Semaine canadienne de l'acoustique 2020
Acoustics Week in Canada 2020

Conférence en ligne - Online meeting

awc.caa-aca.ca/zoom

Programme préliminaire - Preliminary Program

Time	Activity
09:00-09:10	Mot de bienvenue – Welcome Note
09:10-10:00	Table ronde : Opportunités et défis de la COVID-19 dans le monde de l'acoustique - Round Table - COVID-19 Pandemic : impacts, challenges and opportunities for the CAA-ACA community
10:00-12:30	Sessions techniques – Main Session 1. Préservation de l'ouïe - Hearing Conservation 2. Acoustique musicale / Électroacoustique - Musical Acoustics / Electroacoustics 3. Traitement des signaux / Méthodes numériques - Signal Processing / Numerical Methods 4. Aéroacoustique - Aeroacoustics 5. Physio-acoustique - Physiological Acoustics 6. Acoustique sous-marine - Underwater Acoustics 7. Psycho-acoustique - Psychological Acoustics 8. Consultation - Consulting 9. Sciences de la parole - Speech Sciences 10. Acoustique architecturale - Architectural Acoustics 11. Chocs / Vibrations - Shocks / Vibrations 12. Sciences de l'audition - Hearing Sciences 13. Acoustique physique / Ultrasons - Physical Acoustics / Ultrasounds 14. Génie acoustique / Contrôle du bruit - Engineering Acoustics / Noise Control 15. Bio-acoustiques - Bio-Acoustics
12:30-13:20	Trésors acoustiques – Acoustical Gems
13:30-15:30	Présentations étudiantes Pecha-Kucha (session de 10 min de 20 diapos, 20 sec chaque, etc.) - Student Pecha-Kucha Presentations (10 min session for 20 slides, 20 sec each, etc.)
15:30-16:30	Assemblée Générale des members - Annual General Assembly
16:30-17:00	Cérémonie de remise des prix - Award Ceremony
17:00-18:00	Concert - Concert

MAIN SESSION - SESSIONS TECHNIQUES

1 - HEARING CONSERVATION - PRÉSERVATION DE L'OUÏE

In-Ear Audio And Speech Processing Algorithms For Digital Hearing Protectors

Farid Moshgelani, Jérémie Voix

Everyday more than one million Canadians working in noisy environments use Hearing Protection Devices (HPDs) to protect their hearing against the risk of noise-induced hearing loss (NIHL). Recently, a new type of digital HPD has been developed for the benefit of industrial workers exposed to noise. This digital HPD features three categories of specialized in-ear audio algorithms: The first is developed for in-ear speech capture under the HPD to enable the wearer's voice to be captured from inside the earcanal, while ambient sounds are played back at safe levels. The second deals with the classification of non-verbal events captured in the earcanal (e.g. coughing, swallowing, teeth or tongue clicking) to enable health monitoring applications for industrial workers. The third algorithm is dedicated to extracting biosignals from inside an occluded ear (e.g. heartbeat, breathing) to make it possible to track the activity and health of workers. In the current research program, the performance and functionality of the aforementioned specialized in-ear algorithms are optimized and validated across different processing conditions (e.g. different types and levels of ambient noise). Finally these algorithms are optimized for co-implementation and real-time processing into a single Digital Signal Processor (DSP). This presentation presents the required steps of the current work as well as the envisioned work for the development and qualification of a speech intelligibility framework dedicated to in-ear speech signals.

3 - SIGNAL PROCESSING / NUMERICAL METHODS - TRAITEMENT DES SIGNAUX / MÉTHODES NUMÉRIQUES

Acoustical Measurements Of The Roman Theatre Of Verona By Mapping The Sound Reflections In Real Time

Antonella Bevilacqua, Lamberto Tronchin, Francesca Merli, Marco Dolci, Umberto Berardi

Nowadays the auralization has been extensively performed inside enclosed spaces. This paper proposes the use of multi-channel spherical arrays applied to the Roman theatre of Verona. Both speaker and microphone have been used in order to achieve the graphical mapping of the sound reflections measured outdoor. Although the employed equipment is capable to change directionality and beamforming, in this experimentation the loudspeaker has been employed as an omnidirectional sound source, while the microphone continued to divide the impulse response (IR) in small frames by using the MIMO technique. By setting the equipment in this way, the energy associated to each virtual microphone can be computed and represented as a coloured map overlaid by a graphical interpolation. The new representation method shows clearly how the sound ray hits the boundary surfaces before reaching the receiver, over a full spectrum width, having a visible idea of how to identify the effectiveness of those surfaces generating early reflections or to be cause of unpleasant echoes

8 - CONSULTING - CONSULTATION

Noise From Outdoor Live Events In Downtown Montreal: Regulatory Framework, Complaints Management And Predictions Tools

Romain Dumoulin

The Quartier des spectacles, Montreal's major cultural district is a densely populated and rapidly developing sector that hosts numerous festivals and events throughout the year. These activities impact the neighborhood's soundscape and generate noise annoyance and complaints. In an effort to improve the proactivity of its noise management process, we assisted the City of Montreal in conducting a diagnosis of the current situation and provided recommendations on the regulatory framework and the use of environmental noise monitoring and prediction tools. Based on the analysis of complaints data and meetings with the main stakeholders (event promoters, sound engineers, noise control technicians, city officials from the Festival & Events department), we evaluated the complaints management process, the current regulatory framework and the strategy and resources put in place to ensure its application.We noted a sharp contrast between the management process and the information and guidelines presented in official City documentation. Spatial patterns in the locations of the noise complaints were identified and analysed. The noise level limits and its assessment methodology were identified as the main limitations. Our recommendations regarding the design and update of both the regulatory framework and the complaints management are presented. Our recommendations on the use of noise prediction tools are expanded under the following areas: 1. decision-support for city officials and event promoters, 2. Noise policies design for the development of location-specific noise limits, 3. Content creation for communications and outreach purposes.

9 - SPEECH SCIENCES - SCIENCES DE LA PAROLE

Investigation Of Motor Equivalence And Tongue Muscle Excitation Patterns In Simulated English Vowels

Noor Hisham Al-Zanoon, Daniel Aalto

Motor equivalence is a motor system's ability to achieve the same end target with varying input from individual components of that system. In speech, a particular tongue configuration can be produced by many different muscle excitation patterns. Previous studies have used various experimental techniques (imaging and kinematics tracking) to estimate the variability of muscle excitation patterns. However, these studies are limited in understanding to the surface of the tongue and cannot directly observe biomechanical parameters such as muscle activation patterns,and individual muscle contributions. Computational models of the tongue have the potential to measure muscle activation patterns. In the current study, motor equivalence is systematically studied by simulating the motion of tongue between a neutral position and two English vowels (/i, æ/). ArtiSynth is an open-source simulation platform that uniquely supports the combination of multibody and finite element (FEM) models. ArtiSynth provides a generic jaw-tongue model that fully couples the jaw with an FEM tongue. Target muscle excitations for each vowel trajectory are estimated using inverse simulation methods. For each tongue muscle contributing to the trajectory, its relative weight (contribution) will be varied from using probabilistic sampling methods, while keeping other contributing muscles at a constant weight. The excitation patterns have a high dimensionality and are reduced using principal component analysis and clustered. The excitation patterns are compared using correlation index to quantify the similarity of the muscle excitation patterns. By understanding of variation in muscle excitation patterns, further insight into compensatory strategies, individual muscle contributions will be gained.

Socio-Phonetic Tendencies And Linguistic Aspects Of English Contemporary Commercial Music (Ccm)

Gianmarco Perna

The aim of this research is to set up a consistent theorical arrengement concerning the socio-linguistic and socio-phonetic aspects of contemporary commercial music (CCM), which appears to be more and more unevenly and manifold taking into account the previous research by Trudgill (1997), Simpson (1999) and others. The massive use of phonetic variation and style switching in contemporary western singing style is analyzed mainly focusing on its phonetic, cognitive and semantical relevance. Some examples of metric, socio-linguistic and stylistic variation are provided afterwards, so that a global diagram of linguistic habits might be traced. Other collateral aspects are recollected, mostly concerning the effects of CCM experience on L2 users.


COMPRESSOR NOISE
ACOUSTIC ENCLOSURES

10 - ARCHITECTURAL ACOUSTICS - ACOUSTIQUE ARCHITECTURALE

Variance Of Reverberation Time, Clarity, And Speech Transmission Index In Gymnasium

Greer Stanier, Umberto Berardi

No federal standards that specify the acoustical quality of gymnasia exists despite the persistent issue of poor speech intelligibility in these spaces. The only provincial standard to do so is the Alberta Infrastructure and Transportation; Standards and Guidelines for School Facilities, which specifies a maximum reverberation time for gymnasia. Reverberation time is the common metric but it is not the best indicator of speech intelligibility in these educational spaces. Clarity and Speech Transmission Index are more closely correlated to clear speech and are used in other standards to ensure a good environment for instruction in rooms with high background noise. The aim of this work is to characterize four gyms using these three parameters to analyse whether they are good metrics to quantify the acoustic quality of this typology. The analysis involved collecting results from field testing the gymnasiums at two heights and subsequently modelling them in ODEON for comparison. It was shown that when measured in the field under ISO 3382, the clarity has too much experimental deviation, while STI was consistent between both the measured and the modelled data.

The Roman Theatre Of Verona: A Different Representation Method Of The Acoustical Parameters.

Antonella Bevilacqua, Lamberto Tronchin, Francesca Merli, Marco Dolci

Although many parts of the Roman theatre of Verona have been lost for ever, the reconstruction of the cavea at its original shape, after the demolitions of heterogeneous buildings invading improperly the site throughout the centuries, gives the possibility to evaluate the acoustics of this unique place. A specific study was carried out by using a particular software, enabling to predict the room-acoustical characteristics of the theatre. Both monoaural and binaural measurements were undertaken on site, highlighting how the sitting areas can have a good or lack of listening. Whilst many acoustical investigations have already been completed during the 20th century in order to set in function the Roman theatre of Verona with summer shows' programs, the results of this study contribute to refine the acoustical parameters based on a different representation method.

Analysis Of Environmental Sound Levels In Italy During The Covid-19 Emergency

Jacopo Fogola, Giovanni Brambilla, Antonino Di Bella, Gaetano Licitra, Rosalba Silvaggio, Eleonora Carletti

The emergency due to COVID-19 outbreak has led to unique and, perhaps, unrepeatable acoustic scenarios. The closure of schools and break of non-essential economic activities, the widespread use of smart working and the general requirement to stay at home have caused a significant decrease of people mobility and outdoor human life, with a consequent significant variation in noise levels and soundscapes, especially in urban areas. In order to investigate the impact of such restrictive measures in the acoustic climate, a joint research project has been signed between the Acoustical Society of Italy (AIA) and the Italian National System for Environmental Protection (SNPA). The main objectives of the project are: - to collect and organize, through a geographical information system, the data measured by public bodies, companies, acoustic consultants and private citizens; - to perform a large-scale first level analysis on sound level variations by different type of sources; - to carry out an in-depth study in some sample areas on specific acoustic effects, like variations on spectra, noise mapping, people exposure, traffic flows, number of people gathering in "movida" areas, ambulances siren impact, etc.; - to acquire information and knowledge useful for the study of quite zones. This paper will present the general context of the initiative, a summary of the available information and data and the analysis methods applied so far.

The Effect Of Sound Masking On Employees' Acoustic Comfort In Open-Plan Offices

Joonhee Lee, Roderick Mackenzie, Vincent Le Men, François Gariépy, Farideh Zarei

As the ambient sound level self-generated by occupants in an office varies according to the occupancy rates and activity types, ambient sound levels alone cannot be relied upon to provide effective speech privacy and freedom from distraction. Sound masking systems are commonly employed in open-plan offices to achieve a controlled minimum level of background sound. The loudspeakers emit a neutral sound spectrum similar to a ventilation

system so as to decrease the signal-to-noise ratio of unwanted incoming speech, making it less intelligible and thus less distracting. Many previous studies have investigated the optimal level and spectrum of the masking sound in open-plan offices, but most studies have conducted subjective testing in a laboratory setting instead of actual office environments. Thus, this study aims to examine the relationship between perceived speech privacy of employees and sound masking configurations in two operational open-plan offices located in Quebec. The testing was carried out over six weeks, with varying overall sound levels of the sound masking systems. The employees were asked to fill out subjective questionnaires before and after each change in conditions about how the sound environment impacted their comfort and work performance during the study. The statistical results show that the masking sound conditions are significantly more satisfactory than the sound condition without the masking sound. The results can help to understand the optimal sound masking system condition in open-plan offices, but also serve to demonstrate the importance of maintaining controlled minimum background sound levels in office environments.

The Evolution of the Acoustics of a Medieval Church

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1 Introduction

Churches are acoustically complex spaces due to both their large dimensions as well as the acoustically reflective materials used. Over time, the long reverberation times of churches have allowed for the development of sacred music; in fact, the sound tail inside the churches improves the listening to organ music as well as songs, increasing the participation of the faithful in religious functions. In recent decades, churches have been used both for classical and symphonic music concerts as well as conferences and conventions, but the results have not always been positive since the acoustic properties of the churches are different from those of theatres [1]. Furthermore, the Second Vatican Council amended the Catholic liturgy, giving more importance to the verbal communication, although this requirement was not supported by the passive acoustics of the churches with long reverberation times and reduced speech intelligibility [2]. In the years, in order to improve listening conditions, electroacoustic systems are often placed inside churches. However, speech intelligibility can be improved by means of an acoustic correction of the spaces through the reduction of the reverberation by inserting appropriate sound-absorbing materials.

Traditional porous sound-absorbing materials with a suitable thickness, absorb sound energy at medium and high frequencies and are successful for acoustic corrections [3, 4]. However, monumental churches have various problems due to both the presence of low frequencies components of the sound tail as well as the difficulty of using traditional materials for historical and architectural needs: the walls cannot be covered with traditional materials. Therefore, there is a need to experiment innovative solutions for the acoustic correction of the rooms, such as transparent micro-perforated sheets for medium and high frequencies and acoustic resonators for low frequencies. Acoustic resonators can be obtained by perforating ceramic tiles and installing them at an appropriate distance from the rigid wall in order to obtain an adequate sound absorption at low frequencies. This paper is dedicated to the investigation of the acoustic characteristics of the cathedral of Carinola, in Italy which was subject to a restoration for improving its acoustics.

2 Acoustic measurements

The church has a volume of 7.500 m^3 and an internal plant surface of 5.000 m^2. The church has a central large nave and three lateral smaller naves, following a typical Medieval plan. The acoustic measurements were carried out with an omnidirectional sound source located on the altar and 12 microphone points of measurement placed in different positions in an area occupied to the listeners, in accordance with ISO 3382 [5]. The sound source was powered with a MLS signal. During the acoustic measurements the background noise was lower than 35 dBA, the church was empty and the furniture were hard chairs. Figure 1 shows the plan of the church with the indication of the position of the source and of the receiver microphone points. The sound source was placed at a height of 1.6 m from the floor and the microphone at a height of 1.3 m, in an area occupied to the listeners. The monaural acoustic parameters analysed are EDT, T30, C80, and D50. Table 1 shows the average values of the acoustic parameters in the octave band from 125 Hz to 4 kHz. The values of EDT and T30 exceed on average the 4 seconds, the average values of C80 = -5 dB, while the average values of D50 = 0.25. The average value of the parameter STI = 0.33. The church does not meet the criteria of good listening for the music and the speech [2, 5].

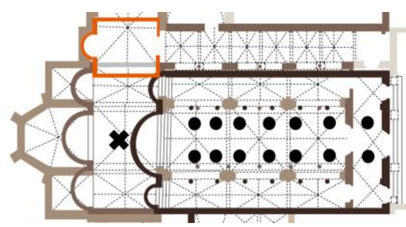

Figure 1: Plan of the church with the indication of the position of the source and of the receivers.

Table 1: Average acoustic parameter measured.

Freq [Hz]	125	250	500	1000	2000	4000
EDT, s	4.8	4.5	4.0	3.7	3.1	2.5
RT, s	4.7	4.6	4.0	3.6	3.0	2.5
C80, dB	-5.1	-6.3	-5.5	-3.9	-3.3	-2.4
D50	0.2	0.1	0.2	0.2	0.2	0.3

Figure 2 shows the reverberation time parameter trend, with distance sound source-receiver at the frequency of 1000 Hz. The parameter does not change with the distance sound source-receiver, proving that the acoustic field is diffuse.

In the late 1960s, Shankland published studies on more than Roman and Medieval churches discussing their acoustic qualities. In all the environments considered the sound was judged to be very widespread and without audible echo due to the richness of the architectural and sculptural details. Figure 3 shows the trend of the reverberation time measured in the considered church compared to others [6].

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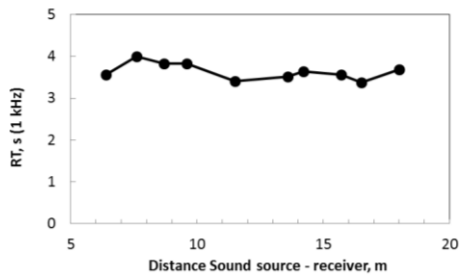

Figure 2: Reverberation time parameter, with distance sound source receiver at the frequency of 1000 Hz.

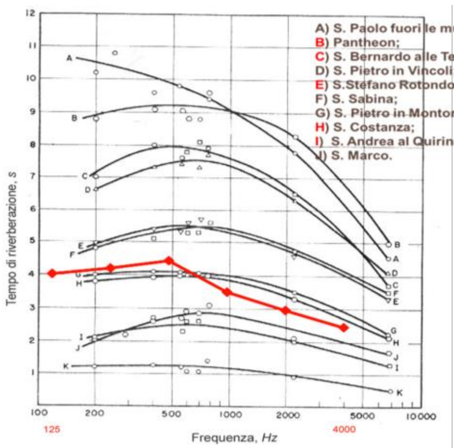

Figure 3: Comparison of reverberation times measured in different churches and in the present case study (in red).

3 Acoustic correction and discussion

The results showed in the previous section show that the church has inadequate acoustic characteristics for a good speech intelligibility, and requires intervention to fully adhere to the suggestions of the Second Vatican Council. The CAD model, shown in Figure 4, was imported into ''Odeon'' Room Acoustics Software. The sound source is placed on the altar and the receivers among the seats of the audience. The first step, the acoustic model calibration, consisted of setting the absorbent coefficient values for all the virtual model surfaces and the scattering coefficient ones. The procedure was stopped when, for each octave band frequency from 125 Hz to 4.0 kHz, the calculated reverberation time value (T30) was within ±5% of the measured one. Table 2 shows the values of absorbent coefficient used in the modeling.

A practical problem for medieval churches that traditional sound-absorbing materials such as polyester cannot be used for acoustic corrections for artistic conservation needs. A possible solution was hence to cover the side walls of the church with sound-absorbing plaster. This solution is not invasive and allows the preservation of the church from a monumental point of view. Table 2 report the sound absorbing values of the sound-absorbing plaster selected.

Figures 5 show the spatial average distribution of RT and EDT at 1000 Hz modelled with the plaster intervention. These figures show the reduction of these acoustic parameters and the clear improvement of the relative listening quality within this church.

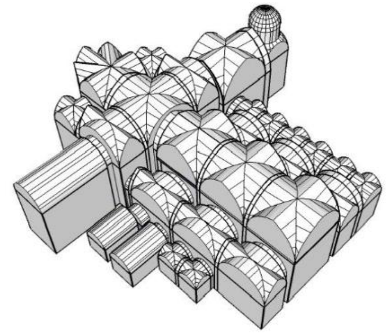

Figure 4: 3D model of the church used to test possible interventions for its acoustic correction.

Table 2: Values of absorbent coefficient of used in modelling.

Freq, Hz	125	250	500	1000	2000	4000
Plaster	0.03	0.04	0.04	0.05	0.06	0.08
Marble	0.11	0.12	0.13	0.14	0.15	0.16
New plaster	*0.2*	*0.3*	*0.6*	*0.7*	*0.8*	*0.8*

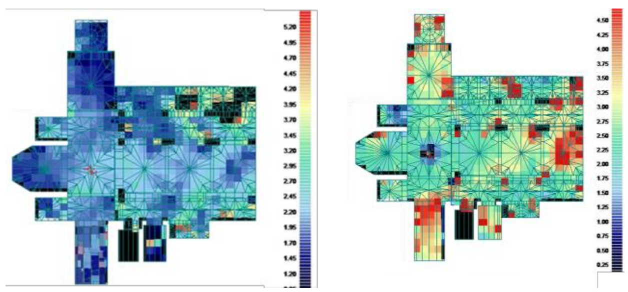

Figure 5: Distribution of the RT (left) and EDT (right), at 1000Hz.

4 Conclusion

This paper presented the study of the intervention for the acoustic correction of the Medieval cathedral of Carinola in Italy. The intervention aimed to shortening the reverberation time by substituting the plaster with a sound absorbing plaster. Acoustic interventions showed the homogeneity of acoustic across the floor plan and the possibility to obtain a reverberation adequate for an easier speech perception.

References

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11 - SHOCKS / VIBRATIONS - CHOCS / VIBRATIONS

A Critical Review On Controlling The Supercooling Of Water Through Ultrasound

Amrit Kumar Thakur

Supercooling is highly undesirable phenomenon and consumes huge amount of energy for energy storage application and thus, finding methods to reduce the degree of super-cooling of water is fundamental to advance the thermal energy storage technology. Different methods are being adopted to mitigate the supercooling using Nanoparticles, fins and metal metrics. However they are not good with increasing number of cycles and also increase the system weight. With the tremendous development in acoustic, utilisation of ultrasound , leads to significant reduction in supercooling degree and huge energy can be saved using the ultrasound application in solidification of water.This paper will discuss the extensive research work carried out in the field of reduction in the supercooling degree with usage of ultrasound and present the key findings for the future work.This review will give a new dimension in using ultrasound for energy saving potential for thermal energy storage application.

14 - ENGINEERING ACOUSTICS / NOISE CONTROL - GÉNIE ACOUSTIQUE / CONTRÔLE DU BRUIT

Design And Experimental Validation By 3d Printing Of A Permeable Silencer Metamaterial

Francesco Bianco, Francesco Brocchi, Umberto Berardi, Gaetano Licitra

The use of acoustical metamaterials (AMMs) can answer several issues in architectural and building acoustics, especially problems related to low-frequency noise control. This paper reports the investigation of an acoustic metamaterial permeable to air. This analysis and design of an AMM, conceived to improve the sound attenuation in the range of middle-low frequencies, using subwavelength structures are reported. A Finite Element Method (FEM) investigation allows us to analyze and optimize the acoustic performance of the AMM. A parametric design leads to tuning the operating frequencies behavior of the AMM. The AMM device is then experimentally validated by 3d printing the prototype and testing it. A comparison between the estimated and measured results is finally reported. This study confirms that the AMM effectively acts as a permeable and highly effective and selective sound silencer, which could be used in wide range of applications, such as building technology to provide natural ventilation to indoor environments, or directly integrated in ducts, for fans noise reduction.

Development Of A Mobile Full Face Tracking System During Speech Production

Andrew(U San) Chao, Roujan Khaledan, Kieran Armstrong, Daniel Aalto

Motion tracking systems, such as Optotrak, are used for capturing facial movements (facialgestures, jaw, and lip motions). However, the standard tracking systems are expensive andhave limited availability. We propose a mobile device app capable of capturing facialmovements with high speed video and audio recordings while a participant reads targetsentences from a mobile device screen. The app is created in Xcode and is available onTrueDepth camera iOS devices. The app uses ARKit, Apple's Augmented Reality platform, andthe iPhone's front facing TrueDepth camera. The app fits a mesh of 1220 points to the face at arate of 60 frames per second (fps) based on built-in infrared surface scanning technology.Simultaneously, video (60 fps) and audio (44.1kHz) are recorded with Apple's ReplayKit. Thespeech task, the speaker's face with the fitted surface mesh, control buttons, and the distanceto the camera are displayed in the user interface. After recording, the acquired data istransferred to a comma separated file with a Python script. The developed pilot mobile app forfull face tracking offers an accessible alternative for traditional motion tracking systems creatingopportunities to remotely collect articulatory data for research and clinical purposes.

MINI-WIND TURBINE NOISE MEASURED INSIDE NEAR-BY HOUSES

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1 Introduction

The possibility of using the wind to rotate blades and to obtain the lifting of the water or to operate millstones is an ancient technique. Today, wind power keeps being fundamental to the increased importance of renewable energy. In Italy, the country of the present study, there were 81 wind farms installed in 2001, with a total capacity of 664 MW; in 2015, there were 2,734 wind farms with a capacity of 9,162 MW. The environmental impacts caused by the construction of wind farms include the occupation of the territory, visual impact, acoustic emissions, electromagnetic emissions, possible interference with flora and fauna, and discomfort for the populations living near the wind farms. In particular, one of the main complaints reported by the resident population is the noise generated by the rotation of the wind turbine blades [1]. The functioning of a wind tower generates the aerodynamic noise produced by the rotating blades and the mechanical noise inside the nacelle. The noise generated by the operation of the wind towers, with the same perceived sound level, is more annoying than other anthropic noises. Although the sound levels caused are modest in the order of 30-50 dBA, this type of noise, due to the particular tonal component, is highly annoying. Usually, the noise emitted by a wind turbine is a broadband noise concentrated in the frequency range 300 Hz – 2000 Hz [2,3]. In this paper, the acoustic measurements inside a house of the noise produced by the operation of a 200 kW wind turbine are reported.

2 Methodology

The house in which the acoustic measurements were carried out is located in a small rural municipality. The area is a plateau within a large basin at about 700 meters above sea level, located in the central area in the South Italian Apennines. The height of the gearbox is 30 m, the rotor diameter is 20 m, and the blades rotation speed is 20 – 60 rpm. Blades began to move with wind speed around 3 m/s; and the rotation is stopped for safety conditions when wind speed is 25 m/s. The highest power production starts for wind speeds over 15 m/s. The tower is located to the east of the house, and the land is flat and has modest vegetation. The distance between the home and the tower is 250 m. During the measurements, the window was open to assess the maximum disturbance.

The acoustic measurements were carried out using a sound level meter model Larson Davis LXT1, which was calibrated with a Larson Davis CAL 200. The instruments were compile with the requirements of the IEC 61672-1 standard "Class 1". The sound level meter was configured to acquire the sound equivalent level of the "A" weighted and the L95 statistical level; this parameter is defined as the level exceeded for 95% of the observation time. The "A" -weighted filter (dBA) was used because the annoyance response to due to wind turbine nose, is related to "A"-weighted levels.

The following regulatory approaches were adopted to evaluate the annoyance produced by the wind turbines [4,5]:

- Differential criterion: the differential noise level is represented by the difference between the ambient noise level and the residual noise level. If the difference between the level of environmental noise and the residual level (wind turbines turned off) is less than 5 dBA daytime and 3 dBA nighttime, then the noise generation is considered acceptable.
- Normal tolerability criterion: the background noise (L95) is measured when wind turbines are stopped, and the equivalent level is evaluated. The normal tolerability limit is 3 dB.

3 Sound propagation theoretical models

One of the problems encountered during the installation of a wind tower is the theoretical evaluation of the noise introduced into the living environment. This evaluation is necessary to establish if the installation of the wind tower will cause annoyance to the people living in the chosen area. The most used model is based on ISO 9613-2 [4]. It considers the sound source as a point, although a wind turbine is a complex system that can be considered a point sound source only when there is a considerable distance between the sound source and the receiver. For the determination of the noise levels inside a receiver point, the standard ISO 9613-2 provides a theoretical method to evaluate the sound attenuation, with the source - receiver distance, in outdoor propagation. The standard calculates the equivalent sound pressure level assuming meteorological conditions that favor sound propagation, by applying the following relationship:

$$Lp = Lw + DI\theta - A_{div} - A_{atm} - A_{gr} - A_{bar} - A_{misc}$$

where Lp: sound pressure level (dBA); Lw: sound power level (dBA); Diθ: directivity; A_{div}: attenuation due geometric divergence; A_{atm}: attenuation due to atmospheric absorption; A_{gr}: attenuation due to ground effect; A_{bar}: attenuation due to a barrier; A_{misc}: attenuation due to foliage or industrial sites.

The application of the calculation model of noise propagation appears to be precautionary as it provides an overestimation of the levels when considering only the attenuation of the noise caused by the geometric divergence, not conside-

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ring the other attenuating factors such as atmospheric absorption, as well as the presence of obstacles and vegetation. Numerical simulations based on engineering approaches are, in many cases, a rapid application. In a simplified way, the sound pressure level is given by the formula: $Lp = Lw - A_{div}$ with $A_{div} = 20 Log(D) + 11$, where D is the distance between the sound source and the receiver and A_{div} is the sound attenuation which occurs during propagation. For a wind tower of nominal power of 200 kW, Lw is 100 Dba; so, the theoretical sound pressure levels is $Lp = Lw - A_{div} = 41$ dBA.

With the ISO 9613, it is possible to evaluate the sound pressure level at the receiving point located in a home. For the evaluation it is necessary to estimate the effect of the attenuation of the open window or the value of the difference between the sound level measured externally and the level measured internally in the home. This value is estimated to be around 4 or 5 dBA [3]. The sound level inside the house is equal to the sound level estimated in the receiving point subtracted from the attenuation value of the open window. A theoretical level inside a home of 37 dBA was predicted.

4 Acoustic measurements and discussion

The acoustic measurements were made by placing the sound level meter at about 1.6 m from the floor, and 1.0 m from the balcony in the first-floor room used as a bedroom. Measurements were performed with windows open in the maximum disturbance condition. Several sessions of measurements were performed corresponding to two operating conditions of the wind turbine. Figure 1 shows a typical measurement session with the time history of the measured sound pressure level. The environmental noise when the source is turned on, the residual noise when the source is turned off, the background noise L95 when the source turned is off and the wind speed are shown in Table 1.

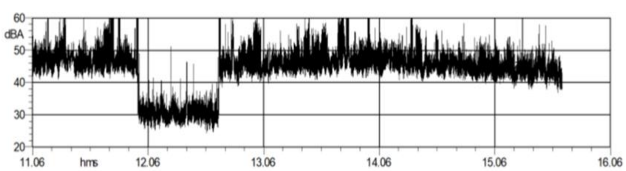

Figure 1: Time history of the sound pressure level.

When the wind turbine is off, the residual noise level, measured inside the home, is equal to 30 dBA, while when the wind turbine is on, the measured sound pressure level is equal to 45-50 dBA. A neglected aspect regarding the emission of noise by small wind turbines with a cylindrical tower is that the noise emitted is not only due to the rotation of the blades. The noise emission is a complex phenomenon, also emitted by the entire tubular structure of the tower put into vibration by the rotation of the blades. The theoretical simulation model therefore underestimates the noise level inside home. Near the wind turbine the following problems were observed: the blade rotation speed is about 30 rpm; the rotation of the blades is discontinuous, it depends from the instantaneous wind speed, and this generates an intermittent noise. Intermittent rotation with a speed of about 30 rpm cancause damage to the blade elements. Finally, it was considered the theoretical relationship for the evaluation of the noise introduced in the home due to the functioning of a tower, applying the ISO 9613 standard. In the hypothesis of point-sound source, it was found that the theoretical relationship gives a value of the sound pressure level of Lp = 37 dBA. Therefore it underestimates the measured value of the sound pressure inside the house. From the comparison, a difference of 13 dBA between measured and calculated values was obtained.

Table 1. Acoustic measurement results.

Time of the day	Environmental noise dBA	Residual noise dBA	Background noise L95 dBA	Average wind speed m/s
19:00 – 22:00	42.4			8
06:20 – 09:00	44.0			9
16:00 – 17:30	46.1			10
18:00 – 22:00	40			8
22:00 – 06:00	40			8
06:00 – 9.30	44.8			8
23:00 – 04:30		30.0	21.3	9
06:00 – 13:00	50.2			15
22:45 – 01:00	40			10
01:30 – 04:30		31.0	20.7	10
01:00 – 03:30	41.0			8
03:40 – 04:10		31.5	27.6	8
21:40 – 22:00	39.0			9
01:40 – 04:40		30.5	24.0	8
06:00 – 10:00	47.6			12
11:10 – 12:00	49.4			13
12:00 – 12:30		31	28	10
12:30 – 15:45	47.0			12
14:00 – 20:00	46.7			12
11:15 – 15:15	46.6			11
19:45 – 22:00	43.7			12
22:00 - 23:45	37.0			5
19:45 – 22:00	43.3			11
22:30 – 01:00	41.2			9
00:00 – 04:30	40.0			9
06:00 – 16:00	44.0			12

5 Conclusions

In this study, we report the acoustic measurements of the noise produced by the operation of a wind turbine with a nominal power equal to 200 kW performed inside a house. In the house, the wind tower increases the sound level of about 10 dBA. The theoretical model for the evaluation of the noise introduced in a house due to the functioning of a tower underestimates the measured value of the sound pressure level.

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STUDENTS PECHA-KUCHA PRESENTATION - PRÉSENTATIONS ÉTUDIANTES PECHA-KUCHA

"listen To Your Heart": Exploring The Link Between In-Ear Audio And Emotions

Danielle Benesch, Corentin Delain, Rachel Bouserhal, Stefanie Blain-Moraes, Jérémie Voix

Wearing passive hearing protection to manage auditory sensitivities can sometimes lead to the opposite end of the spectrum where the surrounding sounds are over-attenuated and wearers can no longer fully engage with their surroundings. A "smart" hearing protection device could satisfy both needs by attenuating distressing sounds while relaying useful sounds. And since each individual may be sensitive to a unique constellation of sounds, this device would ideally be customized to the needs of each wearer. A hearing protection device could conceivably adapt to the idiosyncratic sensitivities of individual wearers by monitoring their biosignals to detect sound-induced distress. When the earcanal is acoustically sealed by a hearing protector, biosignals can be captured due to bone-conduction sound amplification or the "occlusion effect". The amplified biosignal sounds can be recorded with an in-ear microphone and classified automatically. Among the audio events that can be extracted from in-ear microphone data are heartbeat signals. Drawing upon previous psychophysiological research using heart rate and heart rate variability to study emotions, this project proposes inferring an individual's affective state based on the heartbeat extracted from in-ear audio. To assess the suitability of the in-ear heartbeat as an indicator of affective state, stimuli intended to evoke certain emotions are played while audio is recorded inside the ear. Preliminary results are discussed, as well as the feasibility of a potential application: the automatic classification of distressing sounds by a hearing protection device.

In-Ear Speech Capture On Electronic Hearing Protectors: Optimization Of The Denoising Algorithm In Transparent Mode

Corentin Delain, Farid Moshgelani, Rachel E. Bouserhal, Jérémie Voix

Providing industrial workers with satisfactory hearing protection and communication in noisy environments is still a challenge and often requires a compromise of one or the other. An intra-aural hearing protection device (HPD) equipped with an in-ear microphone (IEM), miniature loudspeaker, and an outer ear microphone (OEM) has been recently developed for communication in noise. The voice of the wearer is captured from inside the occluded earcanal and the speech signal is further denoised from residual ambient noise and subsequently enhanced to increase its frequency bandwidth. Currently, when the ambient noise level is greater than 85 dBA, the denoising of the IEM signal is performed in a so-called "noise isolation mode", where the HPD provides full passive attenuation of ambient noise. However, in moderate ambient noise levels, it is desirable to maintain the situational awareness of the wearer by allowing ambient sounds captured by the OEM to be played through the internal loudspeaker (electronic "talk through"). This mode of operation is herein referred to as "acoustical transparency mode" and creates added challenges, as the IEM picks up the voice of the wearer as well as the ambient noise played back by the loudspeaker inside the occluded earcanal. The present study details the denoising approach developed around the use of adaptive filters. The effects of loudspeaker gain, ambient noise level, and fit of the HPD on the transparency mode algorithm are investigated to optimize its performance and integration.

Hpd Fit-Testing Feature Developed Within A Hearing-Care Platform For Musicians

Lucas Einig, Romain Dumoulin, Isabelle Cossette, Jérémie Voix

The hearing-care platform is a dedicated software and hardware solution intended for university music students to assess an individual's noise exposure and to promote hearing health. As the level of exposure is measured using a calibrated microphone set on a smartphone app, the attenuation provided by the -possible- use of a hearing protection device (HPD) needs to be taken into account in the calculation of the protected noise exposure. To quickly estimate the attenuation provided by a HPD, a new app functionality has been developed featuring a) an audio stimulus generator – that can produce loud tones through the smartphone embedded loudspeakers, b) a graphical user interface that can display a count of the audio stimuli perceived, and c) an attenuation prediction algorithm that can estimate the overall attenuation of the HPD under test. The proposed measurement approach relies on a threshold-based method where sequences of octave band-centered narrowband stimuli are played in steps of 5 dB. The user simply counts the number of tonal bursts perceived before stimuli become inaudible in two conditions: when both ears are occluded with the HPD and when they are not occluded. From the two count values, the HPD octave-band attenuation is computed. Attenuation data can be assessed for different HPDs and applied to each activity exposure with an adjustable wearing time (from no HPD to HPD being worn during the entire activity). In addition to improving individualized assessment, this feature is also a great educational tool to demonstrate the benefits of HPDs.

Evaluating The Accuracy Of Lip Motion Tracking Using Surface Scanning Face Tracking Technology

Roujan Khaledan, Andrew (U San) Chao, Caroline Jeffery, Gabriela Constantinescu, Daniel Aalto

Speech-Language Pathologists diagnose Motor Speech Disorders (MSD) by visually inspecting differences in facial motion, specifically range of motion and trajectory of the lips. Visual inspections are subjective, making it difficult to precisely track movement. Digitally tracking facial motion has the potential to refine the diagnostic process, characterize MSD, and monitor treatment impact. In contrast to traditional motion tracking systems, which are only available in specialized centers, mobile devices and the integrated facial tracking systems are ubiquitous and offer an accessible alternative to traditional systems. The goal of the present work is to evaluate the accuracy of lip tracking based on mobile surface scanning technology. Using a custom app based on Apple's augmented reality environment (ARKit), one of the authors recorded their facial movement for both natural speech and extreme non-speech gestures. This app tracks 1220 mesh points on the face while recording video (both at 60 frames per second). The TrueDepth iPhone camera was placed 55 cm from the speaker and positioned at their eye level. Accuracy was defined as the distance between a visible anatomical landmark on the lip (flesh point) and the corresponding tracked mesh point. For each gesture, the deviation of the flesh point from the mesh point was calculated from a single frame. Understanding the accuracy of the app for different tasks can have an impact on future clinical and research use, and it may motivate the development of an automated adjustment for the flesh point-mesh point deviation.

Direct Manipulation Of Variability In The Auditory Feedback System Via Real-Time Formant Perturbation

Daniel Nault

Auditory feedback is an essential part of speech motor control and speech learning. When feedback is perturbed in laboratory settings (e.g., Houde & Jourdan, 1998), speakers, on average, compensate for the perceived error. There is, however, considerable individual variability in the magnitude of speakers' responses to feedback manipulation (e.g., Purcell & Munhall, 2006). Here, we address one potential source of this variability by manipulating the predictability of auditory feedback of 20 female speakers using a real-time formant perturbation system. Participants produced the English word "head" 80 times in 3 different conditions. During the Perturbation phase of each condition (i.e., trials 20-50), subjects were presented with random first and second formant perturbations that produced feedback varying between the vowels in "hid" and "had". Predictability of the perturbations in the conditions varied, such that: (a) on each trial, a different perturbation was introduced, (b) perturbations were consistent for three trials, or (c) six trials. Time series analyses were performed to examine whether compensatory behavior differed among speakers in the three conditions of varying feedback predictability. Results will be discussed regarding the possible role of variability in speech motor control and the importance of developing methods to detect state-change in individual time-series data.

Transfer Of Training Across Speakers And Languages In Learning Time-Compressed Speech

Cynthia Sedlezky

A key criterion for successful learning is generalization beyond the training conditions. Here we examine the perceptual learning of time-compressed speech and study generalization to speech produced by different talkers in different languages. This research tests the importance of learning low-level acoustic properties of speech for transfer of training. Monolingual and bilingual adults were trained in English over 8 sessions using passages of progressively more time-compressed speech from a novel. The speech rate during training sessions corresponded to their threshold of intelligibility as assessed by scoring the SPIN sentences recorded at 14 rates (100-425%). To assess transfer of their ability, a similar task with English or French sentences produced in the same or a different voice was tested at the end of the 8 sessions. Compared to a control group that underwent only baseline training (i.e. sentence rate fixed at 100%), both the monolingual and bilingual experimental groups achieved higher scores following training when tested on their ability to understand time-compressed speech in both English and in French. However, both voice and language changes produced reductions in performance. The results suggest that perceptual learning of time-compressed speech involves low-level acoustic cues that are independent of the language of training.