

where β is the angular position of the source from a perpendicular drawn from the observer to the highway, and $A(\beta)$ is a function of the Fresnel number when the source-to-observer path is perpendicular to the barrier, and a the berm correction (Barry and Reagan, 1978). We observe that, for an infinite barrier, $\beta_1 = -\pi/2$ and $\beta_2 = \pi/2$.

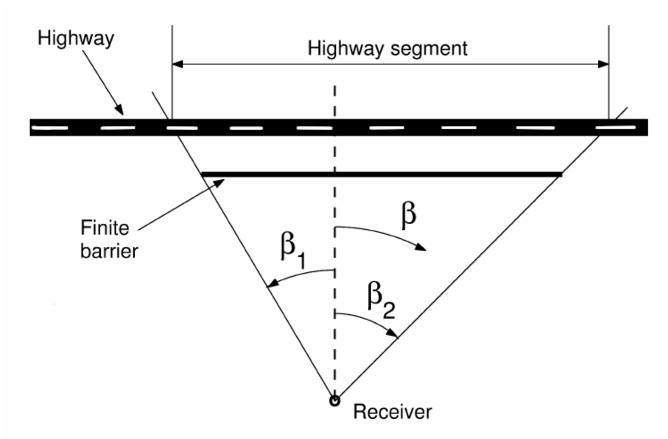

Figure 3. Top view of a finite barrier parallel to a highway.

The value calculated by equation (5) includes just the segment of incoherent source line that the receiver cannot see. Then, the contribution to the total noise of the segments not covered by the barrier should be estimated accordingly (Menge and Rossano, 1998).

It is possible to calculate equation (5) for each frequency band. However, to save time, an effective radiating frequency of 550 Hz is, in general, used as representative of a normalized A-weighted noise spectrum for traffic over 50 km/h. Then, the Fresnel number can be evaluated as $N = 3.21\times\delta$. Under this assumption, the A-weighted barrier attenuation in dB for an infinite freestanding wall, as a function of δ, is shown in figure 4 (Menge and Rossano, 1998).

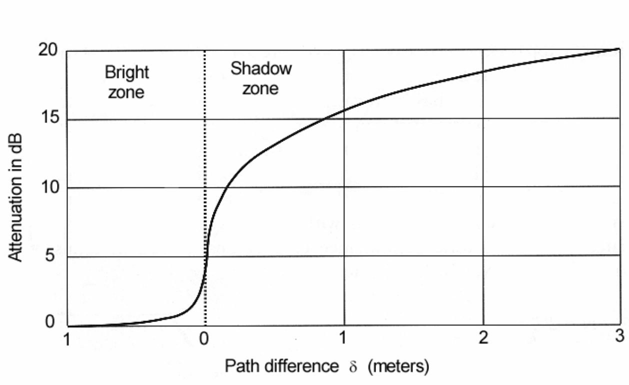

Figure 4. A-weighted attenuation for traffic noise as a function of δ.

3.2 Non-parallel barriers

For evaluating the attenuation of a barrier not parallel to an incoherent source line it is necessary to determine the equivalent path length difference (δ) that gives the effective source position (DTWO, 1988). The geometry is shown in figure 5. Firstly, a line bisecting the angle θ is drawn from the receiver point to the top edge of the barrier (point O). Then, a line is drawn from point O parallel to the source line to meet the vertical plane (i.e. normal to the road surface) which passes through the receiver point R and the effective source position S at B.

Finally, the equivalent path difference is calculated as δ = SB + BR – SR. The attenuation is then calculated for this equivalent δ.

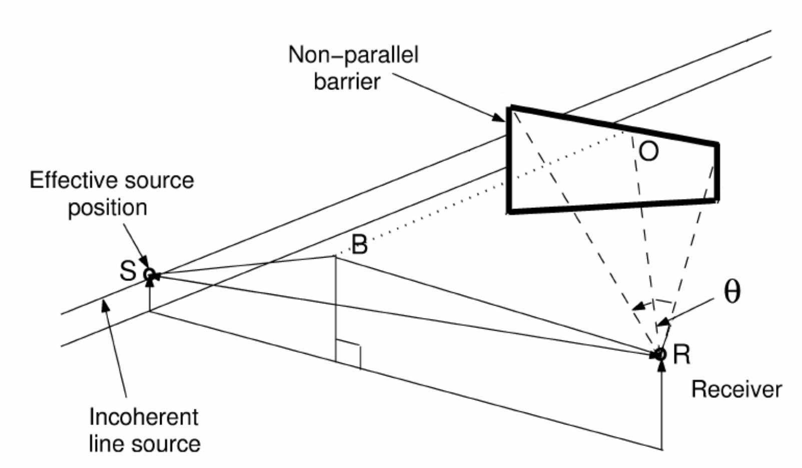

Figure 5. Geometry for a barrier that is not parallel to the source line.

Other reflections can affect the performance of a barrier, in particular when dealing with parallel barriers. This is the case when barriers are constructed on both sides of a road or when the road is depressed with vertical retaining walls. To overcome this problem of multiple reflections and insertion loss degradation, it is possible to 1) Increase the height of the barriers, 2) Use barriers with sound absorbing surfaces facing the traffic (a noise reduction coefficient NRC greater than 0.65 is recommended) or 3) Simply tilt the barriers outward where a tilt of 5^o to 15^o is usually recommended (Arenas, 2007b).

3.3 A building used as barrier

A barrier cannot always be treated as a very thin screen. An existing building can interrupt the line-of-sight between the source and a receiver acting as a thick barrier when its thickness is greater than the wavelength of the incident sound wave. Double diffraction at the two edges of a thick barrier may increase the attenuation. A simplified method to calculate the attenuation of a thick barrier is shown in figure 6. It is necessary to transform the thick barrier, of height H, into an equivalent thin barrier of height H', and then to calculate its attenuation using the usual equations.

A more accurate method has been proposed (Fujiwara et al, 1977). The effect of the double diffraction is to add an additional term to the attenuation due to a thin barrier. In figure 6 the line S'Y is parallel to the line SX. The attenuation of the thick barrier is

$$A = A_0 + K \log \frac{2\pi t}{\lambda}, \qquad (6)$$

where A_0 is the attenuation of a thin barrier for which δ = S'Y + YR – d' (see figure 6), t is the barrier thickness and K is a coefficient which can be estimated using figure 6 (right).

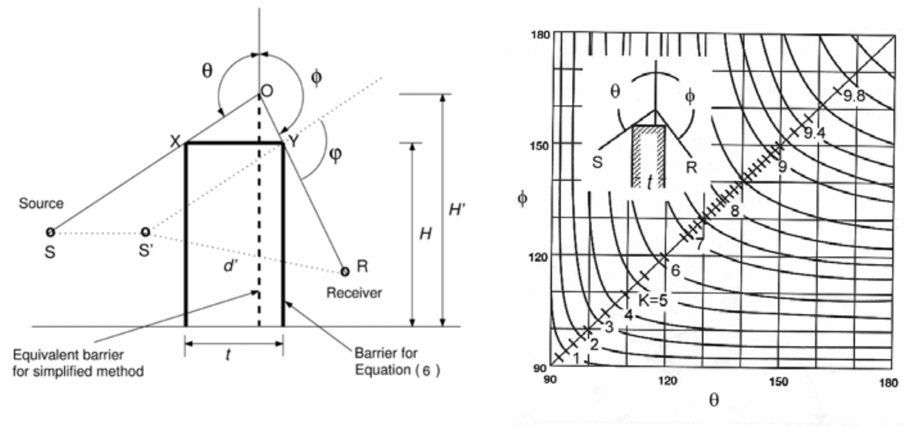

Figure 6. Geometry (left) and correction factor (right) for evaluating the attenuation of a thick barrier, such as a building.

3.4 Improvement of the barrier attenuation

In certain applications it may be necessary to enhance the attenuation provided by a single barrier without increasing its height. One example of this would be the need to increase a barrier's effectiveness without further reducing the view for residents living alongside a road that would be caused by use of a higher barrier. All the studies show that the use of some kind of element over the top of the barrier or the modification of its profile will change the original diffracted sound field (Arenas and Monsalve, 2001). In some cases this alternative can produce a significant improvement of the attenuation.

Theoretical and experimental studies on diffracting-edge modifications include T- and Y-shaped barriers, multiple-edge barriers, and tubular caps and interference devices placed on top of barriers (Fujiwara and Furuta, 1991; Iida et al, 1984; Moser and Volz, 1999). Full-scale tests of the acoustic performance of new designs of traffic noise barriers have been reported by Watts et al (1994).

Other options to improve the performance of a barrier are the use of modular forms of absorbing barriers (Fahy et al, 1995), absorbent edges (Rawlins, 1976), and by developing random profiles of different heights and widths, depending on the acoustic wavelength that has to be taken into account (Ho et al, 1997).

However, these alternatives are still under research and, sometimes, it is difficult to compare the results between different studies since the barrier heights, source position, receiver position and ground conditions are all different.

4 Design of sound barriers

In the design of a barrier all of the relevant environmental, engineering and safety requirements have to be considered. In addition to mitigate the impact of a highway, a barrier will become part of the landscape and neighborhoods. Therefore, some consideration has to be taken to assure a positive public reaction.

Both acoustic and landscape issues, to give guidance on good practice and design aspects, have been discussed widely in the technical literature (Kotzen and English, 1999, HA, 1994).

4.1 Materials and costs

A good design has to take into account that a barrier should require minimal maintenance other than cleaning or repair of damage for many years. Therefore, attention should be paid to the selection of materials used in the construction of barriers, in particular for areas subject to extreme weather conditions. Noise barriers can be constructed from earth, concrete, masonry, wood, metal, plastic, and other materials or combination of materials. A report showed that until 1998 most barriers built in the US have been made from concrete or masonry block, range from 3-5 m in height, and slightly more than one percent have been constructed with absorptive materials (USDT, 2000). Evidently, concrete or masonry walls require little or no maintenance during the service life, but transparent sections need frequent cleaning and might well need replacing after some time. The durability of sound absorbing materials for highway noise barriers has been discussed by some authors (Behar and May, 1980).

Often it is necessary to provide access from the protected side for maintenance purposes and for pedestrians or cyclists, which render a barrier vulnerable to vandalism. In addition, it may be advisable to avoid the use of flammable materials in some fire risk areas and, in general, it may be appropriate to install fire-breaks to limit the spread of fire (HA, 1995). When plants are selected for use in conjunction with a barrier they should generally be of hardy species (native plantings are preferable) which require a low level of maintenance.

A designer should seek detailed information for a specific project in order to estimate the cost of barrier construction and maintenance. This is particularly important when cost effectiveness is a must for positive decision on the construction of a barrier, since in some countries governmental agencies and individual homeowners sometimes share the costs of noise barriers.

Some additional aspects of the design of a barrier that need to be considered are the force caused by wind, aerodynamic forces caused by passing vehicles, the possibility of impact by errant vehicles, earthquakes, noise leaking through any gaps between elements or at the supports, and the effect of snow being thrown against the face of the barrier by clearing equipment (Arenas, 2006; 2007a).

4.2 Human response

The public, increasingly well-informed about the problem of excessive noise, is taking actions on the development of new transport infrastructure projects and improvement to existing infrastructure.

Most of the residents near a barrier seem to feel that highway noise barriers effectively reduce traffic noise and that the benefits of barriers far outweigh the disadvantages of barriers. Some studies have shown that public reaction to highway noise barriers appears to be positive (USDT, 2001). However, specific reactions vary widely. Residents adjacent to barriers have reported that conversations in households are easier, sleeping conditions are better, the environment is more relaxing, windows are opened more often, and yards are used more in the summer. In addition, residents perceived indirect benefits, such as increased privacy, cleaner air, improved views, a sense of ruralness, and healthier lawns. Negative reactions from residents have included a restriction of view, a feeling of confinement, a loss of air circulation, a loss of sunlight and lighting, and poor maintenance of the barrier (Arenas, 2008).

On the other hand, motorists have sometimes complained of a loss of view or scenic vistas and a feeling of being “walled in” when traveling adjacent to barriers.

High barriers substantially conceal the view of existing landmarks from the road, but they also conceal visual clutter, which might otherwise distract the attention of drivers. It is recommended that the appearance of barriers should be designed to avoid monotony. Surveys of drivers in Holland have indicated that a view which is unchanging for 30 s is monotonous (HA, 1994). This suggests that changes in design of barrier face every 800 m are desirable for long barriers adjacent to a high-speed road.

Noise barriers should reflect the character of their surroundings or the local neighborhood as much as possible to be acceptable to local residents. It is always recommended to preserve aesthetic views and scenic vistas. The visual character of noise barriers in relationship to their environmental setting should be carefully considered. For example, a tall barrier near a one-story, single family, detached residential area can have a negative visual effect. In general, it is recommended to locate a noise barrier approximately four times its height from residences and to provide landscaping near the barrier to avoid visual dominance and reduce visual impact (USDT, 2001).

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C05
Software libre y producción musical

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Abstract: At the beginning a brief introduction to the free software concept and to the use of different distributions of GNU\Linux will be given. A description will be made of the configurations that an operating system GNU\Linux needs for the job of musical production and for taking advantage of the hardware (real-time processing through the sound card). Also are mentioned the professional applications that are used to this assignment (audio work stations, sequencers, samplers, effect plug-ins, and virtual synthesizers), from the sound generation to the post-production and master edition, and and their operation is described. Finally, a musical performance putting in action all audio's applications will be given to show its optimum operation.

Resumen: En un principio se hará una pequeña introducción al concepto de software libre y al uso de distintas distribuciones GNU/Linux. Se describen las configuraciones que un sistema operativo informático GNU/Linux precisa para el trabajo de producción musical y el mejor aprovechamiento del hardware indicado (procesamiento en tiempo real de las placas de sonido). Además se nombra y describe el funcionamiento de cada una de las aplicaciones profesionales que se utilizan para este cometido, (estación de trabajo de audio, secuenciadores, samplers, PlugIns de efectos y sintetizadores virtuales), desde la generación de sonidos hasta la post-producción y masterización. Por último se realiza una ejecución musical poniendo en acción todas las aplicaciones de audio, para demostrar su funcionamiento óptimo.

1 Configurar GNU/Linux para producción musical

Cualquier distribución del Sistema Operativo GNU-Linux puede configurarse para trabajar con audio en tiempo real, con “tiempo real” me refiero a acceder a los procesos relacionados con el audio, sin latencia: un ejemplo es el tiempo o retraso que se da entre el disparo de una nota en un teclado controlador y la escucha del sonido producido por un sintetizador virtual. La latencia también se da cuando, por ejemplo, cantamos en un micrófono conectado a una computadora y la señal de audio atraviesa varios procesadores de efectos, los cuales realizan una infinidad de cálculos, hasta que llega a la salida de la placa de sonido. Tanto en Microsoft Windows como Mac OSX se precisan controladores especiales para aprovechar el proceso DSP de las placas de sonido profesionales y no obtener latencia, estos drivers se denominan ASIO (Audio Stream Input/Output). En GNU/Linux existe la ventaja de que no se precisan drivers ni controladores especiales sino que el sistema se configura para este propósito, desde el corazón mismo que se denomina: kernel, permitiendo así mayor control y rendimiento y no dependiendo de que los fabricantes de las placas de sonido, nos provean un driver ASIO.

Voy a tratar de explicar los pasos para configurar un sistema GNU/Linux optimizado para el trabajo con audio y MIDI, basándome en las distribuciones GNU/Linux basadas en Debian más conocidas, igualmente existen distribuciones armadas para esto, pero siempre es bueno saber como se hace a mano.

1.1 Paso 1: Instalar o compilar un kernel en tiempo real

La opción más fácil es descargar desde nuestra página oficial:

http://www.soundebian.com.ar/category/kernels-rt/

un kernel real time adecuado para el tipo de procesador que posea.

Luego para instalar el kernel necesitaremos tener instalado el paquete (o programa) *GDebi* simplemente con hacer doble clic sobre los archivos *.deb* descargados es suficiente. Si no tenemos GDebi, podemos abrir una consola, y con el comando *sudo* o como usuario *root*, podemos utilizar el comando dpkg:

 # dpkg -i *.deb

Que instala todos los paquetes .deb que se encuentran en el directorio. O bien

 # dpkg -i nombre_del_nuevo_kernel_image.deb nombre_del_nuevo_kernel_header.deb

que instala sólo los archivos especificados.

Para compilar nuestro propio kernel las instrucciones nuevamente están en nuestra página:

http://www.soundebian.com.ar/2009/10/como-compilar-nuestro-propio-kernel-rt-en-deb/

1.2 Paso 2: Configurar el archivo limits.conf

Luego lo más importante es editar el archivo /etc/security/limits.conf (para lo que se puede usar como superusuario el editor de textos gedit, nano, mcedit o cualquier otro disponible), otra vez en la terminal escribimos:

sudo gedit /etc/security/limits.conf

y agregamos al final del archivo estas líneas:

@audio - rtprio 99
@audio - nice -10
@audio - memlock 4000000

Guardamos el archivo y debemos agregar nuestro usuario al grupo audio:

 # adduser usuario **audio**

por último reiniciamos la máquina, si no editamos este archivo nunca nos funcionará JACK que es el servidor de audio en tiempo real.

Explicación de la configuración del archivo limits.conf: El valor **rtprio** es la máxima prioridad con la que un usuario del grupo audio puede ejecutar una tarea. El valor **memlock** es la máxima cantidad de memoria que un miembro del grupo audio puede bloquear para una tarea siendo ejecutada en tiempo real. Debería ser inferior a la máxima cantidad de memoria física; siendo una recomendación común establecerla a la mitad exacta, pero en algunos casos se ha usado el valor "unlimited" sin presentar problemas quedando la línea de configuración de esta manera:

@audio - memlock unlimited

El valor **nice** es el mínimo con el que una tarea puede ser ejecutada; se trata de la predisposición de una tarea a liberar tiempo de CPU.

1.3 Paso 3: Identificar el chipset de nuestra placa de audio:

Para esto podemos usar el comando:

cat /proc/asound/cards

1.4 Paso 4: Establecer prioridades RT, o sea aprovechar el kernel real time (Asignar las prioridades correspondientes al IRQ de nuestra placa de audio y de las tareas “timer”)

Este es un paso muy especializado para poder aprovechar al máximo las bondades del kernel realtime. Como es una explicación compleja y muy extensa se indica la dirección web donde consultarla. Sin este paso el sistema GNU/Linux igualmente funcionará a tiempo real con audio:

http://www.soundebian.com.ar/2009/10/configurando-verdaderamente-el-real-time-para-audio/

1.5 Paso 5: instalar y configurar JACK:

JACK es el sistema para la manipulación en tiempo real, audio de baja latencia (y MIDI). Funciona en GNU / Linux, Solaris, FreeBSD, OS X y Windows (y puede ser portado a otras plataformas POSIX conformes). Se puede conectar un número de diferentes aplicaciones en un dispositivo de audio, así como compartir audio entre ellas.

Para instalarlo escribir en una consola como usuario administrador o root:

#aptitude install qjackctl jackd

o bien se puede utilizar el gestor de paquetes Synaptic para instalar de manera gráfica las aplicaciones qjackctl y jackd, qjackctl es una interface gráfica de jackd muy útil y necesaria para conectar entre sí todas las aplicaciones y dispositivos de audio.

Por ultimo debemos configurar correctamente JACK, ejecutando qjackctl y activando en setup (o configuración si está en español) estas opciones:

Realtime (tiempo real) **activado**
No Memory Lock (no bloquear memoria) **activado**
Soft Mode (modo tolerante) **activado**
Force 16 bit (forzar 16bit) **activado** *(si se quiere trabajar en 16 bits)*

Priority (Prioridad)**=70**
Frames Períodos (Cuadros/Períodos)**=128** *(con esto se consegue 5.8 ms de latencia; se puede usar 256 también, si se quiere liberar más el cpu consiguiendo 11.4 ms de latencia —recomendado en pcs mas antiguas—)*
Sample Rate (Frecuencia de muestreo)**=44100**
Periods/Buffer (Períodos/Buffer)**=2**
Port Maximum (Puertos Máximos)**=128**
Timeout (Límite de tiempo)**= 5000 (ms)**
Interface= la placa que uses o default
Dither (Suavizado)**=ninguno**
Audio=duplex
Input Device (Dispositivo de entrada)**=(default)**
Output Device (Dispositivo de salida)**=(default)**
Input Chanels (Canales de Entrada)**=0**
Output Chanels (Canales de Salida)**=0**
Input Latency (Latencia de entrada)**=0**
Output Latency (Latencia de Salida)**=0**

1.6 Paso 6: Instalar aplicaciones

Se pueden instalar del gestor de paquetes synaptic o con el comando aptitude en una consola como root. Por ejemplo:

#aptitude install ardour

Algunas aplicaciones son:

ardour: La alternativa a Pro Tools, súper rápido, estable y muy profesional.
rosegarden: Secuencer Midi-Audio.
nted: Editor de Partituras.
qsynth: Sampler de Sound Fonts.
hydrogen: Sampler profesional y máquina de ritmos
audacity: Editor de audio multipista.
zynaddsubfx: Sintetizador virtual, con varios tipos de síntesis.
lmms: "Linux Multimidia Estudio" Alternativa para los usuarios de Fruity Loops, es un estudio de producción musical virtual que soporta VST.
seq24: Secuenciador midi, basado en loops, con una gran simplicidad de manejo .
tuxguitar: Editor profesional de tablaturas y partituras para bajo y guitarra. Es la alternativa a Guitar Pro e incluso abre el formato de archivo de este.
linuxsampler, qsampler y gigedit: Linuxsampler mediante la interface gráfica qsampler y la utilidad de edición gigedit. Nos permite usar samples de Gigastudio en GNU/Linux.
nmedit: Editor por software para GNU/Linux, para los afortunados poseedores de un Nord Lead Modular.
clam: plataforma para desarrollar con C++ con audio y música. Una herramienta infinitamente potente tanto para los compositores modernos como los ingenieros de sonido.

Plugins LADSPA
Muy buenos plugins LADSPA (los equivalentes a VST de GNU/Linux) para usar con cualquier secuencer o editor de audio son:

caps
tap-plugins
swh-plugins

RESÚMENES Y ENLACES A LOS TRABAJOS COMPLETOS

Sesión A1

Acústica de salas

Jueves 19/11/09 ~ 15:00-16:30

A034R - Techo equipotencial de un auditorio en abanico

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Abstract: First an algorithm is described to design an equipotential ceiling in a longitudinal section. Starting from a source F, an initial point A_1 on the ceiling and several receving points R_1, ..., R_n along the stalls, a reflecting plane is obtained such that FA_1 and A_1R_1 constitute an incident-reflected ray pair. Then the image source is obtained and the first ceiling panel is extended up to a point A_2 such that thes reflection from it reaches R_n. The process is repeated iteratively until the whole ceiling is designed. The preceding algorithm is applied to a fan-shaped auditorium so that the lack of lateral reflections is compensated by reflections from the ceiling. Since the source is not at the center of the fan, the ceiling has been divided into triangular segments and the algorithm aplied to each segment.

Resumen: Se expone en primer lugar un algoritmo para el cálculo de un techo equipotencial en un corte vertical plano. Partiendo de una fuente F, un punto generador A_1 en el techo y varios puntos receptores R_1, ..., R_n en la platea, se obtiene un plano tal que los segmentos FA_1 y A_1R_1 constituyan rayos incidente y reflejado respectivamente. Luego se obtiene la fuente imagen y se completa el primer panel de modo que la reflexión en el otro extremo, A_2, llegue hasta R_n. Estos pasos se iteran hasta completar todo el techo. El algoritmo así obtenido se aplica para la resolución del techo de una sala con forma de abanico, de manera de suplir la falta de reflexiones laterales que suele presentar esta forma. Debido a que el centro del abanico no coincide con la posición de la fuente, se ha subdividido el techo en gajos, y cada gajo tratado con el algoritmo anteriormente descrito.

A036R - El Teatro Avenida y su acústica

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Abstract: The Teatro Avenida was opened in 1908 (the same year as the Teatro Colón). Unfortunately it was partially destroyed by a fire in 1977. It was not rebuilt until 1994 when it was reopened. The author has been appointed as the acoustical consultant for this revival of the theatre. Its main hall was rebuilt according the original design which had been slowly modified through the years, but with new equipments. An auditorium for 120 people was built under the main floor; a restaurant was also built next to the hall. All this produced new sources of noise so it was necessary not only to secure good acoustics in the hall but also a low background noise level. The horseshoe hall has a main floor and four levels of boxes for about 1300 people. It is in a classic decorated style with ornaments and a huge central chandelier. Acoustical measurements were not made shortly after the reopening, but only this year (2009). The hall, its architecture and acoustic conditions are shown and described in this work. This theatre is one of the theatres around the world included in the “Acoustical Design of Theatres for Drama Performance 1985-2010” which will be issued in 2010 by the Acoustical Society of America (ASA).

Resumen: El Teatro Avenida se inauguró en 1908 (el mismo año que el Colón), pero fue parcialmente destruido por un incendio en 1977, permaneciendo abandonado hasta su reapertura en 1994, contando con este autor como su asesor acústico. La sala fue reconstruida respetando el diseño original, que había sido paulatinamente modificado, pero con la incorporación de los adelantos técnicos necesarios. Se incluyó una confitería adyacente y un auditorio para 120 personas ubicado por debajo de la platea y comunicado por su sistema de ventilación. Esto significó nuevas fuentes de ruido, lo que obligó no sólo a contemplar la buena acústica de la sala, sino también a mantener un bajo nivel de ruido de fondo. La sala es del tipo “herradura” con platea y cuatro niveles de palcos, con una capacidad de alrededor de 1300 espectadores. Es de estilo clásico, con ornamentaciones y una gran araña central. No se efectuaron mediciones acústicas

luego de su reapertura, las que recién se efectuaron este año 2009. En la presentación se muestra la sala, se describe su arquitectura y las condiciones acústicas. Esta sala se incluye en el libro "Acoustical Design of Theatres for Drama Performance 1985-2010" que editará la Acoustical Society of America (ASA) en 2010.

A046R - Estudio acústico del Teatro Municipal Coliseo Podestá de la ciudad de La Plata

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Abstract: The "Politeama Olimpo" Theatre was opened on November 19, 1886 and nowadays it is called "Coliseo Podestá". The Coliseo Podestá Municipal Theatre, whose project is atributed to the Uruguayan architect Carlos Zenhdorf, belongs to the Italian theatre type with its plant in horseshoe form. In 1981, the city of La Plata recovers the building and in 1986 the definitive re-opening takes place. This paper describes the acoustic analysis of the present state of the hall carried out by researchers of the Fine Arts Faculty, National University of La Plata. The activity program included the compilation of the existing documentation, the study of the building, the preparation of the graphical material and the measurements of the acoustic field and noise levels according to the ISO 3382 Standard. The conclusions show the particular characteristics of the Theatre based on its hybrid origin.

Resumen: El 19 de noviembre de 1886 se inauguró el Teatro "Politeama Olimpo" llamado en la actualidad "Coliseo Podestá". El proyecto se le atribuye al arquitecto uruguayo. El Teatro Municipal Coliseo Podestá corresponde a la tipología de teatro italiano con planta en forma de herradura. En el año 1897 la compañía circense Scotti - Podestá compra el teatro, que tenía la capacidad de funcionar como teatro-circo. En 1920, el primitivo edificio es remodelado para convertirse en una sala de prosa. En esta nueva etapa, en la sala, se realizaron bailes de carnaval y primavera y se ofrecieron espectáculos de distintos géneros. En el año 1981 la ciudad de La Plata recuperó el edificio y el 19 de noviembre de 1986 se produjo su reapertura definitiva. El presente trabajo describe el estudio acústico del estado actual de la sala realizado por investigadores de la Facultad de Bellas Artes de la Universidad Nacional de La Plata. El programa de actividades incluyó la recopilación de la documentación existente, el relevamiento en obra, la confección del material gráfico necesario para la investigación y la medición de niveles de ruido y de campo acústico según norma ISO 3382. Las conclusiones ponen de manifiesto el carácter particular que presenta la sala en función de su origen híbrido.

A050R - Relevamiento y valoración de parámetros acústicos en dos auditorios argentinos modernos

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Abstract: In this research parameters derived from the reverberation time (RT), that define various aspects of room acoustics, are analyzed. In-situ measurements have been made in two modern Argentine auditoriums: a class and lecture room, and an auditorium for chamber music and prayer ceremonies. Signals were acquired and processed by specific software. Then, the values obtained for each parameter, have been analyzed and compared with the recommendations proposed by acousticians. Several conclusions about the relationship between the constructive characteristics of the enclosure and the measured values are discussed.

Resumen: A lo largo de esta investigación se analizan parámetros derivados del tiempo de reverberación (RT) que definen varios aspectos de la acústica de salas. Para ello se realizan mediciones *in situ* **en** dos auditorios argentinos modernos: una sala de conferencias y clases, y un auditorio para interpretación de música de cámara y ceremonias religiosas. Las señales tomadas son procesadas mediante software. Luego, los valores obtenidos de cada parámetro son analizados y comparados con las recomendaciones propuestas por profesionales de la acústica. De esta forma se deducen conclusiones sobre la relación entre las características constructivas del recinto y los valores obtenidos.

A001R - Medición y análisis de parámetros acústicos en diez salas de concierto y teatros de Bogotá

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Abstract: The acoustical parameters of rooms are defined by equations or algorithms which describe the relation between objective measurements and subjective evaluations about the acoustical behaviour of a hall. Measurements are carried out according to the ISO 3382 methodology. During the XX century investigations had been made to define the optimal values for different acoustical parameters according to the use of the hall. In this work project measurements were taken in ten theatres and concert halls in Bogotá, Colombia with the aim of determining their acoustical parameters and suggesting the best use for them.

Resumen: Los parámetros acústicos de salas se definen mediante ecuaciones o algoritmos matemáticos que permiten relacionar medidas objetivas con evaluaciones subjetivas respecto al comportamiento acústico de un recinto por medio de mediciones cuya metodología se describe por la norma ISO 3382. A lo largo del siglo XX se han llevado a cabo estudios que han definido valores óptimos de los diferentes parámetros acústicos de acuerdo al empleo que se dé a una sala. Con base en lo anterior se realizaron mediciones en diez diferentes salas de la ciudad de Bogotá, Colombia, con el fin de establecer sus parámetros acústicos y definir el mejor uso que a cada una de ellas se puede dar.

A010R - Software para el cálculo de aislamiento acústico de sistemas constructivos

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Abstract: The software for the calculation of sound insulation of airborne noise, is a prediction tool based on calculation methods for systems construction such as walls of single, double and compound, in set its calculation takes some physical and mechanical properties of materials and gives results of system transmission loss constructive in terms of providing noise insulation. It is a very useful tool for professionals dedicated to acoustics, because mount a camera have brings high-cost. Thus simplified calculation method for sound insulation implemented through a software algorithm to be a tool to study this behavior and thus be able to carry out reforms obtain the desired system partition, saving time and money. First the software would emphasize basing on industry Colombiana basing its calculation method entirely local materials, differentiated of software that accentuate its calculation in estates physique mechanics of European and North American materials

Resumen: El software para el cálculo del aislamiento acústico de ruido aéreo, es una herramienta de predicción basada en métodos de cálculo para sistemas constructivos, tales como paredes simples, dobles y compuestas, en conjunto su cálculo toma algunas propiedades físico-mecánicas de materiales y arroja resultados de la pérdida por transmisión del sistema constructivo en cuanto al aislamiento acústico que provee. Es una herramienta bastante útil para profesionales dedicados a la acústica, ya que el montar una cámara de trasmisión trae costos altos. De esta forma un método de cálculo simplificado para el aislamiento acústico implementado por medio de un algoritmo a un software será una herramienta para estudiar este comportamiento y así poder realizar reformas hasta obtener el sistema deseado de la partición, ahorrando tiempo y dinero. Primero el software se enfatizará a la industria Colombiana basando en materiales enteramente locales, diferenciados de software que acentúan su cálculo en propiedades físico mecánicas de materiales europeos y norteamericanos.

Sesión B1

Ruido I

Jueves 19/11/09 ~ 15:00-16:30

A037R - Caracterización de la flota de transporte pesado de la República Oriental del Uruguay desde el punto de vista de sus emisiones acústicas

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Abstract: This paper is based on the activities of the Standardization Group in Acoustics (GESTA Acústico) in 2006 – 2007. In this period, GESTA Acústico has worked in order to regulate the Acoustic Pollution Act No. 17852. This Act has been passed on December 10, 2004. GESTA Acústico decided to set a base line of sound emission from transport in Uruguay, before making a proposal of emission standards for vehicles. There were no available data at the beginning of the work. The Ministry of Transport and Public Infrastructure, as a member of the GESTA Acústico has made the efforts to develop a data base of sound levels from emissions of heavy vehicles measured during the required heavy vehicle inspections done by SUCTA Company. The aim of this paper is to present the processing of these data, taking account of vehicle age, place of registration (Montevideo or inland) and type of vehicle (trucks or buses). Results are compared with different municipal standards and with standards of MERCOSUR.

Resumen: Este trabajo se basa en las actividades desarrolladas por el Grupo de Estandarización en Acústica (GESTA Acústico) entre los años 2006 y 2007, con el objetivo de avanzar hacia la reglamentación de la Ley de Contaminación Acústica Nº 17852. Esta ley marco fue aprobada el 10 de diciembre de 2004. El grupo GESTA Acústico se planteó, entre sus actividades iniciales tendientes a reglamentar dicha ley en materia de emisiones de fuentes móviles, generar una línea de base de emisiones sonoras del transporte terrestre en el país, para luego tomar decisiones en relación con estándares adecuados para fuentes móviles. Al iniciar el trabajo no se contaba con antecedentes en el país en materia de información cuantitativa. El Ministerio de Transporte y Obras Públicas, integrante del GESTA Acústico, realizó las gestiones para generar, a través de la empresa SUCTA que tiene a su cargo el control vehicular de las unidades de transporte carretero, una base de datos de mediciones de ruido de escape y opacidad de emisiones de vehículos estacionados, datos tomados a la hora de la revisión obligatoria de los vehículos de transporte interdepartamental de cargas y pasajeros. El objetivo de este trabajo es presentar un procesamiento de los datos generados, considerando la edad de los vehículos y su lugar de empadronamiento (Montevideo o interior del país), y tratando por separado los vehículos de carga y de pasajeros. Los resultados se comparan con las normativas departamentales vigentes en Uruguay y con la normativa MERCOSUR.

A023R - Mapa de ruido de la Base Vicecomodoro Marambio-Antártida Argentina

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Abstract: This study presents a set of noise maps of the Vicecomodoro Marambio Base, Province of Tierra del Fuego, Antártida e Islas del Atlántico Sur, which belongs to the Argentine Air Force. The noise maps were obtained during various activities carried out within the area of the Vicecomodoro Marambio Base, covering a total area of approximately 194 400 m^2. Software simulation and prediction of noise levels were used to draw the maps.

Resumen: En este estudio se presenta un conjunto de mapas de ruido realizados en la Base Vicecomodoro Marambio, Provincia de Tierra del Fuego, Antártida e Islas del Atlántico Sur, dependiente de la Fuerza Aérea Argentina. Los mapas de ruido fueron relevados durante el desarrollo de distintas

actividades realizadas dentro del área comprendida por la Base Vicecomodoro Marambio, cubriendo una superficie total de aproximadamente 194 400 m^2. Para la realización y diseño se utilizaron además programas informáticos de simulación y predicción de niveles sonoros.

A012R - Mediciones comparativas de niveles de presión sonora a diferentes alturas en el ámbito urbano

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Abstract: The LACEAC has been measuring noise in Buenos Aires since 1970, year in which the first map was created. Newer records allowed to update and correlate values. Following old standards, the measurements were taken at a height of 1,5 m. Recent local legislation and Directives from the European Union set the height to 4 m. This communication compares the values from simultaneous measurements at both heights while emphasizing the weather corrections made by “in situ” conditions versus those taken from the Meteorological Service as an average for the urban areas.

Resumen: El LACEAC ha efectuado mediciones de nivel sonoro en Buenos Aires desde 1970, año de concreción del primer mapa de ruido de la ciudad. Los registros posteriores permitieron la actualización y correlación de valores. Siguiendo viejas normativas, la altura de medición era de 1,5 metros. La legislación local reciente y las directivas de la Unión Europea fijan en 4 metros esa altura. Esta comunicación compara valores a partir de medidas simultáneas a ambas alturas, a la vez que hace hincapié en las correcciones meteorológicas “in situ” versus las recogidas por el Servicio Meteorológico como promedio para el ámbito urbano.

Sesión A2
Acústica Musical

Jueves 19/11/09 ~ 17:40-19:00

A035R - Variación de la altura psicoacústica como medio de expresión en el piano: estudio exploratorio

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Abstract: Players of bowed and plucked string and wind instruments, as well as singers, use extensively the resource of varying subtly the frequency of sounds to enhance expressiveness (for instance, by means of vibrato). This is not possible in the case of the piano, since the musician cannot alter the string tension neither its length. However, the intensity variations due both to the envelope and to the diverse touch pressure applied to successive notes cause psychoacoustic pitch fluctuations, which could impact on the expressiveness. In this paper the likeliness of this idea is explored by computing, using models available in the literature, the pitch variations elicited by this mechanism in isolated notes played by professional pianists. Future works are planned using pianistic passages instead of an isolated note used in this first exploratory stage. Although not conclusive results have been achieved, a methodological framework for futher studies is shown. Particularly, an interactive platform of evaluation was developed for a procedure of experimentation.

Resumen: Tanto en los instrumentos de cuerda frotada o pellizcada y de viento como en el canto se utilizan fluctuaciones sutiles de frecuencia (por ejemplo vibratos) como medio para lograr una interpretación expresiva. Ello no es posible en el piano, dado que el intérprete no puede modificar ni la tensión ni la longitud de las cuerdas. Sin embargo, tanto las variaciones de intensidad contenidas en la envolvente del sonido como las que se pueden lograr entre notas sucesivas variando la presión del toque, provocan variaciones de altura psicoacústica que podrían tener un efecto expresivo. En este trabajo se explora esta hipótesis determinando, mediante los modelos de la literatura, las diferentes variaciones de altura que se generan en notas aisladas ejecutados por pianistas profesionales. Se proyectan trabajos futuros con pasajes pianísticos en lugar de una nota aislada usada en esta primera etapa exploratoria.

A052R - Análisis cualitativo y cuantitativo del timbre de aerófonos de madera

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Abstract: The distinctive timbre of handcrafted single reed wind instruments introduced in the last few decades is analyzed in this work. The specific case of two wooden straight cylinders, which simulate the underlying physical principles of the clarinet, is considered. This instruments´ timbre is very similar to that of the saxophone and the clarinet, and a parallelism is made with their acoustical principles. These two wind instruments were recorded, along with two traditional saxophones and a clarinet, under controlled conditions. Their timbre and spectrum was analyzed taking into account the harmonic presence and spectral content. Quantitative analysis has been made through the microscopic auto-correlation parameters of each recorded sample. In the research and recordings, their acoustics is proven to be analogous to both of the single reed instruments which can be found on any orchestra.

Resumen: En este trabajo se analiza el timbre característico que tienen los instrumentos de viento de lengüeta simple que han surgido en las últimas décadas, tomando como ejemplo específico dos cilindros rectos de madera, funcionamiento análogo al del clarinete. El timbre de estos instrumentos se asemeja al saxofón y al clarinete, y se hace una analogía con el funcionamiento acústico de ellos. Fueron grabados (junto a dos saxofones y un clarinete) bajo condiciones controladas. El análisis cualitativo se efectuó comparando el espectro y las componentes armónicas de ellos. El análisis cuantitativo se realizó utilizando los parámetros microscópicos de la función auto-correlación de cada muestra. En la investigación y las grabaciones se demuestra que su funcionamiento es análogo al de los instrumentos de viento de lengüeta simple que se pueden encontrar en una orquesta hoy en día.

A015R - Patrones de Chladni y factor de calidad para una tapa trasera de guitarra

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Abstract: The behaviour of an acoustic guitar back plate excited by sound waves was studied in the frequency range from 135 Hz to 1200 Hz. Chladni patterns formed on sand spread on the plate were observed and the quality factor at a resonance frequency was determined from the data obtained by an ad-hoc device. The Chladni patterns are the most direct method for the visualization of vibration modes on a surface. They are obtained by spreading a dry light granular substance (e.g., sand) over a vibrational surface. Vibrations cause the grains to accumulate on the nodes making visible their locations. In the first place the guitar back plate's excitation frequency corresponding to modes of osscillation was identified through Chladni patterns sharpness. In the second part of the work, the amplitude of vibration dependence of the excitation frecuency in a specific point of the plate was obtained by means of a home-made device, which consists in a phonograph's pick up connected to an oscilloscope. Finally, the back plate's quality factor arround 450 Hz was determined.

Resumen: Se relevó el comportamiento de una tapa trasera de guitarra estimulada por ondas sonoras de frecuencias en el rango [135 Hz, 1200 Hz]. Se observaron los patrones de Chladni formados por arena colocada sobre la tapa y se determinó el factor de calidad para una de las frecuencias de resonancia mediante un dispositivo ad hoc. Los patrones de Chladni son la forma más directa de visualizar los modos de vibración de una superficie. Éstos se obtienen esparciendo alguna sustancia granular fina (por ejemplo arena) en la superficie en cuestión. La vibración provoca que la sustancia se acumule en los nodos haciendo visible su ubicación. En la primera parte del experimento, la tapa trasera de una guitarra fue excitada a distintas frecuencias y se utilizaron los patrones de Chladni para identificar las resonancias. En la segunda parte se obtuvo la amplitud de vibración en función de la frecuencia mediante un dispositivo hogareño que consta de un pick-up de fonógrafo conectado a un osciloscopio. Finalmente, se determinó el factor de calidad alrededor de los 450 Hz.

A009R - Análisis acústico de un tiple colombiano por medio de la obtención de la respuesta al impulso con método directo

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Abstract: Musical Acoustics is an interesting field of study with important applications to the analysis of musical instruments performance. The knowledge of how that these instruments process a signal allows to obtain useful tools for optimization, record and DSP simulation. For stringed instruments, if one considers the input signal as the vibration produced by the strings and the output signal as the sound produced by the entire instrument, then one can consider the instrument body as the sound modifier or transfer function which describes the instrument performance or behaviour. An electro-mechanic transducer was designed in order to excite the tiple with uniform beats in an optimum frequency range, so that it generates an output signal which is recorded by a flat-response microphone. Then a deconvolution process is carried out to obtain the transfer function. This experiment was conducted in a room that simulates an anecoic chamber in order to avoid the sound reflections. The determination of the impulse response with this method called "direct", opens up many applications in the musical production field.

Resumen: La acústica musical es una interesante rama de estudio con importante aplicación en el análisis del comportamiento de instrumentos musicales. Conociendo la forma en que éstos procesan una señal, se pueden obtener útiles herramientas para su optimización, captura e incluso la simulación de éstos por medio de DSP. Para instrumentos de cuerda, si se considera como señal de entrada la vibración producida por las cuerdas tensadas y como señal de salida el sonido producido por todo el instrumento, se puede también considerar al cuerpo del instrumento como modificador de sonidos o función de transferencia que describe el comportamiento propio del instrumento. Para este experimento se diseñó un transductor que excita el tiple con golpes uniformes en un rango de frecuencias óptimo, de tal manera que se produce una señal de salida que es capturada por un micrófono de respuesta plana para luego realizar un proceso de deconvolución y obtener la función de transferencia. Esto fue hecho en un espacio que simula una cámara anecoica para evitar el efecto de reflexiones sobre la captura. La obtención de la respuesta al impulso con éste método denominado "directo", abre las puertas para múltiples aplicaciones en el campo de la producción musical.

A008R - Obtención de la respuesta al impulso del tiple colombiano por medio del método inverso

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Abstract: In this paper the impulse response of the Colombian tiple is obtained by an indirect method in which the musical instrument is excited using a sine sweep signal reproduced by a loudspeaker. Then the vibration present on different points of tiple surface is recorded by means of an accelerometer and an audio interface, and finally this signal is deconvolved with the sine sweep signal to obtain the impulse response of this instrument.

Resumen: En este artículo se obtiene la respuesta al impulso de un tiple colombiano por medio del método inverso, el cual consiste en excitar el instrumento musical con una señal *sine sweep* generada por un altavoz, para luego grabar la vibración presente en diferentes partes de la superficie del tiple usando un acelerómetro. Posteriormente se realiza la deconvolución de la señal grabada con la generada para obtener así la respuesta al impulso del instrumento.

A007R - Comparación entre el método directo e inverso para la obtención de la respuesta al impulso del tiple colombiano

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Abstract: In this paper two methods for obtaining the impulse response of a plucked string musical instrument, in this case to the Colombian national instrument, the tiple, are compared. The first one is the direct method, which applies an LSS signal on the instrument's body through an electro-mechanical transducer to excite and then capture the signal radiated by it. The second one is the inverse method, which applies a sound field generated by an appropriate sound source to the instrument. The vibrations generated by the sound field are then measured on the body of the instrument by an electro-mechanical transducer. Data from both methods are compared by calculation and spectral analysis.

Resumen: En este trabajo se presenta la comparación de dos métodos de obtención de la respuesta al impulso aplicados a un instrumento musical de cuerda pulsada, en este caso al instrumento nacional colombiano: el tiple. El primer análisis realizado corresponde al método directo, el cual excita con una señal de tipo LSS la caja del instrumento por medio de un transductor electro-mecánico y luego captura la señal irradiada por éste. El segundo análisis corresponde al método inverso, en el que se genera un campo sonoro con una fuente que baña de energía al instrumento y se capturan las vibraciones formadas por el campo sonoro sobre la caja del instrumento mediante un transductor mecano-eléctrico. Obteniendo estos datos se procede a realizar una comparación objetiva mediante cálculos y análisis espectral.

A006R - Mejora de la calidad sonora mediante la convolución de respuestas al impulso con grabaciones de audio, aplicado al tiple colombiano

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Abstract: Based on the experiments to obtain the impulse response of a high quality tiple by the inverse and direct methods, a tool is developed to improve the sound quality of recordings using the convolution between a regular tiple audio signal with the impulse response of our tiple. Once the signal is processed, an objective analysis is done to study its behaviour as a function of frequency (FFT) and possible variations in the ADSR envelope. Furthermore it is important for this application to make a subjective analysis which refers to a process of critical listening performed by musicians and sound engineers with an outstanding work in the Colombian music field.

Resumen: Con base en los experimentos referentes a la obtención de la respuesta al impulso de un tiple de alta calidad utilizando los métodos inverso y directo, se desarrolla una herramienta con el propósito de mejorar la calidad sonora de grabaciones mediante la convolución entre la señal de audio de un tiple de baja calidad con la respuesta al impulso de un tiple de alta calidad. Una vez obtenida la señal procesada, se realiza un análisis objetivo que consiste en estudiar su comportamiento en función de la frecuencia (FFT) y las posibles variaciones que surgen en el envolvente acústico. Otra tarea importante realizada es el análisis subjetivo, el cual consiste en un proceso de escucha crítica realizada por músicos e ingenieros de sonido con una importante trayectoria de trabajo en el campo musical colombiano.

Sesión B2
Acústica Arquitectónica

Jueves 19/11/09 ~ 17:40-19:00

A005R - Soluciones de Aislamiento acústico en edificios nuevos y reciclados con la utilización de lanas de vidrio

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Abstract: The aim of the present work is to show different technical constructive solutions for different acoustical problems. On the one hand, an integral project of acoustic insulation of a new construction is presented. This project's objective is to mitigate the negative environmental impact due to the noises that will be produced both during the construction stage and as a consequence of the activities that will be developed in the building after its inauguration. On the other hand, solutions to problems related to noises from discrete sources both in new buildings as in existing ones will be also described.

Resumen: El objetivo del presente trabajo es exponer diferentes soluciones técnico constructivas para distintos problemas acústicos. Por un lado se presenta un proyecto integral de aislamiento acústico de una obra nueva. Este proyecto tiene como finalidad mitigar el impacto ambiental negativo debido a los ruidos generados tanto en la etapa de construcción como en la actividad a desarrollar en el edificio luego de la inauguración. Por otro lado se describen soluciones para problemas relacionados con ruidos provenientes de fuentes puntuales, tanto en edificios nuevos como en existentes.

A029R - El diagnóstico en el proyecto de aislamiento acústico. Estudio de un caso particular

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Abstract: Dealing with issues related to improvements in the sound insulation of an enclosure presents certain difficulties in the stages previous to the solution design. An appropriate diagnosis allows delimiting the scope and extent of proposed actions. A case of acoustic insulation will be presented, showing the strengths and weaknesses of actions and the need to intensify the search for diagnostic tools that allow conforming an action plan within an overall conception of the situation.

Resumen: El abordaje de las cuestiones relacionadas a las mejoras en las condiciones del aislamiento acústico de un recinto presenta ciertas dificultades en las instancias previas al diseño de la solución. Un diagnóstico adecuado permite delimitar el campo de acción y los alcances de las intervenciones propuestas. Presentamos un caso concreto de aislamiento acústico, donde se muestran las debilidades y fortalezas de las intervenciones realizadas y la necesidad de profundizar en la búsqueda de herramientas de diagnóstico que permitan conformar un plan de acciones dentro de una concepción global de la situación.

A059R - Diseño acústico óptimo en recintos industriales basado en un modelo de difusión

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Abstract: Acoustic control is a discipline of growing interest in the industrial workplace. For the purpose of designing acoustic solutions, it is necessary to make predictions of the sound levels that will occur once implemented. The most simple and yet effective method for this is the classic Sabine model.

This theory provides an explicit analytical formula for noise levels depending on the geometric arrangement of the sources, their powers and the averaged absorption coefficient. Despite of its great success in acoustical engineering, several situations were identified for which their results are very imprecise. Another well established and more accurate technique in this field is the ray tracing method. However, computation times are often large. A new concept for the acoustic enclosure prediction was recently proposed (Picaut, 1997), that allows to calculate the acoustic field distribution in non-uniform reverberant enclosures, extending the Sabine theory. This approach is used for developing an optimal acoustic design method, along with mathematical optimization techniques.

Resumen: El control acústico es una disciplina de interés creciente en el ambiente laboral industrial. A los efectos de diseñar soluciones acústicas, es necesario efectuar predicciones de los niveles sonoros que tendrán lugar una vez implementadas las mismas. El método más simple y aún efectivo para realizar esta tarea es el modelo clásico de ruido difuso de Sabine. Esta teoría brinda una fórmula analítica explícita de los niveles acústicos en función de la disposición geométrica de las fuentes, sus potencias y el coeficiente de absorción promediado. A pesar de su gran éxito en la ingeniería acústica se han detectado varias situaciones para las cuales sus resultados son muy imprecisos. Otra técnica bien establecida y de mayor precisión en este campo es el método de trazado de rayos. Sin embargo, los tiempos de cómputo empleados suelen ser importantes. Un nuevo concepto para la acústica de recintos fue propuesto recientemente (Picaut, 1997), el cual permite calcular la distribución del campo reverberante no uniforme en recintos, extendiendo la teoría de campo difuso. Tal enfoque se utiliza para un método de diseño acústico óptimo, empleando técnicas de optimización matemática conjuntamente con el modelo aludido.

A060R - Un modelo de caja gris para la acústica de recintos industriales

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Abstract: Noise pollution due to industrial noise sources is a serious problem within industrial rooms. Because of the high cost involved in noise control systems, their design is an important aspect to consider. There are several computer models for this purpose, from simple analytical formulations to more complex methods based on geometric acoustics. An alternative approach to model the different relationships involved in such situations is the use of artificial neural networks (ANN). This tool provides a robust method for solving classification and prediction problems, allowing great flexibility and accuracy. One difficulty associated with this type of technique is the limited predictive extrapolation properties, which lead to erroneous results. This paper deals with an approach that combines the theoretical structure of a Sabine-type acoustic model corrected by an ANN in situations where it is known to present a considerable inaccuracy. This is an attempt to improve the properties of extrapolation.

Sesión A3
Ruido II

Viernes 20/11/09 ~ 08:50-10:30

A033R - Aplicación de la Norma IRAM 4062 para ruidos molestos de muy bajo nivel

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Abstract: Frequently there are complaints regarding noises that do not violate the local noise ordinance in spite of being indeed disturbing. In this paper an approach is introduced that may help circumventing this problem. The IRAM 4062 Standard, which uses a differential criterion to decide whether a noise is disturbing or not, is combined with non-quantitative criteria such as the one contained in the Civil Code of the Republic of Argentine, which prohibits the emission of any noise that exceeds normal tolerance.

Resumen: Se describe la aplicación de la Norma IRAM 4062 sobre ruidos molestos al vecindario en combinación con criterios no cuantitativos como el del artículo Nº 2618 del Código Civil de la República Argentina, referido a ruidos que superan la normal tolerancia, para el caso de ruidos de bajo nivel en un contexto de ruido ambiente de muy bajo nivel. De esta forma se aporta un método para objetivar un criterio aparentemente subjetivo mediante un criterio de decisión comparativo en lugar de absoluto, como el que se adopta en muchas ordenanzas municipales sobre ruido.

A016R - Propagación de bajas frecuencias: ¿Medición en dBA, dBC o espectral?

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Abstract: Recent development of natural gas in Peru has produced a growth in the installation of thermal power plants with large electrical generation turbines in excess of 100 MW, and small ones for individual industries, letting aside pumping stations, cryogenic compression plants, etc. The studies of environmental impact statement for those facilities underestimated or even ignored the effects in the low frequency range by deriving their conclusions from A-weighted measurements. This work reports the analysis of real cases, where all measurements have been made spectrally, with “A” and “C” weighting.

Resumen: El reciente desarrollo del gas natural en el Perú ha producido un crecimiento en la instalación de centrales térmicas con turbinas de generación eléctrica mayores a 100 MW y pequeñas usinas particulares para industrias, sin contar las estaciones de bombeo, de compresión, plantas criogénicas, etc. Los estudios de impacto ambiental realizados para esas instalaciones minimizaron o incluso ignoraron los efectos en las bajas frecuencias del espectro sonoro por haber estimado las conclusiones en mediciones con ponderación “A”. Este trabajo presenta análisis de casos reales, donde la totalidad de las mediciones fueron realizadas en el espectro, aplicando ponderación “A” y “C”.

A017R - Medición de ruido urbano: ¿Global o espectral? Resultados en más de diez ciudades del Perú

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Abstract: A Paper of the V Latin American Congress of Acoustics presented measurements of Lima centre in dBA. Since they were similar to each other the authors concluded that the environmental noise in Peruvian cities is uniform. Informing A-weighted levels instead of the spectrum is misleading. A tone of 125 Hz and 101 dB, if measured with A weighting will read 85 dBA, the same as another tone of 1 kHz and 85 dB. Informing that both sounds have an A-weighted sound pressure level of 85 dBA could lead nonexperts to conclude mistakenly that they are equal. This work reports spectral measurements results of environmental noise in different Peruvian cities. Comparison and analysis of measurements of traffic noise from different categories (L. M, N) and subcategories of vehicles in more than ten Peruvian cities throughout a five-year period of environmental studies are presented. As it will be shown, while the same LAeq can be measured at different locations and cities, sound spectra are radically different.

Resumen: Un trabajo publicado en el V Congreso Iberoamericano de Acústica presentó mediciones del centro de Lima en dBA. Por ser similares los autores concluyeron que el ruido ambiental en ciudades peruanas es uniforme. Informar los niveles con ponderación A y no el espectro crea confusión y errores. Un tono de 125 Hz y 101 dB, medido con ponderación A tendrá un nivel de presión sonora ponderado de 85 dBA, al igual que un segundo tono de 1 kHz y 85 dB. Si sólo se informra que ambos sonidos son de 85dBA la gente no experta podría concluir erróneamente que son iguales. Este trabajo informa los resultados de mediciones espectrales de ruido ambiental en distintas ciudades del Perú. Se presenta la comparación y análisis de las mediciones de nivel de presión sonora proveniente de fuentes móviles terrestres en todas sus categorías (L, M, N) y sub-categorías, en más de diez ciudades del Perú a lo largo de cinco años de estudios ambientales. Se pondrá de manifiesto que, a pesar de registrar el mismo NSCE en dBA en ciudades o localidades diferentes, los espectros sonoros son completamente diferentes

A055R - ASOLOFAL y su Comité de Ecología y Ruido: 21 años de actividad hacia un entorno saludable

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Abstract: At a joint meeting with the Otorhinolaryngological Society on “Noise induced hearing loss - Its social and work place consequencies”, held in 1988, ASOLOFAL proposed the constitution of a Committee devoted to this issue. On October 10, 1988, an interdisciplinary team with the participation of medical doctors, lawyers, psychologists, architects, engineers and phonoaudiologists started to work under the designation of the Scientific Interdisciplinary Ecology and Noise Committee. The main objectives were: Do interdisciplinary research on noise pollution; contribute to the solution of individual and social noise problems; increase the awareness of citizens about noise by means of seminars, courses, and conferences. The Committee carries out an intense social action through information to the community. I was also created the Juniors Committee with the aim that youngsters spread the word about the risks of noise among their mates, since teenagers are considered at risk. The action against noise requires much perseverance. The same can be said about trying to get noise ordinances adequately enforced. All this action takes into account that people have the right to life and physical and mental health, and to an adequate environment that allows the development of the human being. We all have, on the other side, the duty of preserving that environment.

Resumen: ASOLOFAL, a partir de una jornada conjunta con la Sociedad de Otorinolaringología sobre “Alteraciones del oído inducidas por el ruido - Repercusión social y laboral” realizada en el año 1988, propone la constitución de un Comité que se aboque a esta problemática. El 10 de octubre de ese mismo año, sustentado por un equipo constituido por médicos, abogados, psicólogos, arquitectos, ingenieros y fonoaudiólogos se inicia una enriquecedora tarea interdisciplinaria. Los principales objetivos fueron: realizar investigaciones multidisciplinarias sobre contaminación por ruido; contribuir mediante la difusión a resolver los problemas de ruido individuales y sociales; concientizar a la ciudadanía acerca del ruido mediante seminarios, cursos, jornadas y congresos. El Comité desarrolla una intensa acción social concretada en formación e información a la comunidad. Se creó el Comité Juvenil para concientizar y difundir los efectos perjudiciales de este flagelo entre los jóvenes, por considerarla una población de riesgo. La acción contra el ruido requiere, finalmente, mucho valor y perseverancia, al igual que lograr el cumplimiento de ordenanzas que tiendan a alcanzar un equilibrio sonoro y la concientización de la sociedad sobre el tema. Estas actividades se llevan a cabo teniendo en cuenta que todos los individuos tienen derecho a la vida y a la integridad física y mental, así como a disfrutar de un medio ambiente adecuado para el desarrollo de la persona, y el deber de conservarlo.

A020R - Evolución del proyecto de ley de protección ambiental de la calidad acústica en la provincia de Buenos Aires

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Abstract: The evolution in the legislative approach to noise pollution in the province of Buenos Aires is presented in this work. Also, the current draft of the Law for the Environmental Protection of the Acoustic Quality is described. This project was developed considering two previous projects: the draft of the provincial law called Setting the Prevention, Monitoring, and Remediation of Noise Pollution in Jurisdiction of the Province of Buenos Aires, and the National project called Minimum Assumptions Law for the Acoustic Quality Environmental Protection. The first project was passed by the Senate of the Buenos Aires Province (2003), but then expired at the Provincial House of Representatives in the discussion stage. The second one, was passed by the National House of Representatives (2007), and then expired in the discussion stage in the Senate. These references have made it possible to discuss and review the original provincial project, arriving to a set Minimum Assumptions Law requirements. This made it possible that the new instrument became more compact and in line with the national project currently in revision

Resumen: En este trabajo se analiza la evolución del enfoque legislativo de la contaminación acústica en la Provincia de Buenos Aires, y se describe el actual anteproyecto de Ley para la Protección Ambiental de la Calidad Acústica. El mismo se elaboró considerando básicamente dos documentos previos relacionados con el tema: el anteproyecto provincial denominado “Estableciendo la prevención, vigilancia, y corrección de la contaminación sonora en jurisdicción de la Provincia de Buenos Aires”, y el anteproyecto nacional denominado “Ley de presupuestos mínimos de protección ambiental de la calidad acústica”. Ambos llegaron a tener media sanción, en 2003 y en 2007 respectivamente, pero caducaron antes de obtener la plena aprobación, mientras se encontraban en etapas de discusión, el primero en la H. Cámara de Diputados de la provincia, y el segundo en la H. Cámara de Senadores de la Nación. Estas referencias permitieron discutir y revisar el anteproyecto provincial original, que además fue adecuado a la Ley de Presupuestos Mínimos, por lo que el nuevo instrumento resultó más compacto y acorde al anteproyecto nacional actualmente en revisión.

Sesión B3
Audio

Viernes 20/11/09 ~ 08:50-10:30

A004R - Empleo de técnicas de grabación para emulación de tridimensionalidad en una aplicación estereofónica

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Abstract: For a long time several investigations have been conducted in order to explain how humans localize sound sources in a natural listening environment. Localization in the horizontal plane is mainly influenced by the time and intensity differences of the arriving sound at the ears, and the pinna´s filtering information is important for vertical plane localization, when the location and distance of the sound source is changed. The stereophonic recording techniques provide considerable information about sound sources in the horizontal plane, supporting their localization. For that reason the emulation of the pinna´s properties through artificial modeling is investigated, in order to obtain the synthetic pinna's impulse response, therefore a convolution with a stereophonically recorded signal can be done, resulting in a stereophonic signal with the pinna´s characteristics. This provides a sound with a spatial hearing impression as it is perceived in natural hearing and provides auditory sensation that depends on the position of the sound source.

Resumen: Por mucho tiempo varias investigaciones han sido dirigidas a la explicación de cómo los seres humanos ubican una fuente sonora en un ambiente de audición natural. La localización sonora en el plano horizontal es básicamente influenciada por las diferencias de tiempo y de intensidad con la cual el sonido arriba a los oídos, mientras que para la ubicación de sonidos en el plano vertical se utiliza la información dada por el filtrado de la pinna, cuando la dirección y la distancia de la fuente sonora es variada. Las técnicas de grabación estereofónicas proporcionan información de importancia para fuentes sonoras ubicadas en el plano horizontal, facilitando su localización. Por tal motivo se busca emular las propiedades de la pinna y, mediante un modelado artificial de la misma, obtener su respuesta al impulso que permita, por convolución, aplicar estas características a una señal estereofónica previamente capturada. Esto proporciona un sonido con una espacialidad tal y como se tiene en la escucha natural y causa ciertas sensaciones auditivas dependiendo de la posición de la fuente.

A013R - Diseño y desarrollo de un amplificador de audio de potencia de medición clase D con SMPS

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Abstract: This paper presents the design and development of a measurement audio power amplifier (MAPA), Class D, analog closed loop topology with a self-oscillating sigma-delta modulator (ΣΔM). The relevant features are: large bandwidth (1 Hz to 50 kHz, ref −3 dB), high efficiency (> 90 %, 350 W, 1 kHz, 4 Ω), low weight (4,580 kg), high power output (350 W, 4 Ω), low total harmonic distortion plus noise (0,1 %, ref 1 kHz, 324 W, 4 Ω) and low cost. The power stage amplifier is powered by a stabilized switched mode power supply (SMPS), half-bridge topology. In order to make the MAPA more reliable and robust, it includes protection features for overvoltage (OVP), overcurrent (OCP), direct current (DCP) protection and double electromagnetic insulation for electromagnetic compatibility (EMC). The MAPA has a very large application field; usually it is used as an audio processor in measurement systems applied to room acoustics, architectural acoustics, sound and vibration sources, infrasound and ultrasound.

Resumen: En este trabajo se presenta el diseño y desarrollo de un amplificador de audio de potencia de medición (MAPA: Measurement Audio Power Amplifier) Clase D, topología lazo cerrado analógico (ACL: Analog Closed Loop) con modulador sigma-delta auto oscilante (ΣΔM). Sus características principales son: gran ancho de banda (1 Hz a 50 kHz, ref −3 dB), alta eficiencia (> 90%, 350 W, 1 kHz, 4 Ω), bajo peso (4,580 kg), alta potencia de salida (350 W, 4 Ω), baja distorsión armónica total más ruido (0,1 %, ref 1 kHz, 324 W, 4 Ω) y bajo costo. La etapa de amplificación de potencia (APS: Amplifier Power Stage) es alimentada por una fuente estabilizada de potencia conmutada (SMPS: Swithed-Mode Power Supply), topología medio puente. Se implementaron circuitos de protección por: sobre tensión (OVP: Overvoltage Protection), sobre corriente (OCP: Overcurrent Protection), corriente continua (DCP: Direct Current Protection) y un doble blindaje electromagnético para mejorar características como confiabilidad, robustez y compatibilidad electromagnética. El MAPA posee un amplio campo de aplicación, debido a que usualmente es utilizado como procesador de audio en sistemas de medición referidos a acústica de salas, acústica arquitectónica, fuentes de sonido y vibraciones, infrasonido y ultrasonido.

A048R - Medición y análisis del ruido eléctrico producido por las fuentes de alimentación en sistemas de audio

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Abstract: In the present paper the main sources of electric noise in audio systems are analyzed. They are primarily caused by leak and parasitic currents found between the power supply and the ground system connections. In the first place a typical audio chain is set up, and an equivalent model of its power supply systems and ground connections is made. Subsequently, measures in the search of parasitic currents and capacities are made. Besides the internal noise sources, the incidence of balanced and unbalanced transmission lines incidence is also measured as regards the appearance of ground loops.

Resumen: En el presente trabajo se analizan las principales causas de ruido eléctrico en sistemas de audio. Las mismas son ocasionadas primordialmente por las corrientes parásitas y de fuga presentes entre fuentes de alimentación y sistemas de conexionados a tierra. En primer lugar se configura una cadena típica de audio, y se confecciona un modelo equivalente de sus sistemas de alimentación y puestas a tierra. Posteriormente, se procede a las mediciones en búsqueda de las corrientes y capacidades parásitas. Además de los inconvenientes internos, también se mide la incidencia de las líneas de transmisiones balanceadas y desbalanceadas en su contribución a la aparición de bucles de tierra. Finalmente se analizan métodos de atenuación de ruido mediante la reducción de capacidades parásitas y de fuga.

A056R - Representación de señales de audio con descomposición empírica de modos y submuestreo adaptativo

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Abstract: The importance of lossless audio coding has been increasing in the last years. Many coding algorithms have been developed and some of them have become international standards. Despite their effectiveness, these methods assume linearity of the signal under study, which is not always true since acoustic signals are highly affected by nonlinearities. In this paper, a novel representation of acoustic signals is presented by using the empirical mode decomposition method, which has been proved suitable for nonlinear signals analysis. Subsequent subsampling of the sequences obtained from this decomposition is perfomed, giving a reduced data set. This method is applied to a short time signal and its performance is evaluated by means of the resulting compression factor.

Resumen: En los últimos años se ha incrementado la importancia de la compresión de audio sin pérdidas. Para tal fin se han desarrollado numerosos algoritmos, y algunos de ellos se han convertido en estándares internacionales. A pesar de la buena eficiencia obtenida, estas metodologías asumen linealidades en las señales analizadas, lo cual no siempre es cierto si se tiene presente que las mismas son altamente no lineales. En este trabajo se presenta una nueva representación de señales acústicas por medio de la descomposición empírica de modos, la cual se corroboró que es adecuada para el análisis de señales

no lineales. Posteriormente, se submuestrea cada señal obtenida en la descomposición a tasa suficientemente baja, resultando un pequeño conjunto de datos. Este método se aplica a una señal de corta duración temporal, y su performance se evalúa mediante el factor de compresión resultante.

A061R - Trabajos de investigación desarrollados en la materia Electiva Técnica de octavo semestre

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Abstract: Within the curricula of the Sound Engineering degree, students have to take, at different stages of the career, several subjects generically known as “Technical Electives”. During these subjects and based on previously acquired knowledge, they develop different tasks such as information search, background research, analysis of manufacturing feasibility, practical verification of different theories, mechanisms or devices associated with the subject matter, as proposed by the professor in charge. Under this scenario, the students whose names are listed, had to choose one subject among thirty five possibilities, assuming the responsibility of developing it successfully in a seven-week timeframe in order to pass the subject. The present work gives a summary of some of the results.

Resumen: Dentro de la actividad curricular que desarrolla el programa de Ingeniería de Sonido, los alumnos, en las diferentes etapas de aprendizaje, deben cursar materias denominadas genéricamente “Electivas Técnicas”. En ellas, y con base en los conocimientos ya adquiridos, desarrollan un trabajo de búsqueda de información, investigación de antecedentes, factibilidad de construcción y verificación de comportamiento de alguna teoría, mecanismo o artefacto vinculado con la especialidad, a propuesta del docente. En este caso, para aprobar la materia, los alumnos debieron elegir un tema entre treinta y cinco posibles, asumiendo el compromiso de desarrollarlos satisfactoriamente en el escaso tiempo de siete semanas. En este informe se presenta una reseña de algunos de los resultados obtenidos.

Sesión A4

Temas misceláneos

Viernes 20/11/09 ~ 11:40-12:40

A002R - Relación entre la absorción acústica y la porosidad total en un suelo volcánico

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Abstract: There are several methods for determining physical properties of soils; however, many of them require too much time and have high costs. This study proposes a methodology to obtain the total porosity of soil from acoustical measurements. Through specific acoustic impedance measurements of a volcanic soil correlation levels are obtained between average absorption coefficient and the porosity of soil samples. On the other hand, the behaviour of the theory of the *quasi-homogeneous absorber material* is studied for the soil under test, arriving at good approximations. Finally, through a linear relationship between porosity and average absorption coefficient an simple alternative model is proposed that allows to obtain the total porosity of the soil as a function of the volume moisture, the average absorption coefficient and other physical, chemical and/or biological parameters that may have some influence on the sound absorption of the soil, represented by constants obtained through a statistical analysis.

Resumen: Existen diversos métodos que permiten determinar propiedades físicas de los suelos, sin embargo muchos de ellos requieren de tiempo y costo elevado. En este estudio se propone una metodología que permite obtener la porosidad total de un suelo a partir de mediciones acústicas. A través de mediciones de impedancia acústica específica de un suelo volcánico se obtienen niveles de correlación entre el coeficiente de absorción medio y la porosidad de las muestras obtenidas. Por otra parte, se estudia el comportamiento de la teoría del *material absorbente cuasi-homogéneo* para el suelo analizado, obteniéndose buenas aproximaciones. Finalmente, a través de una relación lineal entre la porosidad y el coeficiente de absorción medio se propone un modelo simple que permite obtener de forma alternativa la porosidad total de un suelo en función de su humedad volumétrica, el coeficiente de absorción medio y otros parámetros físicos, químicos y/o biológicos que puedan tener alguna influencia en la determinación de la absorción acústica en el suelo, representados por constantes obtenidas a través de un análisis estadístico.

A030R - Análisis del ruido producido por bombas de agua para motores de combustión interna

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Abstract: The noise caused by water pumps of internal combustion engines that generate clearly audible sounds at random has been studied. In order to determine what pump element was causing the problem, accelerations in the three axes were measured and acquired by a piezoelectric accelerometer fixed to a metal bucket attached to the body. Data were stored in a digital audio tape (DAT) recorder and processed in the laboratory using a program developed for that purpose in MatLab. The study consisted of: a) comparison of measurements taken in noisy and quiet pumps and b) determination of the spectral features of the vibrations to find out which was the pump element that caused noise. The noise was found to be caused by the pump integral axle due to a poorly polished track roller bearing shaft.

Resumen: Se estudió el ruido producido por bombas de agua de motores de combustión interna que de manera aleatoria resultan ruidosas y producen sonidos claramente audibles. Con la finalidad de determinar qué elemento de las bombas era el causante del problema, se midieron aceleraciones en los tres ejes, captadas con un acelerómetro piezoeléctrico fijado a un cubo metálico solidario con el cuerpo de las mismas. Los datos se almacenaron en un grabador digital DAT (Digital Audio Tape) y se procesaron en laboratorio utilizando un programa desarrollado para tal fin en MatLab. El estudio consistió en: a) comparar las mediciones realizadas en bombas consideradas ruidosas y silenciosas y b) determinar las características espectrales de las vibraciones para dilucidar cuál era el elemento de la bomba que originaba el ruido. Se concluyó que el causante del ruido era el eje integral de la bomba debido a un pulido deficiente de los asientos de las bolillas y rodillos de los rulemanes. Además, se caracterizaron las vibraciones de las bombas silenciosas para diferentes condiciones de funcionamiento.

A061R - Trabajos de investigación desarrollados en la materia Electiva Técnica de octavo semestre

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Abstract: Within the curricula of the Sound Engineering degree, students have to take, at different stages of the career, several subjects generically known as "Technical Electives". During these subjects and based on previously acquired knowledge, they develop different tasks such as information search, background research, analysis of manufacturing feasibility, practical verification of different theories, mechanisms or devices associated with the subject matter, as proposed by the professor in charge. Under this scenario, the students whose names are listed, had to choose one subject among thirty five possibilities, assuming the responsibility of developing it successfully in a seven-week timeframe in order to pass the subject. The present work gives a summary of some of the results.

Resumen: Dentro de la actividad curricular que desarrolla el programa de Ingeniería de Sonido, los alumnos, en las diferentes etapas de aprendizaje, deben cursar materias denominadas genéricamente "Electivas Técnicas". En ellas, y con base en los conocimientos ya adquiridos, desarrollan un trabajo de búsqueda de información, investigación de antecedentes, factibilidad de construcción y verificación de comportamiento de alguna teoría, mecanismo o artefacto vinculado con la especialidad, a propuesta del docente. En este caso, para aprobar la materia, los alumnos debieron elegir un tema entre treinta y cinco posibles, asumiendo el compromiso de desarrollarlos satisfactoriamente en el escaso tiempo de siete semanas. En este informe se presenta una reseña de algunos de los resultados obtenidos.

Sesión B4
Acústica computacional

Viernes 20/11/09 ~ 11:40-12:40

A032R - Contrastación de algoritmos de análisis de espectro sonoro con un instrumento normalizado

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Abstract: Several paradigms for spectrum analysis of acoustic signals are discussed, such as digital band filters either with finite impulse response (FIR) or infinite impulse response (IIR) and Fast Fourier Transform (FFT). The use of the FFT as a computationally efficient tool for FIR band filtering is also discussed. The normative conditions for band filters included in International Standard IEC 61260 are described as well as the design of digital filters that comply with the corresponding templates. The frequency responses obtained with these algorithms are compared to those of a standard spectrum analyser (Brüel & Kjaer 2250). The methods studied include FFT with classification in the frequency domain, IIR filtering with undersampling (multirate filtering) and FFT filtering with overlap-add techniques. Both systems' responses to the same synthetic signals are obtained. The signals are pure tones of several frequencies.

Resumen: Se describen los paradigmas para el análisis de espectro de señales acústicas, tales como filtros de bandas digitales de respuesta al impulso finita (FIR) e infinita (IIR) y la transformada rápida de Fourier (FFT). También se describe el empleo de la FFT como herramienta computacionalmente eficiente para el filtrado de banda FIR. Se discuten los requisitos de la Norma IEC 61260 sobre filtros para mediciones y el diseño de filtros que satisfagan las plantillas para su implementación por software. Se comparan las respuestas de los algoritmos desarrollados con la de un analizador de espectro normalizado (Brüel & Kjaer 2250). Los métodos estudiados son la transformada rápida de Fourier (FFT) con clasificación en el dominio frecuencial, el filtrado con respuesta al impulso infinita (IIR) y técnicas de submuestreo (filtrado multitasa), y el filtrado FFT mediante técnicas de solapado-suma (overlap-add). Se evalúan las respuestas de ambos sistemas ante las mismas señales digitales. Las señales usadas son tonos puros estacionarios con frecuencia ajustable.

A051R - Noise Source Identification by Beamforming Technique

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Abstract: The noise produced by complex systems such as automobiles, airplanes, machines, etc. contributes to the overall noise of an environment. In orfer to reduce noise levels, it is necessary to identify the prominent sources. In many cases the beamforming technique is capable to carry out this task creating a visual map of the sound source. The goal of this work is to provide a basic understanding of beamforming techniques including software and hardware, and to illustrate its wide applicability to common noise control and vibration problems.

Resumen: Los ruidos producidos por sistemas complejos como automóviles, aeronaves, máquinas, etc., contribuyen al ruido global de un ambiente. De modo de reducir estos niveles, se hace necesario identificar las fuentes prominentes. En muchos casos, la técnica de *beamforming* es capaz de realizar esta tarea creando un mapa visual de la fuente sonora. El objetivo de este trabajo es contribuir a la comprensión básica de dicha técnica, incluyendo software y hardware, e ilustrar su amplia aplicabilidad en problemas comunes de control de ruido y vibraciones.

Resumo: Os ruídos produzidos por sistemas complexos como automóveis, aeronaves, maquinas, etc. contribuem para o ruído global de um ambiente. De modo a reduzir estes níveis, a identificação das fontes proeminentes se faz necessária. Em muitos casos, a técnica de beamforming é capaz de realizar esta tarefa criando um mapa visual da fonte sonora. Este trabalho objetiva fornecer o entendimento básico sobre a técnica de beamforming, incluindo software, hardware e ilustrando sua ampla aplicação para o controle de ruído.

A057R - Un modelo continuo de asignación de tráfico para el estudio de ruido urbano

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Abstract: Trip distribution and assignment in urban transportation networks is approximately governed by means of the first principle of Wardrop (user equilibrium). This principle establishes that each motorist chooses its minimum- travel-time route. The mathematical formulation of this principle leads to an optimization problem. It is a hard problem from the computational point of view because of the high number of unknowns involved (the traffic flows corresponding to the different streets of a city). A simplified approach having sufficient accuracy for analyzing the general features of traffic generated by potential urban changes, is given by means of the continuum model. This theory may be formulated as a system of non-linear partial differential equations. From its solution, the general characteristics of the traffic may be predicted. This work deals with the mathematical formulation of the continuum traffic model. The finite element method is used for solving the problem. Finally, the traffic noise may be predicted by coupling acoustic formulae with the aforementioned model.

Resumen: La distribución de viajes en una red de transporte urbano viene aproximadamente gobernada por el primer principio de Wardrop y se conoce como “equilibrio de usuario”. Este principio establece que los conductores eligen las rutas de mínimo tiempo de recorrido y ninguno podrá experimentar un tiempo menor utilizando otra ruta. La formulación matemática de dicho principio lleva a un problema de optimización cuyas variables corresponden a los flujos de tráfico circulando por las diferentes arterias de una ciudad. Se trata de un problema computacional muy costoso atendiendo a la enorme cantidad de incógnitas. Un enfoque simplificado y aún suficientemente preciso para analizar las características generales del tráfico ante diferentes modificaciones urbanas viene dado por el modelo continuo. Esta teoría se reduce a un sistema de ecuaciones diferenciales a derivadas parciales no lineales. A partir de la solución de las mismas puede obtenerse el patrón aproximado de tráfico urbano. En este trabajo se plantean las ecuaciones gobernantes del problema de asignación de tráfico desde el punto de vista de un enfoque continuo. Las ecuaciones resultantes son resueltas mediante el método de los elementos finitos. Finalmente se analiza el impacto acústico de diferentes modificaciones urbanas acoplando al modelo citado una formulación predictiva de ruido automotor.

Sesión A5

Salud y protección auditiva

Viernes 20/11/09 ~ 14:30-16:30

A014R - ¿Qué sabemos sobre los efectos del ruido?

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Abstract: Noise pollution is defined as the significant increment of sound levels in the environment that contributes to health and life quality impairment in human beings. A great deal of research has been done concerning the harmful effects of noise on people, but until now, no studies have been found about the knowledge that population has about such effects. As an approach to this problem, an instrument has been developed at the Acoustics Research and Transference Center of the National Technological University that at present is in the validation process. One of variables studied is Sound Knowledge, defined as all the information that people say they have about noise pollution as well as its effects on health and life quality. This work is the result of processing data acquired from one of the sub-dimensions of that variable named Noise Effects, from a sample of students from the National University of Córdoba and the National Technological University.

Resumen: La contaminación sonora puede definirse como el incremento significativo de los niveles sonoros en el medio, los que contribuyen al detrimento de la salud y la calidad de vida. Existen numerosas investigaciones referidas a los efectos nocivos que produce el ruido en la salud, pero hasta el momento no se han encontrado estudios sobre el conocimiento que posee la población sobre dichos efectos. Para investigar esta problemática se construyó en el CINTRA un instrumento que se encuentra actualmente en proceso de validación. Una de las variables a indagar es Conocimiento Sonoro definido como toda información que las personas manifiesten poseer en relación a la problemática de la contaminación sonora, así como sus efectos en la salud y en la calidad de vida de las personas. Se presentan los resultados obtenidos del procesamiento de datos de una de las sub dimensiones de dicha variable denominada Efectos del Ruido, en una muestra de estudiantes de la Universidad Nacional de Córdoba y de la Universidad Tecnológica Nacional-Facultad Regional Córdoba.

A019R - Evaluación de los hábitos de uso de reproductores portátiles de música por adolescentes

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Abstract: The advancement of portable sound reproduction technology and the decrease of cost are causing a great increase in the usage of this kind of devices. Besides, the demographic growth of urban centers and the subsequent increase of transportation imply an increase in environmental noise. These factors combined cause an increment in sound exposure. As a consequence, people that insist on using portable audio equipment may suffer permanent hearing impairment. In this paper an assessment of the use of portable audio players with earphones by teenagers has been done. To this aim, a survey has been conducted in that population in Buenos Aires and neighbouring areas. The main variables were daily exposure time, age at which they started to use those devices, subjective playing leveles, kind of earphones and awareness of potencial hearing impairment.

Resumen: El avance en las tecnologías de reproducción sonora portátil y la reducción de sus costos están provocando un gran incremento en el uso de este tipo de dispositivos. A su vez, el crecimiento demográfico en los grandes centros urbanos de la región y la consecuente intensificación de los medios de transporte generan un aumento en los niveles de ruido ambiental. Estos factores combinados inducen a que la exposición sonora se esté incrementando en las ciudades. Una de las consecuencias adversas es que

las personas que persisten en el uso de los reproductores portátiles en el ámbito urbano pueden sufrir hipoacusias irreversibles. El presente trabajo consiste en la evaluación de los hábitos de uso de reproductores sonoros portátiles mediante auriculares por parte de los adolescentes. A tal fin, se aplicaron encuestas a dicha población en la Ciudad de Buenos Aires y áreas aledañas. Las principales variables estudiadas fueron: tiempo diario de exposición, edad de iniciación en el uso, niveles subjetivos de reproducción, tipo de auriculares empleados y conciencia del potencial daño provocado a la audición.

A024R - Inmisión sonora, emisión y porcentajes de uso de voz en operadores de call centers

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Abstract: A set of noise immission measurements carried out in Call Center workplaces are analyzed. At the first stage the immission levels to which workers are exposed were surveyed, making use of the so-called “manikin technique”. The experience and the data obtained allowed to plan a second stage of the study, consisting in survey of the immission levels under international standards ISO 11904-2 and thus to be able to compare directly these levels with the risk levels in the workplace set by regulation 295/03. Additionally, this second stage has included an assessment of operators voice efforts through the measurement of emission levels, based on the sound pressure level that produces the voice. Finally, the percentage time of voice use is analyzed to cover a comprehensive study in these areas.

Resumen: Se analizó un conjunto de mediciones efectuadas en operadores de Call Centers. En una primera etapa se relevaron los niveles de inmisión sonora a los que están expuestos, utilizando para ello la denominada “técnica del maniquí”. La experiencia llevada a cabo y los datos obtenidos permitieron diagramar una segunda etapa en el estudio, consistente en relevar los niveles de inmisión sonora bajo la normativa internacional ISO 11904-2 y así poder contrastar directamente los niveles obtenidos con los niveles de riesgo en ámbitos laborales establecidos por la Resolución 295/03. Adicionalmente, en esta segunda etapa se ha incluído una evaluación del esfuerzo de la voz de los operadores a través de la medición de los niveles de emisión, basada en el nivel de presión sonora que produce la voz. Por último, se analizan los porcentajes de tiempo de uso de la voz para abarcar un estudio integral en estos ámbitos.

A021R - Atenuación de ruido de protectores auditivos del tipo orejera según la técnica MIRE

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Abstract: When using a hearing protector at the workplace, it is necessary to quantify its noise attenuation by laboratory measurements. This paper aims at developing an objective methodology for measuring hearing protector noise attenuation of earmuff type, using the "Microphone In Real Ear" (MIRE) method. The methodology uses the "Insertion Loss" (IL) as a parameter, which is the difference between the sound pressure level with and without the hearing protector, using MIRE, and then calculating the hearing protector IL. The results for four different hearing protectors are compared with the subjective method "Real-Ear Attenuation At Threshold" (REAT). Correction factors such as physiological noise and bone conduction are quantified using the relation between MIRE and REAT method, and then the results are compared with values obtained from the literature.

Resumen: Para el uso de los protectores auditivos en el ambiente de trabajo es necesario cuantificar la atenuación de ruido por mediciones hechas en laboratorio. Este documento tiene como finalidad el desarrollo de una metodología objetiva para medir la atenuación de ruido de protectores auditivos del tipo orejera, utilizando el método "Microphone In Real Ear" (MIRE). La metodología utiliza el parámetro "Insertion Loss" (IL), que es la diferencia entre el nivel de presión sonora con y sin protector auditivo, usando MIRE, y calculando el IL del protector auditivo. Los resultados obtenidos para cuatro diferentes protectores son comparados con el método subjetivo "Real-Ear Attenuation At Threshold" (REAT). Factores de corrección, tales como el ruido fisiológico y la conducción vía ósea, son cuantificados utilizando la relación entre los métodos MIRE y REAT para luego ser comparados con valores obtenidos en la literatura.

A022R - Aplicación de métodos normalizados de ensayos acústicos y físicos a protectores auditivos en el CINTRA

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Abstract: The main objective of using hearing protectors is to reduce to acceptable levels the noise that people are commonly exposed to in labour environments, as well as to provide a better protection for the hearing organ. To achieve this aim there are now standard methods for acoustic and physical testing, which are applied at the CINTRA, to verify the quality and attenuation provided by the protector. In addition, the methodology that has been applied allows the comparison between performance data or functional behavior obtained in various laboratories under similar conditions. These data can be used to classify and select different brands and models as well as to assess design and constructive features that influence hearing protectors performance. Given the conditions and development attained by the CINTRA in such tests, it is committed to managing the quality of the data, demonstrating technical competence to produce results in accordance with the framework of Standards IRAM 4060 (Part 1, 2 and 3) and IRAM 4126 (Part 1, 2 and 3).

Resumen: El uso de los protectores auditivos tiene como finalidad la reducción a niveles aceptables del ruido al que se está comúnmente expuesto en los ambientes laborales, intentando proveer la mejor protección posible al órgano de la audición. Para lograr este objetivo existen en la actualidad métodos normalizados para la realización de ensayos acústicos y físicos, los que son aplicados en el CINTRA, a fin de comprobar tanto la calidad como la atenuación brindada por el protector. A su vez, la metodología aplicada permite comparar los datos de rendimiento o comportamiento funcional obtenidos en distintos laboratorios, pero en condiciones similares. Estos datos pueden utilizarse para la clasificación y la selección de distintas marcas y modelos, así como la evaluación de las características de diseño y construcción que influyen sobre su desempeño. Dadas las condiciones y desarrollo alcanzado por el CINTRA en este tipo de ensayos, se asume el compromiso de gestionar la calidad de los mismos, demostrando competencia técnica para generar resultados válidos en el marco de las Normas IRAM 4060 (Parte 1, 2 y 3) e IRAM 4126 (Parte 1, 2 y 3).

A049 - La musicoterapia y la ingeniería acústica

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Abstract: The five music therapy models accepted worldwide are described and summarized in this work. Three cases of music therapy are also included, chosen from several cases treated with these models. This allows professionals of different areas to understand easily the global concepts of music therapy. You can see a compilation of physiological or psychological human affects, educational and work improvements of students or employees which were exposed to music therapy sessions. The results show changes in the pacient's behaviour to be analyzed statistically in the future; they can be useful in future scientific research in this field too. On the other hand, definitions of common terms in several areas (medicine, neuropsychology, audiology and speeh therapy, music, music therapy and sound engineering) are given in order to minimize systematic mistakes when professionals communicatewith each others. Finally future lines of research to investigate in music therapy or other disciplines from the acoustic engineer point of view are given.

Resumen: En el presente trabajo se estudian y sintetizan los cinco modelos de musicoterapia aceptados internacionalmente, además de tres musicoterapias escogidas mediante el estudio de algunos casos de investigación tratados con éstas, posibilitando a los profesionales de diferentes disciplinas asimilar de una manera práctica y sencilla los conceptos macroscópicos fundamentales. Al recopilarse casos de pacientes con patologías psicológicas y/o fisiológicas, educativas o laborales tratados con musicoterapia, se puede observar que se obtuvieron cambios en sus comportamientos como resultado, lo cual constituye una fuente de información útil para posteriores análisis estadísticos y futuras investigaciones científicas. Se analizan también las distintas definiciones de términos comunes a varias disciplinas (medicina, neuropsicología, fonoaudiología, música, musicoterapia e ingeniería de sonido) con el propósito de que los diferentes profesionales puedan acceder fácilmente a éstos y así minimizar los errores sistemáticos en la comunicación interdisciplinaria. Por último se presentan futuras líneas de investigación posibles en las que la Ingeniería Acústica puede efectuar aportes a la musicoterapia, entre otras disciplinas.

Sesión B5

Psicoacústica

Viernes 20/11/09 ~ 14:30-16:30

A026R - Herramientas para el estudio de la percepción, valoración y efectos en el ser humano en relación al contenido espectral y envolvente temporal del ruido

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Abstract: A new paradigm for the research on the perception, assessment and global effect of environmental noise in the human being is presented. It is based on the analysis of the sounds of diverse nature that compose this environmental noise. This paradigm studies the instantaneous psychoacoustic parameters of every sound that composes the environmental noise and its relation with global versions of those parameters, both in a time-integration sense and in the sense of the superposition of several simultaneous sources of environmental noise. Algorithms published elsewhere are described emphasizing their applications to the study of the effects of the noise in the human being. The tools developed within this paradigm framework are: an algorithm for the controlled combination of sounds (in the sense of achieving a target spectrum), spectral analyzers (octave band, 1/3 octave band and critical-band), a system of virtual auralización and a visual interactive platform for experiments of magnitude estimation of psychoacoustics parameters, among others. The controlled combination tool generates an environmental realistic sound with a controlled spectrum. The system of virtual auralización uses an algorithm for calculating the impulse response of rooms and a calibration protocol for audiometric earphones. The interactive interface of experimentation uses several anchors sounds for the evaluation of test sounds.

Resumen: Se expone un paradigma de investigación de la percepción, valoración y efecto global del ruido ambiente en el ser humano basado en el análisis de los sonidos de diversas naturalezas que lo componen. Este paradigma estudia parámetros psicoacústicos instantáneos de cada sonido que compone el ruido ambiente y su relación con versiones globales de dichos parámetros, tanto en el sentido de integración temporal como de superposición de fuentes simultáneas de ruido ambiental. Se describen algoritmos publicados en otras reuniones científicas con un enfoque que enfatiza sus aplicaciones al estudio de los efectos del ruido en el ser humano. Las herramientas desarrolladas para el estudio enmarcado en este paradigma son: un algoritmo de combinación controlada de sonidos, analizadores espectrales y de bandas de octava, tercios de octava y críticas, un sistema de auralización virtual y una plataforma visual interactiva para estudios de estimación de magnitud de parámetros psicoacústicos, entre otras. La herramienta de combinación controlada genera un sonido ambiental realista con un espectro controlado. El sistema de auralización virtual utiliza un algoritmo generador de respuestas impulsivas de salas y un protocolo de calibración de auriculares audiométricos. La interface interactiva de experimentación utiliza varios sonidos de referencia para la evaluación de sonidos de prueba.

A025R - Protocolo de calibración de auriculares audiométricos para su uso en investigación psicoacústica

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Abstract: A method for the calibration process of audiometric earphones is presented, taking as an example the Sennheiser model HDA 200 extended range audiometric earphones. This method was developed with the aim of presenting to the listener a known sound field during auralisation tasks. A block diagram of the electroacoustic chain involved in the sound virtual playback is described, drawing attention to those elements that must be compensated for a suitable auralisation.

Resumen: Se presenta un método para el proceso de calibración de auriculares audiométricos ejemplificando con los auriculares audiométricos de rango extendido marca Senheisser modelo HDA 200. El método fue desarrollado con el fin de poder presentar al oyente un campo sonoro perfectamente conocido en tareas de auralización. Se describe un diagrama en bloques de la cadena electroacústica involucrada en la reproducción sonora virtual, particularizando en aquellos elementos que deben ser compensados para una adecuada auralización.

A027 - Ponderación psicoacústica de la distorsión no lineal de dispositivos de audio: una propuesta para su medición

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Abstract: This paper researches the possibilities of using new techniques for the measurement of audio devices. In particular, the use of frequency sweeps offer the main advantages of giving a large dynamic range and the possibility to separate the linear and the non-linear responses of the system under test (SUT). By means of this separation of responses, it is possible to implement a non-linear distortion estimator considering the psychoacoustics characteristics of loudness and critical bands. A loudness model based on Artificial Neural Networks (ANN) will be used, as well as a simultaneous masking model and an ERB critical-band spectral integration. Some examples of the measurement of audio devices and electrodynamic loudspeakers are presented.

Resumen: Este trabajo investiga las posibilidades de uso de nuevas técnicas de medición para dispositivos de audio. En particular, el uso de barridos de frecuencia que ofrecen como principal ventaja un gran rango dinámico y la posibilidad de separar la respuesta lineal de la respuesta no lineal del sistema bajo prueba (SBP). Mediante esta separación de respuestas, es posible implementar un estimador de distorsión no lineal usando características psicoacústicas de sonoridad y bandas críticas. Se utiliza un modelo de sonoridad basado en Redes Neuronales Artificiales (RNA), un modelo de enmascaramiento simultáneo y una integración espectral en bandas críticas ERB. Se presentan ejemplos de medición para dispositivos de audio y altavoces electrodinámicos.

A028R - Análisis comparativo entre modelos de sonoridad: propuesta de un modelo de ponderación dinámica

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Abstract: The reference standard ISO 532:1975 is commonly used for calculating the approximated loudness level for complex acoustic signals. The so-called Method B proposes an analysis based on 1/3 octave bands for the approximation to the critical bands of the human hearing system. In this paper an alternative model is proposed which uses the warping technique to obtain a critical band frequency domain and a dynamic weighting model implemented by means of an Artificial Neural Network (ANN). The model is compared with the ISO 532-B method and with the Patterson-Holdsworth auditory model, which is based on the concept of Equivalent Rectangular Bandwidth (ERB).

Resumen: La norma de referencia ISO 532:1975 es comúnmente utilizada para el cálculo aproximado del nivel de sonoridad para señales acústicas complejas. El denominado Método B plantea el análisis en bandas de un tercio de octava para la aproximación en bandas críticas del sistema auditivo humano. En el presente trabajo se propone un modelo de sonoridad alternativo, usando la técnica de *warping* de frecuencias para obtener un dominio en bandas críticas y un modelo de ponderación dinámico implementado mediante una Red Neuronal Artificial (RNA). La propuesta es comparada con el método ISO 532-B y con el modelo auditivo de Patterson y Holdsworth, este último basado en el concepto de *Equivalent Rectangular Bandwidth* (ERB).

A031R - Umbrales de eco en participantes ciegos

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Abstract: The precedence effect is a strategy that listeners unconsciously employ to cope with multiple arrays of directional cues in reverberant spaces. It refers to the auditory phenomenon that occurs when two similar sounds —lead and lag— are presented from different locations with a brief delay between them, and only one sound is heard whose perceived location is dominated by the sound arriving first. Three aspects are involved: fusion, localization dominace and discrimination suppression. The first one —the lead and lag perceptually fuse into one auditory event— is useful to avoid multiple sound images. In order to study the possible relation between echolocation at short distance and the precedence effect, we have evaluated blind and sighted subjects with three auditory tests that measure the above mentioned aspects. In this article we present theoretical and methodological considerations and the main results obtained by blind adults in the fusion test, i.e., echo threshold measurements.

Resumen: El efecto precedente es una estrategia inconscientemente utilizada para enfrentar la información sonora conflictiva de los ambientes cerrados. Se lo refiere como el fenómeno perceptual de fusión espacial que ocurre cuando dos sonidos similares -líder y retardado- se presentan desde diferentes lugares, separados por un breve retardo de tiempo. El individuo escucha sólo un sonido que localiza según la dirección del sonido que llegó primero. Involucra tres perceptos: fusión, dominancia en la localización y supresión de la discriminación del sonido retardado. El primero se refiere a la fusión de los dos sonidos en una sola y coherente imagen auditiva, lo cual resulta útil para evitar imágenes sonoras múltiples. Con el propósito de avanzar en el conocimiento de la ecolocación humana a distancias cercanas y su relación con el efecto precedente, implementamos y administramos a participantes ciegos y con visión normal, tres pruebas auditivas especialmente diseñadas para medir los perceptos mencionados. En este trabajo se presentan aspectos teóricos, metodológicos y principales resultados obtenidos por adultos ciegos en una de las pruebas -i.e., la prueba de fusión que mide umbrales de eco-.

A053R - Re-descubriendo la ventana de integración temporal del sistema auditivo: consecuencias en la ciencia acústica

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Abstract: In this research the variability of the integration window duration of the auditory system as a function of the perceived acoustic signal is verified. Results validate the perception model based on the specialization of the brain’s hemispheres and the existence of the autocorrelation and cross-correlation functions performed at the brainstem.

Resumen: En el presente trabajo se comprueba la variabilidad de la duración de la ventana de integración temporal del sistema auditivo en función del tipo de señal acústica percibida. Los resultados obtenidos validan el modelo de percepción basado en la especialización de los hemisferios cerebrales y la existencia de las funciones matemáticas de autocorrelación y correlación cruzada realizados en el tronco cerebral.

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