ABSTRACTS

[144] *Blind Localization of Room Reflections with Application to Spatial Audio*

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Estimating the direction and delay of room reflections from audio signals captured by microphone arrays is crucial for various applications, such as speech enhancement, dereverberation, source separation, room geometry estimation, and optimal beamforming. However, this is a challenging problem, particularly when the estimation is performed blindly, directly from the recorded signals, such as reverberant speech. Most existing methods can only detect a limited number of room reflections. Recently, a novel method called PHALCOR has been proposed, which uses the phase-aligned transform on the signal correction matrix to estimate the parameters of multiple early room reflections, thereby providing superior performance. However, this method is restricted to spherical microphone arrays, which limits its practical applicability.

An extension of the mathematical formulation of PHALCOR was recently presented to overcome the limitation related to array geometry. This extension employs frequency-focusing to achieve frequency independence of the steering matrix, enabling support for microphone arrays of arbitrary configurations. This method achieves frequency-focusing by multiplying the steering matrix with a focusing matrix for each frequency in the band. The method was investigated compared to PHALCOR by studying the number of detections for each method. It was shown that these two methods achieved similar performance for small rooms, but the extended method detected slightly fewer reflections compared to the original algorithm. Although initial results were presented for the extended method, no comprehensive performance analysis was shown, and no insight was provided for its limits of performance.

This study presents an advance by evaluating the algorithm's performance in a wider range of conditions. Additionally, performance in terms of perception is investigated through a listening test. This test involves synthesizing room impulse responses from known room acoustics parameters and replacing the early reflections with the estimated ones. The importance of the estimated reflections for spatial perception is demonstrated through this test.

[194] *Estimation of Diagonal Volterra Kernels of an Audio System During Normal Operation with Multiple Least Mean Squares Adaptive Filters*

Daniel Pinardi (University of Parma), Angelo Farina (University of Parma), Marco Binelli (University of Parma) and Andrea Toscani (Università di Parma).

The usage of Complete Volterra Kernels for emulating the nonlinear behavior of sound systems has been investigated for decades. Due to the computational load, the real-time implementation is typically limited to the second distortion order and not feasible for higher distortion orders. This is usually unsatisfactory for audio systems in which the disturbing distortions occur mostly at orders three and five. The same authors of this work already solved the problem with the Diagonal Volterra Kernels technique, which allowed to model arbitrarily high distortion orders. The estimation of the coefficients was obtained by exciting the system with an Exponential Sine Sweep signal. However, the result was often suboptimal since the signal reproduced by the sound system is usually different from a sinusoid. In this paper, a new method for estimating the Diagonal Volterra Kernels coefficients is proposed, by employing a signal being played by the sound system in real-time. Multiple Least Mean Square algorithms are used to estimate the coefficients up to the 5th distortion order, thus allowing to emulate the nonlinearities of a typical audio system in real-time.

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[350] *Soundscape as a Tool for Place-Making in Industrial Heritage Sites*

Lei Sun (Università di Bologna).

Soundscape plays a crucial role in place-making at industrial heritage sites, enhancing visitors' understanding of the site's heritage value. This study investigates public attitudes towards industrial heritage site soundscapes and analyzes the impact of sound-based place-making, using the Museum of Industrial Heritage in Bologna, Italy as a case study. Field research and questionnaires were employed to gather data. Findings reveal a positive reception among the public towards soundscape design in heritage sites, highlighting its numerous advantages. Soundscapes not only enhance visitors' perceptions and attractiveness of heritage sites but also establish a sensory connection between individuals and the heritage environment. Moreover, soundscapes enrich the visitor experience by simulating authentic sounds, creating an immersive atmosphere, and deepening the sense of presence. To facilitate effective soundscape design, a design strategy for industrial heritage sites is proposed. It emphasizes public perception, interactive elements, control of decibels and disturbances, and the creation of rich and realistic soundscapes. The empirical analysis underscores the positive influence of soundscapes on place-making in industrial heritage sites, reinforcing the connection between tangible and intangible heritage while preserving authenticity and integrity.

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[356] *Evaluation of noise annoyance of urban noise in Singapore using web-based subjective listening tests*

Prachee Priyadarshinee (Singapore University of Technology and Design), Balamurali B T (Singapore University of Technology & Design), Christopher Clarke (Singapore University of Technology and Design), Ivan Fu Xing Tan (Singapore University of Technology & Design), Johnny Lee (Singapore University of Technology and Design) and Jer Ming Chen (Singapore University of Technology & Design).

A large population of the densely populated city-state of Singapore resides in public housing, living in close proximity to each other as well to MRT, highway, bus routes, and construction sites. Several thousands of noise complaints are received each year. To investigate the short-term annoyance of the residents due to noise from in and around the public housing, we designed an easily accessible web-based subjective listening test. The sampled audio were composed of noise sources generally encountered by the residents such as traffic (bus, heavy vehicles, highway), trains (MRT/LRT), aviation, neighbourhood (playground, school, hawker centre, funeral/wake, wildlife, home renovation), and construction. Listeners were asked to imagine different times of the day and subjected to 35 noise stimuli (15-30 sec long). They were asked to rate their perceived annoyance for the loudness normalized stimuli on a 5-point rating scale, 5 being most annoying. The purpose of this study was to relatively rank the noise stimuli and investigate the effect of time of the day on the reported annoyance. Anchoring using the most annoying and the least annoying sounds were initially introduced. This study offers insight into the assessment of the perceived annoyance, which is a critical step toward the development of systems to offer improved quality of life, comfort, and productivity.

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[643] *Personalized Spatial Audio Tools for Immersive Audio Production and Rendering*

Kaushik Sunder (Embody), Saarish Kareer (Embody), Scott Murakami (Embody) and Axel Borgmo (Embody).

In this paper, we present a set of personalized spatial audio production tools for immersive applications. Our tools consist of a 3D Panner, Visualizer, and a Virtual Studio Renderer with personalized spatial audio. We discuss how these tools can be used for the production and rendering of immersive and 3D audio. We explore different formats, and standards, for the delivery of spatial audio, and how they can be integrated with these tools. Finally, we discuss the design and implementation of interactive audio frameworks and applications, which can be used to further enhance the immersive experience. Camera based Head Tracking tools are also discussed to improve the spatial audio experience.

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[659] *Preliminary evaluation of a sound system employing a cancelling auralizer for producing virtual rooms in a modern recording studio*

Gianluca Grazioli (McGill University), Vlad Baran (McGill University), Kathleen Ying-Ying Zhang (McGill University), Aybar Aydin (McGill University) and Wieslaw Woszczyk (McGill University).

We investigate the operation of a virtual acoustic system (VAT) in the Immersive Media Laboratory (IMLAB) at McGill University, and the benefits offered by a real-time cancelling auralizer currently under development at CCRMA of Stanford University and the University of Limerick. The feedback-cancelling system reduces the likelihood of feedback and howling in live real-time auralizations of acoustic spaces through in-room microphone capture and loudspeaker reproduction. Amplified microphone signals are sent to a feedback cancellation network that convolves them with a variety of captured or synthesized multichannel impulse responses, and reproduces them via loudspeakers. One of the research purposes of VAT is to investigate the degree of similarity that can be achieved between the acoustics of a real space and its recreated virtual version. We present a preliminary analysis focused on measuring the main acoustical parameters according to ISO 3382-1 created under different virtual acoustical conditions. Ongoing research examines the interactions of professional musicians with the current state of the system, which will be continuously investigated in the near future. The work is carried out in collaboration with the authors of the feedback-cancelling technology.

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[669] *Exploring the past with virtual acoustics and virtual reality*

Nima Farzaneh (Stanford University), Eito Murakami (Stanford University), Jonathan Berger (Stanford University) and Luna Valentin (Stanford University).

Recent novel work integrating virtual acoustic au- ralization with architectural models implemented in virtual reality (VR) allow scholars of the humanities and sciences to reimagine and reconstruct sites and to reanimate and contextu- alize situations that are no longer in extant. By introducing the audio domain, researchers are creating new hypotheses regarding everyday life and ritual practices in ancient societies. These methods are proving to be vital in advancing research as well as making the past more accessible. Here we outline methods and emerging practices. We demonstrate the potential of auralization and VR with two case studies: Chauvet-Pont-D'Arc, a cave sanctuary for paleolithic art in the Arde`che valley in France, and a funerary tomb from the third dynasty in Saqqara, Egypt. We demonstrate how spatial audio in immersive simulations offers researchers the ability to create and test novel hypotheses about the sites and structures they study. We further describe the creation of an auralization and immersive model of a church in Venice, Italy and a soundscape in Isfahan, Iran. We propose that these examples demonstrate the enormous potential in incorporating immersive simulations and describe methods that offer new perspectives and critical anthropological knowledge to humanities research.

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[722] *From prayer to music: acoustic studies of a theatre in Crema realized inside the St Dominic church*

Ruoran Yan (Department of Architecture, Bologna), Giacomo Tentoni (Gruppo CSA SpA Rimini), Alessandro Martinetti (Gruppo CSA SpA Rimini) and Van Tonder Cobi (Department of Architecture, Bologna).

The medieval church honored St Dominic has an interesting story. Adjacent to the convent of the white friars, a church has been built from the pilgrims' donations arriving in Crema. During the French invasion, the church was transformed into a military barrack where soldiers were used to staying during the war of the 18th century. After the unification of Italy, the church fell onto the property of the municipality and was used as a laic public place, hosting local markets and

becoming an educational building other than a gym for citizens. The church was transformed into an auditorium due to the lack of performance arts space in the city. Acoustic measurements have been carried out inside the church before the transformation works, according to ISO 3382. The analysis of the measured data has been compared to the optimal values required for an auditorium where classical music is the dominant style of the new room function. A brief description of the history of the church has been added to better understand the determination of the geometry and the interior material decorating this cultural heritage.

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[920] *How to Spatial Audio with the WebXR API: a comparison of the tools and techniques for creating immersive sonic experiences on the browser*

Matteo Tomasetti (University of Trento, Department of Information Engineering and Computer Science), Alberto Boem (University of Trento, Department of Information Engineering and Computer Science) and Luca Turchet (University of Trento, Department of Information Engineering and Computer Science).

The WebXR Device API provides a powerful set of functionalities that can be used for creating immersive experiences that can be accessed directly from a web browser. However, while creating a compelling visual experience is easy to implement using the WebXR API and related tools, creating the audio part of the experience is less clear and with more open questions and possibilities. This paper presents a comparison of six tools oriented to practitioners and used for implementing and delivering interactive spatial audio experiences in XR applications running on a browser. These tools are selected and evaluated based on their compatibility with the WebXR Device API, their ease of use, the provided support and documentation, as well as specific features like reverberation, head-related transfer functions (HRTFs), sound scene rotation, and real-time microphone input. The paper then discusses the integration of these spatial audio tools into WebXR applications, highlighting their potential and limitations. It also addresses the absence of standardized spatial audio frameworks for the web. The challenges of choosing the right tool in terms of resource allocation, learning curve, and compatibility with XR systems are acknowledged. The aim of this overview is to provide insights into the current state of spatial audio tools for the web, making them more accessible to developers, musicians, and researchers seeking guidance on selecting suitable 3D spatial audio tools for experiences based on WebXR.

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[993] *Latency of spatial audio plugins: a comparative study*

Matteo Tomasetti (University of Trento, Department of Information Engineering and Computer Science), Angelo Farina (University of Parma) and Luca Turchet (University of Trento, Department of Information Engineering and Computer Science).

The use of spatial audio plugins (SAPs) with Ambisonics processing and binaural rendering has become widespread in the last decade, thanks to their increased accessibility and usability. SAPs are particularly relevant in scenarios involving real-time music playing with headphones, such as networked music performance and individual recreational music-making using backing tracks. However, a crucial issue that has been largely overlooked thus far is the measurement of the processing latency introduced by currently available SAPs. Identifying which SAPs are the fastest is essential to enable designers, musicians, and researchers to create time-sensitive applications involving 3D audio. To bridge this gap, we compared seven systems formed by different SAPs that enable 3D audio management. We measured the latency of each system throughout the third-order Ambisonics plugins pipeline: encoding, room simulation, sound scene rotation, and binaural decoding. In particular, the measurements were performed utilizing different buffer sizes. Results showed that to achieve a minimization of the latency, it is necessary to use a combination of different SAPs from different systems. Based on our measurements, we propose two spatial audio systems that mix different SAPs. Considering a sampling rate of 48 kHz, a Dell Alienware x15 R2 laptop running the Windows 10 operating system, and an RME Fireface UFX sound card, the two systems achieved an overall latency of 0.33 ms and 0.94 ms respectively.

[1113] *Instrument Position in Immersive Audio: An Empirical Review of Award Winning Practices*

Afonso Lopes (Faculty of Engineering of the University of Porto), Gilberto Bernardes (Faculty of Engineering of the University of Porto and INESC TEC) and José Ricardo Barboza (Faculty of Engineering of the University of Porto).

Immersive audio technologies have broadened post-production strategies for spatial audio, gaining popularity among mainstream audiences. However, there is a lack of defined procedures and critical thinking regarding audio mixing guidelines for surround sound in popular music. In this context, we conducted an empirical study to identify trends concerning instrument position, trajectories, and dynamics from surround mixings. Furthermore, we assess the degree to which they differ from their stereo renderings. Seven award-winning songs in the Grammy category for Best Immersive Album were analyzed, including surround 5.1 and stereo versions. The study found consistent instrument positions in the songs, with rhythmic instruments and bass in the center, lead vocals spread across front channels, and harmonic instruments in wider positions. Solo instruments occupied left, right, and center channels, with dynamics emphasizing lead vocals and solos. Trajectories were rarely used, indicating channel-based thinking. Limited adoption of immersive audio dimensions and reliance on stereo techniques were observed, with no notable differences between the surround and stereo versions. Identified song outliers are discussed and offer avenues for exploration, highlighting the importance of diverse musical expressions in informing immersive audio mixing.

[1241] *The Complex Image Method for Simulating Wave Scattering in Room Acoustics*

Orchisama Das (University of Surrey) and Enzo De Sena (University of Surrey).

The Image Method (IM) has become increasingly popular for small-room acoustics simulations. While it gives an exact solution of the wave equation in shoebox-rooms with rigid walls, the assumption of rigidity is not valid in real rooms. Based on spherical wave reflection from an infinite wall, several authors have independently developed what is known as the Complex Image Method (CIM). However, its adoption in room acoustics has been rare, although it has been shown to give performance equivalent to the boundary element method in shoebox rooms with soft-walls. In this paper, we revise the theory behind CIM, starting from the reflection of a spherical acoustic wave from a non-rigid infinite plane. Then, we study directional scattering patterns as a function of wall impedance. Finally, for a highly symmetrical shoebox room, we show that room impulse responses simulated with CIM have less sweeping echoes than those simulated by IM. We also provide a Python package to enable widespread testing and adoption of the complex image method.

[1271] *Recovering the intangible acoustic heritage of rock art sites: El Tajo de las Figuras as case study*

Lidia Alvarez-Morales (Departament d'Història i Arqueologia, Universitat de Barcelona, Spain), Neemias Santos da Rosa (Departament d'Història i Arqueologia and Institut d'Arqueologia, Universitat de Barcelona, Spain), Daniel Benítez-Aragón (Departament d'Història i Arqueologia, Universitat de Barcelona, Spain), María Lazarich (Departamento Historia, Geografía y Filosofía, Universidad de Cádiz, Spain) and Margarita Díaz-Andreu (Departament d'Història i Arqueologia and Institut d'Arqueologia, Universitat de Barcelona, Spain; ICREA, Spain).

Since the end of the 19th century, scholars have highlighted the importance of rock art as a priceless heritage that may provide clues about the cultural and spiritual practices of prehistoric societies. However, most studies developed so far have focused only on the materiality of such archaeological remains. In this sense, the ERC Artsoundscapes project – funded by the European Research Council (H2020 programme, Grant Number 787842)– aims at exploring the intangible dimension of rock art sites in relation to auditory experience. Through a multidisciplinary approach comprising archaeological research, ethnomusicology, impulse response measurements, psychoacoustic tests based on auralisations and neuropsicological tests, the project investigates the role of sound and emotion in relation to the sacred by recovering the acoustic heritage of rock art sites in several rock art landscapes of the world. In this paper, we present a case study centred on the El Tajo de las Figuras (Cádiz, Spain), one of the most remarkable rock art sites of the Iberian Peninsula. The results indicate the presence of reverberation, which is especially relevant when considering its small size and, that the site is partially open. This, along with the great strength values obtained, suggests that the shelter morphology could have contributed to creating an acoustic environment suitable for reproducing certain types of music and rhythms that prehistoric people might have used in ceremonies and rituals potentially developed in this decorated space.

[1297] *An Evaluation of the Presentation of Acoustic Cues for Shorelining Techniques*

Minh Nguyen (University of Technology, Sydney), Howe Zhu (University of Technology, Sydney), Lil Deverell (The University of Sydney), Craig Jin (The University of Sydney), Chin-Teng Lin (University of Technology, Sydney), Vincent Nguyen (University of Technology, Sydney), Avinash Singh (University of Technology, Sydney) and Huiyuan Sun (The University of Sydney).

The study explores various scenarios for providing shorelining information via spatial-audio auditory sensory augmentation with a view to assisting people who are blind or have low-vision. Shorelining is a strategy often associated with the use of the "white cane" and generally refers to path-following approaches that rely on contours of structures within a built environment. The aim is to support clear, logical paths of travel. In one approach, auditory shorelining cues are provided via a sequence of two auditory earcons rendered spatially along the shoreline and at the participant's left or right side. In another approach, the auditory cues are rendered spatially as "gate posts" in between which the participant should walk. The rendering of the auditory earcons follows the participant's path of travel so that at any given time, there is only ever a sequence of two auditory earcons that provides local shorelining information. The auditory earcons are rendered via two methods: (i) using loudspeakers appropriately placed along the path of travel and (ii) using binaural rendering of virtual speakers via smart glasses. We compared the performance of spatial-audio sensory augmentation with non-spatialized spoken language instruction. Participants' performance was measured in terms of task accuracy, time, and walking behavior. The cognitive workload was measured using a self-reporting questionnaire. Results indicate that sensory augmentation provided via spatial-audio earcons can provide faster and smoother navigational assistance and guidance compared with spoken verbal instructions. The results also suggest that current binaural rendering using simple augmented-reality tool sets is not as robust as real sound in providing navigational guidance. Lastly, the "gate post" shorelining approach seems more effective than the left and right shorelining techniques.

[1371] *Computer-aided acoustic design of concert halls*

Benoit Beckers (Urban Physics Joint Laboratory), Inès de Bort (Urban Physics Joint Laboratory) and Jairo Acuña Paz Y Miño (Urban Physics Joint Laboratory).

Twenty years ago, the Radit2D program offered acousticians an interactive tool to design with the first reflection the shape of a room cross-section. Based on the method of images, it was enough for the series of straight-line segments drawing the enclosure to follow a regular curve for its orthotomy to appear by connecting the points of the image. This new curve quickly appeared as a guide, especially when it was necessary to direct the sound energy towards a particular area of the room, without focusing or dispersing it. We then studied the logarithmic spiral, the best candidate to assume this role, typically for the acoustic ceiling of a large conference room. Recent advances in ray tracing techniques, with extremely fast shots computed directly in the CPU, now allow us to provide a new representation of the set of specular reflections in three dimensions, not only for a particular receiver chosen from the public, but also, and simultaneously, for the entire public concerned. This should make it possible, for example, to find the ideal shape of a concert hall, as complex as it may be, which guarantees for the whole audience the best possible compromise between reverberation and the Lateral Energy Fraction.

[1379] *Measurements of Room Acoustic with Two Different Methods - A Case Study*

Ruoran Yan (Univerity of Bologna), Cobi van Tonder (University of Bologna) and Lamberto Tronchin (DA - CIARM).

Room acoustics has always been an important study in indoor environmental science, and the investigation of its acoustical characteristics is an indispensable basis for subsequent research. In this paper, a conference room located in the engineer's office of Treviso province was investigated. The experiments were conducted firstly with an omnidirectional microphone and a dummy head, from which the traditional values of the main acoustic parameters were collected for the further data analysis; followed by a spherical array microphone and a 360° camera were utilized for the measurements that ensued, and a video of the real-time room impulse response was obtained, and numerous shots were used instead to visualize the sound reflections taking place across the entire volume.

[1450] *Conversion of a church into a concert hall: discussions about concerns and acoustic design solutions*

Gino Iannace (University of Campania), Luigi Guerriero (University of Salerno), Umberto Berardi (Toronto Metropolitan University – Canada,), Giovanni Amadasi (SCS-ControlSys - Vibro-Acoustic, Padova, Italy) and Silvana Sukaj (Department of Engineering and Architecture, European University of Tirana).

Increasing demand for performance spaces is prompting experts to develop design projects to adapt non-purpose existing buildings for music performances. Local authorities also try to include ecclesiastical buildings in the process of changing their use, since the heritage of this type of building in Italy is very rich and belongs to different architectural styles. These objectives lead to the challenge for the acousticians to meet the criteria for a volume of space suitable for conferences and/or classical music events. The church of the Holy Spirit in Aversa, located in southern Italy, is one of the historic and artistic buildings to be improved. Built in the 18th century, the church was destroyed by the 1980 earthquake. The damage led to a slow collapse of the roof and complete neglect for decades. Restoration works began in 2021 with the reconstruction of a new roof and the restoration of the floor and walls. This paper deals with the acoustic design project of the Holy Spirit Church of Aversa, which includes the construction of a stage in place of the altar and the installation of absorbing panels to control reverberation. Acoustic measurements were carried out without audience, before and after each acoustic design application. Acoustic measures have been predicted using simulation tools before being applied in situ.

[1525] *Reverberation time prediction using diverse models*

Nicola Granzotto (Progetto Decibel S.r.l.) and Marco Caniato (Free University of Bozen-Bolzano).

In present days, the indoor spaces need more and more attention on their correct acoustic design of enclosed environments such as classrooms, conference rooms and offices. In order to ensure high speech intelligibility and to contain internal noise levels, indoor sound field should be studied properly. In light of this, the inclusion of sound-absorbing ceilings could not be the solution to reach indoor comfort, because the reflections effect caused by parallel walls. When one tries to predict the indoor sound filed, numerical models such Sabine or Eyring models could lead to an underestimation of the reverberation times. In this work, we compare the reverberation time measured and calculated in two small rooms using (i) Sabine and Eyring models, (ii) one commercial simulation software and (iii) the EN 12354-6 standard procedure. Results clearly show that calculated results are not in agreement and differ from measured values.

[1527] *A Generic Reverberation Characterization Metric for Accurate Simulation in Virtual and Augmented Reality Environments*

Jeroen Koppens (Koninklijke Philips N.V.,), Patrick Kechichian (Koninklijke Philips N.V.) and Sam Jelfs (Koninklijke Philips N.V.,).

Reverberation is one of the key audio effects for producing highly immersive experiences in virtual acoustic environments (VAEs). Yet, traditional architectural acoustic metrics are not always suitable for characterizing reverberation for such environments. Metrics such as the Direct-to-Reverberant ratio and critical distance are well known to acousticians and sound engineers, but their relation to the excitation of the room is achieved through the direct path energy of the sound from a source to a receiver. That implies that these metrics are heavily dependent on the receiver's position relative to the source. This dependency is specifically problematic when the user and/or sound sources) can move freely throughout the VAE, such as during rendering for VR and AR. After all, given an excitation of the environment, the reverberation level, by definition, is consistent across the environment and independent of the user's position. A generic metric for reverberation level must therefore relate it to the excitation of the room.

To characterize a VAE in a generic way, a new metric is required that characterizes a reverberation level in relation to the excitation, yet independent of the source and receiver positions. The Diffuse to Source Ratio (DSR) metric is introduced that was developed in the context of MPEG-I immersive audio, an upcoming audio standard for efficient transmission and storage of 6DoF audio. It links the reverberation energy to the total radiated sound source energy.

[1799] *Sound field interpolation via sparse plane wave decomposition for 6DoF immersive audio*

Orhun Olgun (Middle East Technical University (METU)), Ege Erdem (Middle East Technical University (METU)) and Hüseyin Hacıhabiboğlu (Middle East Technical University (METU)).

Volumetric capture of audio for six-degrees-of-freedom (6DoF) reproduction requires recording a sound field at multiple positions using microphone arrays. When microphone arrays capable of recording higher-order Ambisonics (HOA) are used, rendering of 6DoF audio becomes an interpolation problem that involves the calculation of HOA signals at a location intermediate to the original recording positions. We present a sound field interpolation method using multi-point sparse-plane decomposition followed by directional interpolation. The proposed method operates in the time-frequency domain and relies on the decomposition of time-frequency bins into a dominant directional component and a residual which are interpolated separately. The directional component which represents the dominant sound source is interpolated by translation of the associated plane wave components calculated for each microphone array to the interpolation position and calculation of a single, interpolated plane wave. We present an objective validation of the method based on the directional statistics of the interpolated sound field.

[1822] *Application of different types of microphones in room impulse response measurements.*

Maciej Jasiński (Warsaw University of Technology) and Jan Żera (Warsaw University of Technology).

The features of the acoustic field in a room are generally described using omnidirectional and spatial parameters. Those are determined from the recorded Room Impulse Responses (RIR). Standardized RIR recording methods are described in ISO 3382 and require three different types of microphones. An omnidirectional microphone to determine the RIR at a given point in the room and bidirectional or binaural microphone to get RIRs containing the spatial properties of the acoustic field. Instead, popular but not standardized ambisonic technology makes it possible to measure a Spatial Room Impulse Response (SRIR). Signal processing technology lets to obtain omnidirectional and binaural RIRs from an SRIR. In this study, RIRs were recorded using the following types of microphones: omnidirectional, binaural (2 different types of acoustic manikin) and ambisonic (first and third order). Presented results show the comparison of room acoustic parameters calculated from RIRs measured with different microphone techniques. The results show ambisonic technology's usefulness in assessing room acoustic parameters.

[1992] *Spatial Audio Production with a New Volumetric Amplitude Panning and Diffusion Technique*

Saarish Kareer (Embody) and Kaushik Sunder (Embody).

The world of spatial audio production is expanding rapidly with multichannel and Dolby Atmos formats becoming increasingly popular in music, cinema, and gaming. To keep up with the pace, it's crucial to have tools that accurately represent sound objects in 3D, are intuitive, and effectively translate ideas into perceptual reality. A 3D panner is an essential tool that accurately places a sound source or an object in any point of the 3D space. However, traditional techniques such as Vector Based Amplitude Panning (VBAP) require the listener to be positioned at a "sweet spot" surrounded by loudspeakers on either a 2D ring or a 3D sphere, which limits their effectiveness. Our research presents a novel algorithm for volumetric amplitude panning that does not assume any sweet spot and can work for any loudspeaker layout, both asymmetric and symmetric. Additionally, we've included geometric and distance-based diffusion techniques that guarantee smooth spread across all directions, distances, and diffusion amounts. With this algorithm, we can improve the accuracy and efficiency of sound object placement and enhance the overall spatial audio experience.

[2028] *Enhancing Object Audio Control within an Immersive Sound System*

Richard Foss (Rhodes University) and Lukas Klingebiel (Coda Audio).

Real time control over sound source positioning and movement is a primary advantage of object-based sound systems over channel-based systems. This paper describes a controller, processing engine, and associated library that enhances this object control. Apart from graphic spatialisation control, the controller allows, for each object, the selection of its rendering algorithm, distance-based level, high frequency attenuation, reverb parameters and speaker isolation or locking. OSC control within the controller enables further control options.

[2061] *Effects of the types of headphones and sound sources on spatial audio quality*

Yoshiharu Soeta (National Institute of Advanced Industrial Science and Technology (AIST)) and Takanobu Nishiura (Ritsumeikan University).

Acoustic characteristics of sound fields reproduced by headphones are a major concern to evaluate spatial audio quality. There are several specifications for acoustic characteristics, such as frequency response, impedance, and sensitivity. Although previous studies clarified the acoustic characteristics, they are not enough to evaluate the spatial audio quality reproduced by the headphones. They do not provide enough information about the effects of headphones and sound sources. In this study, we carried out psychological evaluations and acoustical measurements of sound fields reproduced by headphones to find the acoustical factors that explain the spatial audio quality reproduced by the headphones. Three types of sound excerpts were played with seven types of headphones, and the psychological evaluations were carried out by 30 participants. Envelopment and apparent source width were evaluated as psychological evaluations. The early to late-energy ratio, the center time (Ts), the maximum amplitude (IACC) of the interaural cross-correlation function (IACF), and the width of the IACC (WIACC) were calculated from the binaural impulse response. The IACF and autocorrelation function (ACF) factors were calculated from the recorded sound sources. The characteristics of the headphone and the sound source affected the psychological evaluations. The difference in spatial audio quality among headphones and sound sources was explained by the amplitude of the first maximum peak of the ACF, Ts, and WIACC.

[2117] *A study on the spatial sound propagation characteristics of the Argentina Theater in Rome*

Lamberto Tronchin (University of Bologna), Cobi van Tonder (University of Bologna) and Ruoran Yan (University of Bologna).

The "sound" of cultural heritage is often neglected and fragile, but it has the same historical value and research significance as tangible cultural heritage. The current study was driven by the investigation and preservation of the acoustic characteristics of the Teatro Argentina, one of Rome's historical theaters. In this paper, the impulse response (IR) of several of the main auditory parameters identified by ISO 3382-1 during the investigation was firstly recorded using standard methods. Secondly, an innovative acoustic research methodology was used to show the direction of arrival of the sound rays and to visualize the sound intensity. This methodology improves in understanding the early and late reflections of sound hitting elements of architecture.

[2124] *Soundscape as a place-making tool for industrial heritage sites*

Lei Sun (Università di Bologna).

This thesis explores the utility of soundscapes for industrial heritage landscapes as instruments for place-making and heritage conservation. The viewer's imagination can be stimulated by the design of soundscapes that recreate the aural atmosphere of bygone industrial production. This helps to preserve the authenticity and integrity of industrial heritage sites. A questionnaire was used in this research project to obtain responses from the general public regarding their preferences and needs for the soundscapes of industrial sites and examined their effects on tourists at industrial historical sites by analyzing them in the context of case studies. The findings suggest that the creation of soundscapes can improve the experience of visitors and promote the interpretation and protection of industrial heritage. This will enable members of the public to gain a deeper understanding and importance of the city's industrial past and will also contribute to the rehabilitation of sites through the incorporation of historically appropriate sounds into the space. This

research provides new insights into the field of industrial heritage conservation by highlighting the value of soundscapes.

[2211] *ISO X3D 4.0 Audio Graph for web3D applications. Latest trends in declarative 3d sound for the web.*

Maria Papadaki (Dept of Electrical and Computer Engineering), Eftychia Lakka (Faculty of Computing, Engineering and Science, University of South Wales), Georgios Daskalakis (Department of Electrical and Computer Engineering, Hellenic Mediterranean University), Richard Puk (Intelligraphics Incorporated), Don Brutzman (Naval Postgraduate School, Web3D Consortium) and Athanasios Malamos (Dept of Electrical and Computer Engineering, Hellenic Mediterranean University).

Our work, as 3D Audio Working Group of Web3D Consortium, aims to integrate acoustic properties associated with geometric shapes together with 3D spatial sound and insert this new technology into the X3D v4.0 ISO standard. In this paper we present X3D 4.0 Audio Graph API as it is embedded with all the latest improvements and examples. Moreover, we introduce novel ideas and use cases on how X3D v4.0 may improve immersion experience in theatre or music performances and concerts.

https://www.web3d.org

[2217] *Immersive spatialized live music composition with performers: a case study, Le vent qui hurle*

Nicola Giannini (Université de Montréal - CIRMMT).

Le vent qui hurle is a live music piece for ten performers of semi-modular synthesizers, metal sheets and a dome of speakers. The work was written for the Ensemble d'oscillateurs of the University of Montreal directed by Nicolas Bernier. The aim was to evoke the sound of the wind that whistles loudly, an immersive sonic phenomenon with unpredictable spatial behaviour. During the piece, some performers play the synthesizers while others move around the audience playing the metal sheets. The aim is to explore the relationship between the speakers' spatialization of synthesized sounds with acoustic sources' spatialization. The goal is also to foster an immersive experience for the audience, eliminating the barrier between spectators and performers and trying to stimulate a reflection on the concert ritual. A score was created which provides the notation for the performers' movements, the metal sheets and the synthesizers parts. Compositional techniques were developed to exploit the characteristics of analogue sound in a spatialization context. Special attention was paid to the relationship between the synthesizers' performers' position and the position of the sounds they create. To control the spatialization, a tool based on the analysis of audio descriptors named MapSPAT was created. With this tool is possible to link in real time the spatialization of a sound to its characteristics, e.g. the spectral centroid, in those contexts in which it is not possible, or not desired, to plan the spatialization before the concert. Thanks to the tool's matrix interface, users can link sound parameters with spatial parameters. A variation in sound thus causes a variation in space.

[2230] *Enhancing Virtual Audio Immersion Using Binaural Mesh*

Ivica Bukvic (Virginia Tech).

Binauralization of sound sources has become a dominant form of virtual spatialization in arts, entertainment, and industry. To date, rendering of each source has typically relied on a separate instance of a binaural algorithm, thus allowing for per-source localization. In terms of the required computational power, such an approach scales linearly. When coupled with computationally heavy Head Related Transfer Functions, such an implementation often relies on various optimizations that can compromise the fidelity of a spatial image. In the following paper the author presents a novel approach to binauralizing virtual sound sources using D4 library's Binaural Mesh. The paper presents the rationale for the Binaural Mesh, outlines its newfound affordances, including Spatial Mask, scene management, as well as ultradirectionality and a capacity for improved spatial image, and presents a feasibility study focusing on its computational overhead in source rich environments that exceed the capacity of the existing HRTF-based counterparts.

[2231] *Perceptual evaluation of Adaptative Higher Order Ambisonics diffusion*

Adrien Vidal (Aix Marseille Univ, CNRS, PRISM), Mitsuko Aramaki (Aix Marseille Univ, CNRS, PRISM), Sølvi Ystad (Aix Marseille Univ, CNRS, PRISM) and Richard Kronland-Martinet (Aix Marseille Univ, CNRS, PRISM).

Higher Order Ambisonics (HOA) is a technology aiming to capture and reproduce 3D soundfields. HOA has many advantages in comparison to other technologies, but its main drawback is that the optimal reconstruction area, called sweet-spot is relatively small at low orders. To overcome this constraint, we propose to adapt the HOA rendering in real-time according to the listener's position by computing the HOA decoding matrix for each listener's location. Nevertheless, the HOA rendering is very sensitive to the loudspeakers' disposition (and especially to their regularity) and to the decoding algorithm. For this reason, the adapted HOA rendering needs to be assessed.

In this paper, we investigate the perceptual rendering of such an adaptation for a five order HOA system. A perceptual test was conducted considering the following factors: adaptation (with and without), listener's position (10 cm, 30 cm and 60 cm offsets from the center of the system), geometry of the loudspeaker array (spherical and cubic), decoding algorithm (Energy preserving and All-Round decoding) and optimization (none and in-phase). Results showed that using the adaptation whatever the condition, the perception of spatial attributes of the sound source is preserved until a 60 cm translation. Moreover, without using the adaptation for the slightest translation tested here (10 cm), the perception of spatialization was less altered using the cubical geometry than the spherical geometry. Listeners did not perceive major differences between the two decoding algorithms tested here.

[2280] *Reconstructing the Dynamic Directivity of Unconstrained Speech*

Camille Noufi (Stanford University), Dejan Markovic (Meta Reality Labs Research) and Peter Dodds (Sonos, Inc. (Formerly at Meta Reality Labs Research)).

This article presents a method for estimating and reconstructing the spatial energy distribution pattern of natural speech, which is crucial for achieving realistic vocal presence in virtual communication settings. The method comprises two stages. First, recordings of speech captured by a real, static microphone array are used to create an egocentric virtual array that tracks the movement of the speaker over time. This virtual array is used to measure and encode the high-resolution directivity pattern of the speech signal as it evolves dynamically with natural speech and movement. In the second stage, the encoded directivity representation is utilized to train a machine learning model that can estimate the full, dynamic directivity pattern given a limited set of speech signals, such as those recorded using the microphones on a head-mounted display. Our results show that neural networks can accurately estimate the full directivity pattern of natural, unconstrained speech from limited information. The proposed method for estimating and reconstructing the spatial energy distribution pattern of natural speech, along with the evaluation of various machine learning models and training paradigms, provides an important contribution to the development of realistic vocal presence in virtual communication settings.

[2585] *Virtual Media Prototyping Engine - Using VR to prototype museum audio designs during a pandemic.*

Robert Brinkworth (Resonate-Audio & Nest.media).

During the pandemic, we developed a digital twin of the entire UAE pavilion at the World Expo in order to collaborate, create and test audio designs and mixes in VR. We developed a world-first audio media pipeline via a custom coded driver, allowing us to build an accurate audio model for each area; complete with individual point source audio (up to 48 virtual emitter points), all calibrated using inverse square laws, realistic room reverbs and a custom collaborative CDN to allow all art directors, contributors, designers and clients to collaborate from home. We further designed the system to include visual media, all sound sychronised to allow a full 360 degree creative ideation solution to move our designs forward. This process was a first for new technology prototyping of AV design media, and had the knock-on effect of vastly sped up physical installation times, better mixes through our VR snapshots & multiple design iterations. We are currently deploying this same system on new museum design projects and working with an architectural team to further broaden the scope for this tool.

[2694] *Explore the acoustics of Teatro dell'Opera of Rome*

Lamberto Tronchin (University of Bologna), Antonella Bevilacqua (University of Parma), Ruoran Yan (University of Bologna) and Cobi van Tonder (University of Bologna).

The Opera House, as one of the typical representative buildings of Europe, has an outstanding cultural heritage value. In addition to the conservation, restoration and preservation programs, including the establishment of databases for the main body of the building, the acoustics, as a more fragile intangible cultural heritage, should also attract the attention of scientists and scholars. The paper follows the requirements of the standard set out in ISO 3382-1 and consists first of all of a investigation and collection of the traditional acoustic parameters of the Teatro dell'Opera of Rome. A more innovative approach is then applied, that is, the use of multi-channel spherical array microphones and the creation of superimposed videos showing real-time impulse response (IR), to visualize the interactions that exist between the sound waves and the architectural elements at the room boundaries. The results of the study show that Teatro Rome still has excellent acoustics over time, even when compared to contemporary opera houses.

[2715] *Acoustic characterization of the new theatre in Amarante*

Octávio Inácio (InAcoustics, Lda.), Filipe Martins (InAcoustics, Lda.), André McDade (InAcoustics, Lda.) and Daniel José (InAcoustics, Lda.).

Inaugurated in 1947 the Amarante Cine-Theatre, in the northern region of Portugal, was used for cinema and theatre presentations up until 1980. From then on it was used for other, non-artistic activities, until a decision was taken, by the Amarante municipality, to bring it back to life with an Architectural competition for its refurbishment in 2011. After several years of difficulties to complete the project and finish the construction, the new theatre was finally inaugurated in 2023. With a total capacity of 386 seats, this small theatre was designed to host different types of events, from orchestra to theatre performances, and built with tight budget constraints. This paper presents an overview of the design process and the results of the acoustical measurements performed with the hall completed. The acoustic assessment was made using the ISO 3382-1 methodology, including measurements with a binaural head and a 64 microphone spherical array to help identify specific reflections. The new ODEON omnidirectional sound source was used, and the results are compared with other measurements using a dodecahedral source. The final results are also compared with the other halls of the same size and purpose.

[2831] *Towards a Data Driven Rendering Algorithm for Visually Impaired Film and Television Audiences*

Michael McLoughlin (University of York), Mariana Lopez (University of York), Kristián Hofstâdter (University of York) and Gavin Kearney (University of York).

In stereo sound systems, the listener’s ability to localise sound in the reproduced stereo field is affected by a variety of factors such as their listening position and the degree of separation between the loudspeakers. Understanding the effect of these factors is important for creating object-based mixes, where the panning position and volume of sounds can be adjusted to suit the user at the point of media consumption. Being able to control the panning and volume of different sources in the renderer is especially useful for visually impaired audience members, as it can allow for greater immersion and narrative understanding for film and television. For this reason, it is important to conduct listening tests on how these audiences perceive sound over stereo systems. These listening tests can then be used to inform renderer panning algorithms. However, before inviting visually impaired people to take part in listening tests, a robust methodology and control group comparison must first be established. Here, we present an algorithm that is driven by the results of a Minimum Audible Angle (MAA) staircase listening test for different listening positions and loudspeaker base width combinations. This consisted of 22 trials across 21 sighted participants. We calculated the mean MAA for these 22 trials and used these results to determine our panning angle when given a listener position and stereo base width. We then carried out a smaller listening test to determine the effectiveness of a perceptually informed rendering algorithm (8 tests across 8 participants). We achieved this by calculating the F-Score of our method against a control method that does not account for these factors. The mean F-score was 0.62 for our method against a mean of 0.36 for our control. Our results indicate that our algorithm has the potential to be used for object-based mixes where spatial

separation for all audience members is more important than precise object placement. Future research will carry out the same experiment with visually impaired audience members, and compare the results with this pilot study.

[2980] *On the acoustic track of the Teatro del Maggio Musicale of Florence*

Antonella Bevilacqua (University of Parma) and Lamberto Tronchin (Univerity of Bologna).

The new Italian theatres have been deeply studied for different scopes of research studies, with particular attention by the scholars related to the latest generation of technology that gives the possibility to investigate such cultural spaces more deeply. The results obtained by the acoustic survey inside the Teatro del Maggio Fiorentino, located in Florence, have been analyzed in accordance with the standard requirements stated in ISO 3382. In addition to the standard analysis, an overlay video shows a real-time impulse response (IR) and the interactions between the soundwave and the surrounding architecture. This supplementary investigation was possible to be realized by using a multichannel spherical array microphone, that allows to have control of the sound propagation through the space. A brief history of the Teatro del Maggio Fiorentino is also included.

[3016] *Extraction of Ambience Sound from Microphone Array Recordings for Spatialisation*

Vlad-Stefan Paul (University of Southampton), Nara Hahn (University of Southampton) and Jacob Hollebon (University of Southampton).

It is often desirable to separate main (primary) sound sources and background sound in microphone array recordings where the microphone signals are a combination of these two components, and then render primary and background sounds using different rendering techniques. Considerable research has investigated the extraction of the primary sources from the mixture using source extraction and localisation techniques, to then re-spatialise the extracted signals using the desired spatial audio reproduction format (for example, loudspeaker panning, Ambisonics or binauralisation). Little research, however, has focused on the separation and spatialisation of the ambience (background) component of the mixture, like secondary sources, diffuse field, or reverberation. This component is crucial for an accurate and realistic reconstruction of the overall 3D audio scene. This paper presents a new approach to extracting a spatialised version of the background field from a set of microphone array signals containing the original sound field mixture. The approach utilises an LMS algorithm to perform the separation between the primary sources and background sound. The fundamental theory of this technique is presented and validated with experimental measurements.

[3267] *Sweet melody: the acoustic analysis of the Auditorium Paganini realised inside a dismissed sugar factory*

Antonella Bevilacqua (University of Parma), Giacomo Tentoni (Gruppo CSA SpA Rimini) and Alessandro Martinetti (Gruppo CSA SpA Rimini).

The sugar factory Eridania headquartered in Parma was one of the industrial centers offering jobs to many people of the region Emilia Romagna. From the old structure characterized by a shoebox geometric volume, the architect Renzo Piano realized the Auditorium Paganini, dedicated to the most famous violinist who was called by Duchess Maria Luisa to conduct the orchestra of Parma. The new design consists of the installation of acoustic reflecting panels above the stage and the first rows of seats, which improve a good quality of acoustics inside a construction dominated by parallel walls and flutter echoes. Acoustic measurements have been carried out inside the auditorium according to ISO 3382. The measured results have been compared to the optimal values of modern concert halls having similar volume size. A description of the history and the evolution of the external envelope has been briefly added.

[3326] *Enhancing Audio Object Control within an Immersive Sound System*

Richard Foss (Rhodes University) and Lukas Klingebiel (Coda Audio).

Real time control over sound source positioning and movement is a primary advantage of object based sound systems over channel based systems. This paper describes a controller, processing engine, and associated library that enhances this object control. The library API enables the rendering of mix levels for a source to a zone of speakers, where the zone, source position and rendering algorithm can be specified. Speakers can be omitted for any source. The mix levels are implemented via a matrix mixer within the engine, which incorporates reverb buses. The controller allows, for each object, the selection of its rendering algorithm, distance based level and high frequency attenuation, reverb parameters and speaker isolation or locking. Object position and parameters can be saved and restored via snapshots, although a user can indicate for each snapshot which parameters of an object not be affected by this snapshot. In a snapshot each object can be set to move in one of three ways: moving to a static position, moving on a random path with adjustable range for each of the randomized parameters or on a user-defined path. There is OSC control over sound source positioning. Once again, for each object the nature of the position controller can be specified – mixing console, tracking systems, external OSC controllers, internal control groups, or DAW plugin. The OSC ID of an object can be specified so that third party controllers can be allowed this same level of object individuality.

[3381] *Spatial audio in YouTube VR videos and its impacts on audience engagement*

Huyen Nguyen (Kansas State University) and Madeline Willson (Kansas State University).

In media discourse, spatial audio or immersive audio was mentioned as an emerging technology that directly influences users' visual attention and enhances their immersive experience by simulating sound in three-dimensional space (Holm, Väänänen, & Battah, 2019; Hirway, Qiao, & Murray, 2022; Wincott, Martin, & Richards, 2021; Bosman, Buruk, Jørgense, & Hamari, 2023). However, prior studies also indicated the lack of audience research on the true impacts of spatial audio in virtual reality (VR) videos although more video producers started adopting the technology (Wincott et al., 2021; Wincott, 2022). This is a gap that needs to be addressed in order to fully understand the potential of spatial audio as a tool for creating engaging VR experiences. Based on the audience engagement model (Steensen, Conill, & Peters, 2020), we collected viewer statistics displayed along 1,116 360-degree videos on the official YouTube VR channel between between 2015 and 2023 and conducted a content analysis of the entire population in order to detect the impacts of spatial audio on the number of views, likes, and counts. Multiple regression analyses did not detect statistically significant impacts of spatial audio on viewer statistics. However, among videos with spatial audio, fictional videos appeared to be more favorable than non-fictional videos, after controlling for days on YouTube. Our preliminary results are aligned with prior studies which pointed to the lack of equipment to experience spatial audio, the lack of ability to detect spatial audio, and discomfort feelings caused by mixed points of audio (POA) (Wincott et al., 2021; Wincott, 2022). Future studies may want to look more in-depth into the POAs for a better explanation of audience behaviors.

[3588] *Evaluation of virtual acoustic environments with different acoustic levels of detail*

Stefan Fichna (Carl von Ossietzky University Oldenburg), Steven van de Par (Carl von Ossietzky University Oldenburg) and Stephan D. Ewert (Carl von Ossietzky University Oldenburg).

In virtual acoustics, the creation and simulation of realistic acoustic environments enables the use of ecologically valid daily-life situations in hearing research and audiology. However, it remains unclear what acoustic level of detail (ALOD) is necessary to capture all perceptually relevant effects in the room acoustics simulation, particularly with respect to simplifications required for real-time applications. This study investigates the effects of varying ALOD on the simulation of three different real environments, a living room with a coupled kitchen, a pub, and an underground station. ALOD was varied by generating different numbers of image sources for early reflections, by excluding geometrical room details specific for each environment, or by reducing the simulation to only the direct sound. The simulations were perceptually evaluated i) using headphones in comparison to binaural room impulse responses measured with a dummy head in the corresponding real environments, as well as ii) in a 3-dimensional loudspeaker array. The study assessed spatial audio quality for a pink pulse, a music sample, and a speech token, with a focus on auditory distance perception and externalization. Furthermore, plausibility and speech intelligibility were evaluated. The results provide insights into the perceptually required ALOD depending on the environment and the stimulus.

[3632] *Acoustic features of Teatro Nuovo of Spoleto*

Ruoran Yan (University of Bologna), Giacomo Tentoni (Gruppo CSA SpA Rimini), Van Tonder Cobi (Department of Architecture, Bologna) and Alessandro Martinetti (Gruppo CSA SpA Rimini).

The Teatro Nuovo of Spoleto was opened in 1864. Composed of 5 orders of balconies and with a capacity of 800 seats, Teatro Nuovo is one of the most famous Opera theatres in northern Italy. Different refurbishment works affected the acoustics of the main hall, especially the interventions on the ceiling by the accentuation of the curvature that changed the uniform distribution of sound as it was at the beginning. Different campaigns of acoustic measurements have been undertaken by placing the sound source on the stage and in the orchestra pit. The analysis of the measured data recommends that the inclination of the ceiling shall be lowered to the original height such in order to avoid shadow zones to the audience.

[3671] *Listen to the theatre! Exploring Florentine performative spaces*

Andrea Gozzi (Dipartimento SAGAS, Università degli Studi di Firenze) and Gianluca Grazioli (Schulich School of Music – McGill University).

A music performance space constitutes the frame as well as the content of the listeners' experience. The acoustic environment forces continuous negotiations that differ according to a listener's role and position as conductor, performer or audience member. The aim of this research is to investigate the acoustics of a performative space, the Teatro del Maggio Musicale Fiorentino in Florence, following two complementary paths, both based on an interactive model. The first one offers an impulse-response experience: the user can virtually explore the opera hall by choosing between the binaural reproductions of 13 different listening positions whether off-site (using an audiovisual web app) or on-site, through bone conductive headphones. The second one is about the aural and visual perception of a rehearsal of the romance "Una furtiva lagrima" from Donizetti's opera L'elisir d'amore. Through the use of ambisonics recordings, 360 degrees videos and a virtual reality headset, the user can experience the performance from three different points of the theatre alternatively: on stage, in the orchestra pit and in the audience area. The three different perspectives can be switched instantaneously using a remote controller.

[3772] *Teatro della Fortuna of Fano: acoustic evaluation*

Antonella Bevilacqua (University of Parma), Giacomo Tentoni (Gruppo CSA SpA Rimini) and Alessandro Martinetti (Gruppo CSA SpA Rimini).

Teatro della Fortuna is a small Opera theatre in the city of Fano. It was realized during the 18th century and refurbished by Ferdinando Bibiena in 1718. It was built as a fixed theatre after temporary installations were created in a lightweight wooden structure. The horseshoe shape plan of the main hall is considered suitable for both prose and Opera. Acoustic measurements have been carried out in accordance with ISO 3382. The measured results have been assessed with respect to the optimal values of concert halls having similar volume sizes. A brief description of the history has been added to better understand the construction evolution of the theatre.

[3844] *Immersive networked music performance systems: identifying latency factors*

Luca Turchet (University of Trento, Department of Information Engineering and Computer Science) and Matteo Tomasetti (University of Trento, Department of Information Engineering and Computer Science).

Music playing with binaural systems has thus far received remarkably little attention by the research community. Such an activity is particularly relevant for the case of networked music performances (NMPs) where geographically displaced musicians play together over a telecommunication network. Recent findings in simulated settings have shown the preference of musicians for spatialized listening during collaborative playing using headphones, as opposed to listening with conventional stereophonic systems. This result has highlighted the need to enhance current NMP systems with the spatial rendering of the acoustic scene, leading to the development of immersive networked music performance (INMP) systems. A crucial aspect that needs to be addressed is the contribution of the spatial audio system to the overall

latency of the audio processing and transmission chain between the network nodes. In this paper we provide an overview of the INMP topic and identify the latency contributions of the components of an INMP system. We then relate such components to state-of-the-art hardware and software systems. Based on our analysis we draw concluding remarks and discuss the open challenges for progressing the INMP field.

[3878] *A New HRTF Interpolation Approach for Nonlinear 3D Audio Systems*

Valeria Bruschi (Università Politecnica delle Marche), Nefeli Dourou (Università Politecnica delle Marche), Alberto Carini (University of Trieste) and Stefania Cecchi (Università Politecnica delle Marche).

Binaural synthesis plays a vital role in 3D audio systems, relying on head-related transfer functions (HRTFs) to achieve realistic sound rendering. However, the reproduction system used in the HRTF measurement can introduce nonlinearities that impact the immersive experience. In light of this, the present study introduces a robust HRTF identification method designed to address these nonlinearities. Additionally, a novel HRTF interpolation technique is proposed, aiming at reducing the required number of measurement points while still ensuring faithful reproduction of the virtual auditory scene. Specifically, the interpolation is applied following a frequency warping process, which enhances the representation of lower frequencies within the HRTFs while simultaneously reducing the number of frequency bins. The resulting interpolated HRTFs are then unwrapped and utilized for binaural reproduction, contributing to an improved audio experience.

[4265] *Effect of the finishing on the sound absorption characteristic of a mineral wool*

Edoardo Alessio Piana (University of Brescia) and Andrea Breviario (Independent consultant).

Sometimes, mineral wool cannot be used “as it is” for civil applications but needs some type of finishing. In this manuscript, a test case is analysed where a mineral wool ceiling was finished using a paint suggested by the company producing the material. The reverberation time of the room before painting the ceiling was suitable for the destination of use of the room. On the other hand, when the wool was painted, the reverberation time and in general the acoustics of the room resulted to be unsatisfactory. This research aims to characterise the material before and after being painted. The measurements on the material were performed using a four-microphone impedance tube.

[4275] *Teatro India “Sala B”: acoustic analysis and descriptions*

Cobi van Tonder (University of Bologna), Ruoran Yan (University of Bologna) and Lamberto Tronchin (University of Bologna).

The construction of avant-garde theaters, which provide the essential acoustic conditions for emerging performances, stems from the evolving times and the emergence of new artistic expressions. In this research, we undertake a comprehensive examination of the conventional acoustic parameters of Sala B in Teatro India, Rome. The measurements conducted strictly adhere to the specifications outlined in ISO 3382-1. To visualize the direction and intensity of sound reflections, we employ a combination of multi-channel microphones and a panoramic camera. By documenting the acoustic characteristics of Sala B and evaluating its acoustic quality, we offer valuable insights and guidance for the design and construction of diverse performance scenarios.

[4460] *The sound field and the sound object*

Giuseppe Pisano (Norges Musikkhøgskole).

A reflection over the technical choices in the project: “Spontaneous music diffusions” and the potentiality of the use of Ambisonics recordings in the context of sonic anthropology.

[4464] *Setup for choir recordings in virtual churches*

Pascal Palenda (Institute for Hearing Technology and Acoustics, RWTH Aachen University), Lukas Aspöck (Institute for Hearing Technology and Acoustics, RWTH Aachen University), Stefan Morent (Department of Musicology, University of Tübingen) and Michael Vorländer (Institute for Hearing Technology and Acoustics, RWTH Aachen University).

Singing is an important part of most religious rites. In the research project “Sacred Sound - Sacred Space” the influence of the room acoustics on the performance and the perception of singing is investigated. In this project, ensemble singers are performing in virtual recreations of historic spaces. So far, recordings in two spaces were completed: The still existing Cistercian church of Maulbronn monastery and the Maior Ecclesia in Cluny, of which only remnants exist. The current set-up allows for a static scenario with pre-simulated, binaural room impulse responses corresponding to a real singing situation in a room. For the recordings, the singers were situated in an hemi anechoic chamber and recorded using a headset microphone, while perceiving the reflections of the virtual churches via open headphones. In this talk, the technical setup of the recordings, the applied room acoustic simulations and the limitations of this approach will be presented, complemented with a discussion regarding potential improvements of the setup and the first steps towards an analysis of the recordings.

[4518] *Measuring the influence of audio on immersive experience in extended reality and digital games: a systematic review*

Jacob Hedges (University of Technology Sydney), Robert Sazdov (University of Technology Sydney) and Andrew Johnston (University of Technology Sydney).

A key goal of extended reality (XR) technologies and digital games is to create immersive experiences for the user. Immersive experience is a multifaceted phenomenon that encompasses perceptual and psychological factors, and audio has been recognized as a key modality that can influence such experiences. Despite extensive research on the subject, there is no consensus regarding the ways in which audio impacts immersive experience. This is also the case in terms of the most appropriate methodology for evaluating this phenomenon in the context of end-user experience in XR and digital games. This paper aims to establish a comprehensive overview of the experimental methodologies associated with this topic through a systematic literature review of the last 20 years of research. The inclusion criteria focuses on research papers that attempt to measure audio-related independent variables (e.g. spatial resolution) on immersion-related dependent variables (e.g. sense of presence), in the context of XR environments and digital games. The results reveal a preference for VR HMD and headphone-based experiments, identifying a lack of AR/MR exploration, as well as the necessity for standardized, audio- specific questionnaires in data collection for a more comprehensive understanding of the field.

[4764] *The effect of immersive audio rendering on listeners' emotional state*

Nefeli Dourou (Università Politecnica delle Marche), Valeria Bruschi (Università Politecnica delle Marche), Andrea Generosi (Università Politecnica delle Marche), Maura Mengoni (Università Politecnica delle Marche) and Stefania Cecchi (Università Politecnica delle Marche).

Immersive audio rendering techniques allow for generating a 3D scenario where the listener can perceive the sound from all directions. An important aspect of these approaches is the subjective perception of the listener and how these types of systems are perceived from the emotional point of view and how they can influence the listener’s mood. In this context, a deep investigation of immersive sound perception considering subjective perception in terms of flowing emotion is performed. Starting from a 4-channels immersive audio system and an emotion-aware system based on the analysis of the user’s facial expressions, several experiments have been performed to investigate a correlation between immersive perception and the listener’s emotions.

[4935] *Performance Optimization of Personal Sound Zones with Crosstalk Cancellation*

Yue Qiao (Princeton University) and Edgar Choueiri (Princeton University).

Two optimization approaches are proposed to enhance the performance of personal sound zone (PSZ) systems with crosstalk cancellation (XTC). The two approaches adjust the trade-off between two important performance attributes of a system: acoustic isolation and crosstalk cancellation, by either modifying the cost function in the optimization problem (the direct approach) or controlling the amount of crosstalk in the target transfer functions (the indirect approach) in the filter generation process. The effectiveness of the two approaches is evaluated using metrics of inter-program isolation (IPI) and XTC level, through numerical simulations based on both a free-field system model and actual transfer function measurements. The results show that at low frequencies, the two approaches can effectively control the trade-off between the two attributes, but their effectiveness is reduced at high frequencies due to the small wavelength. Furthermore, the results demonstrate that the direct approach is more effective in manipulating the trade-off, whereas the indirect approach provides more precise control over the desired XTC performance.

[4952] *A Comparative Analysis of Speaker and Headphone-Based Immersive Audio in VR and Gaming Applications*

Kyle Marais (Rhodes University) and Richard Foss (Rhodes University).

Precise and accurate audio spatialization is crucial for gaming and Virtual Reality (VR) applications, as it is an essential aspect of immersion. When the audio of a virtual environment is synchronized with its visual sources and localized accurately, users are more likely to feel deeply engaged and suspend their disbelief. Audio Middleware software is commonly used in cooperation with game engines to meet the audio requirements of virtual environments created by game developers. This research paper presents a comparative analysis of speaker-and headphone-based immersive audio systems within the context of VR. The objective is to identify and analyse the differences between these two systems. Consequently, the paper details the implementation of an enhanced speaker-based Audio Middleware sound capability for Unity game engine developers. This capability utilizes the functionality of a well-known Audio Middleware software suite, FMOD, and a speaker-based spatialization system, ImmerGo. The headphone spatialization system used for comparison with the bespoke speaker-based system is the Oculus Audio Spatializer. To evaluate the systems, a VR application was developed that incorporated both spatialization techniques and served as the testing framework for the experimentation. This application enables users to control 3D sound in real time. The Leap Motion controller and Oculus Rift Head-Mounted Displays (HMD) enable users to select an audio source with a gesture and subsequently manipulate its position in three-dimensional space. The research findings indicate that, despite the headphone-based system offering more accurate audio localization and a higher 'perceived audio quality', the speaker-based system resulted in a greater sense of 'immersion'.

[4988] *Preliminary evaluation of a sound system employing cancelling auralizer for producing virtual rooms in a modern recording studio*

Gianluca Grazioli (McGill University - CIRMMT), Gianluca Grazioli (McGill University), Vlad Baran (McGill University) and Wieslaw Woszczyk (McGill University).

We investigate the operation of a virtual acoustic system on-site in the Immersive Media Laboratory (IMLAB) at McGill University and the benefits offered by a real-time canceling auralizer currently under development at CCRMA of Stanford University and the University of Limerick. The feedback canceling software system reduces the likelihood of feedback and howling in live real-time auralization of acoustic spaces through in-room microphone capture and loudspeaker projection of sound. Amplified microphone signals are sent to a feedback cancellation network which then convolves the signals with a variety of captured and synthesized impulse responses, projecting the output signal via loudspeakers. We present a preliminary analysis of the main acoustical parameters created under different acoustical conditions by the virtual acoustic system implemented in the controlled IMLAB environment. Ongoing research examines the interactions of professional musicians with the current state of the system, which will be continuously investigated in the near future. The work is carried out in collaboration with the authors of the feedback-canceling technology.

[5088] *Immersive Venice - A Thousand Echoes*

Giulia Vismara (Royal Academy and Conservatory Anvers), Cobi van Tonder (University of Bologna) and Angela MacArthur (Dept. Archeology University College of London).

The soundscapes of Venice are as fascinating as its architectural wonders, rich history, impressive art and vibrant contemporary life. The timeless essence of the city,carved in marble over the millennia, appears firm and fluid, echoing the ebb of the tides. Venice is a city engaged in an incessant dialogue with its past, bridging the gap between the ephemeral and the lasting. Following the technique of auralization, we embark on a journey to explore the echoes of Venice, capturing its acoustic footprints and the existing and imaginary immersive sound. We venture into the quiet nights of the city, recording impulse responses from unusual viewpoints, capturing different moments in time and space. Our collection includes field recordings that include the rhythmic cadence of thousands of steps, the daily buzz of voices on the streets and the gentle caress of water against boats, docks and seabeds, all contributing to the unique acoustics of the city. Through our artistic practice we challenge the notion of objectivity in scientific research, using technological means that respond to a specific definition of space, we intersect our subjective sound practices with a scientific approach. Living a notion of space as physical and sensory and a notion of time layered and non-linear, we agree all these elements to present an exploration enriched the auditory landscape of Venice. All this material that reveals the deep connection between the sound and the essence of this extraordinary city, will then serve to nourish 3 compositions and an acoustic mapping. This article contains a presentation of meta-perspectives that considers paradigms objectivity/subjectivity and the importance of recognizing multiple perspectives when working with acoustic and sonic footprints of places.

[5096] ***An examination of the acoustic characterization of Teatro India "Sala Oceano" in Rome.***

Cobi van Tonder (University of Bologna), Ruoran Yan (University of Bologna) and Lamberto Tronchin (University of Bologna).

Can old theater spaces provide the acoustic environment needed for emerging artistic performances? What are the acoustic characteristics of a theater with modern structural features, as distinct from the traditional Italianate theater? These two questions prompted the authors to explore the acoustic characteristics of Teatro India. The conventional acoustic parameters of the Sala Oceano of the Indian theater in Rome were first collected and assessed following the ISO 3382-1 criteria. The arrival direction and strength of sound reflections were detected as well employing a combination of multi-channel microphones and a 360°camera. The acoustic quality of the Sala Oceano was evaluated and analyzed in comparison with measurements of the other two main performance spaces of the India theatre, Sala A and Sala B.

[5103] ***Acoustics of the Manoel Theatre of Malta***

Lamberto Tronchin (University of Bologna), Cobi van Tonder (University of Bologna) and Antonella Bevilacqua (University of Parma).

In 1731 António Manoel de Vilhena commissioned the construction of a municipal theatre dedicated to the citizens of Malta. The project has been modeled upon the Teatro Massimo of Palermo, although the original shape of the Maonel theatre was circular and then transformed to a horseshoe shape in plan. Many refurbishment works that occurred during the 19th century modified the original design due to compliance with the current regulations in terms of fire safety and thermal comfort. With a total capacity of 623 seats and with an oval geometry, the Manoel theatre is entirely built with a wooden frame with a magnificent decoration of the ceiling giving the illusion of a dome. The architectural style of the interior design is typical Rococò. Acoustic measurements have been carried out inside the theatre according to ISO 3382 in order to study the acoustic response of the main hall. The measured results have been compared to the optimal values of other Opera theatres having similar volume size.

[5207] ***A Machine Learning Approach to Predicting Personalized Head Related Transfer Functions and Headphone Equalization from Video Capture Data***

Nikhil Javeri (Embody), Prabal Bijoy Dutta (Embody), Kaushik Sunder (Embody) and Kapil Jain (Embody).

In recent years, personalized Head-Related Transfer Functions (p-HRTFs) have become crucial for achieving high-quality spatial audio in extended reality (XR) applications. However, obtaining personalized HRTFs is a complex and time-consuming process. In this paper, we propose a novel technique to predict personalized HRTFs based on 2D images or video captures. We present the different components in this process including the 3D reconstruction of an ear based on 2D images or video followed by the HRTF estimation using Boundary Element Methods or HRTF prediction using Neural Networks. Additionally, we present a new method for estimating Personalized Headphone Equalization (p-HPEQ) curves based on optical data of the ear. This approach can improve the accessibility and convenience of personalized spatial audio and enhance the personalization of headphone audio for tailored listening experiences. The accuracy of both p-HRTFs and p-HPEQs are validated with objective and subjective experiments.

[5308] ***Psychoacoustics of rock art sites: the case study of the shelters Diosa I and Horadada (Cádiz, Spain)***

Samantha López-Mochales (Brainlab - Cognitive Neuroscience Research Group, Institute of Neurosciences, UB, Barcelona), Lidia Alvarez-Morales (Institut d'arqueologia UB, Departament d'història i arqueologia UB), Neemias Santos da Rosa (Institut d'arqueologia UB, Departament d'història i arqueologia UB), María Lazarich (Departamento de Historia Geografía y Filosofía, Universidad de Cádiz), Margarita Díaz-Andreu (Institut d'arqueologia UB, Departament d'història i arqueologia UB, ICREA) and Carles Escera (Brainlab, Institut de Neurociències UB, ICREA, IRSJD).

The Artsoundscapes project seeks to understand the role of acoustics in the selection by past communities of certain environments to set activities involving rock art production. Within this framework, this article addresses the subjective perception of the acoustics of two rock art sites located in Cádiz (Spain): Diosa I and Horadada. The psychoacoustics of these two rock art sites is investigated by means of two separate listening tests. In the first test, a group of participants developed a corpus of words to subjectively describe the acoustic features of the selected sites. In the second test, a different group of participants rates the descriptors assigned by the first group. The auralizations, rendered via a third-order ambisonics speaker array, consisted of ten sounds of different characteristics (including singing, speech and music) convolved with a set of impulse responses gathered at the selected sites.

[5351] ***Convolution, virtual acoustics, and interactions with impossible worlds***

Eoin Callery (Irish World Academy of Music and Dance), Jonathan Abel (CCRMA - Stanford university), Elliot Canfield-Dafilou (Institut Jean le Rond d'Alembert, Sorbonne Université/CNRS) and Jelena Perišić (Irish World Academy of Music and Dance).

This paper presents a discussion on a series of artworks created by the authors that explore the potential of using sound produced in virtual spaces as artistic materials. Specifically, the works under discussion generate unique spatial and auralization effects that are achieved through the use of novel convolution techniques and immersive virtual acoustic technologies. We demonstrate our novel convolution techniques for deriving impulse responses from non-spatial sources, such as audio signals, and explain how these impulse responses have been employed in creating immersive installations for both web-based experiences and in-person interactive installations and performances. The impulse responses used in this work are obtained from sources including recordings of medieval hand-bells, the singing of Gesualdo madrigals, spoken texts, ambient street sounds, and more. The online works discussed in this paper are diffused through loudspeakers and headphones, while the interactive in-person installations and performances are diffused over headphones or through the loudspeakers of mobile convolution-based virtual acoustic systems. Several audio-visual examples to illustrate the techniques and outcomes of the work are included.

[5613] ***Ambisonic room impulse responses extrapolation guided by single microphone measurements***

Yake Fan (Center of Intelligent Acoustics and Immersive Communications, Northwestern Polytechnical University) and Wen Zhang (Center of Intelligent Acoustics and Immersive Communications, Northwestern Polytechnical University).

Sound fields and room impulse responses represented using higher-order-Ambisonics (HOA) is an efficient way to generate six-degree-freedom (6DoF) binarual rendering. However, measuring the room impulse responses of the entire space is impractical in reality. While many methods have been proposed for Ambisonic room impulses response (ARIR) interpolation and extrapolation, in this work, we propose a robust ARIR extrapolation method by combing the ARIR, which is measured at a fixed point inside the room, with the monaural RIR measured at the extrapolation point. The ARIR is decomposed into a direct part, distinct early reflections, and the residual reverberation. For the direct and early reflection parts, extrapolation is performed by estimating the source positions based on a parametric analysis of the measured ARIR and monaural RIR. The extrapolation of the residual reverberation is achieved by manipulating the monaural RIR, where the weights are derived from ARIR measurements. Finally, objective and subjective evaluations are conducted to evaluate the accuracy and robustness of the proposed method. The results show that the proposed method approximates the true ARIR in terms of objective indicators and subjective perception.

[5719] *Quest-Driven Spatial Songs – Applying Spatial Narratives in 6DoF Popular Songs*

Yunyu Ong (University of Technology Sydney), Robert Sazdov (University of Technology Sydney) and Andrew Johnston (University of Technology Sydney).

This research explores the potential of spatial narrative design frameworks to create Quest Driven Spatial Songs (QDSS). QDSS challenges the traditional linear format of music and offers a highly spatialized musical experience for music recordings. While increased accessibility to spatialized formats like Dolby Atmos in Apple Music has afforded greater spatiality in music recordings, it typically limits the audience to a fixed point in space (3DoF). By using space-time adaptive Game Engines' Spatial Algorithms and/or their corresponding middleware, this research proposes an opportunity to create song structures that behave like buildings, showing different musical "faces" depending on when and where the listener is in space-time (6DoF).

The creative component of this research is an iOS app called Empires. It serves as an illustration and exploration of QDSS and spatial narrative design frameworks. The app offers a 6DoF experience where the song shows different arrangements and lyrics based on the listener's position in space-time. Drawing inspiration from spatial narratives in game quests, the QDSS approach adapts these frameworks to music creation, where each instrument is spatialized in virtual space. Empires showcases the potential of QDSS so as to create new and innovative musical experiences that leverage spatial technologies to enhance audience engagement and create a deeper connection with music

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[6068] *Browser Based Webcam Head-Tracked Ambisonics (WHAM)*

Mark Dring (University of Derby) and Bruce Wiggins (University of Derby).

This paper describes the development and implementation of a real-time head-tracked auralisation platform using Higher Order Ambisonics (HOA). The headphone-based audio is decoded binaurally using open-source and freely available web technologies without needing specialist head-tracking hardware. The room scanning and head-tracked presentation technique discussed can provide accurate auralisation of real (captured/measured) and/or modelled spaces using up to 15th-order Ambisonics, much higher than current Ambisonic microphone technology allows. The differences between measuring the acoustic response of a space using Ambisonic microphones and rotated Head and Torso Simulators (HATS) are discussed along with the importance of removing the left/right symmetry assumption used in current Ambisonic to Binaural implementations. A Dynamic Binaural Reverberation Acquisition Technique (D-BRAT) is introduced allowing for a full 3D binaural capture and reproduction of an acoustic space while allowing dynamic head rotation by transformation into the Ambisonic, Spherical Harmonic domain. Open-source Software has been created (based on JSAmbisonics) to allow this Binaural Room Impulse Response (BRIR) Data to be auditioned using up to 15th-order Ambisonics with dynamic head rotations considered in real-time with accompanying 360-degree graphics display. An example implementation of this work can be found at: https://brucewiggins.co.uk/WHAM/.

[6258] *Study on Sound and Sound Design for Accessibility in Cultural Heritage Contexts in Ireland*

Caitlin Kelly (MTU Cork School of Music, SFI Advance CRT).

From our tangible heritage such as Newgrange, our native instruments, and the Book of Kells, to the intangible heritage of our myths, legends, and oral traditions, Ireland is a place with a colourful and expansive cultural heritage (CH). However, many of the country's tangible CH sites remain if not partially, then completely inaccessible to the disabled community. Ireland signed the UN Convention on the Rights of a Person with Disabilities in 2007, and therefore acknowledges that equal access to Cultural and Leisure activities for disabled people is a right. However, change can be slow, and the ratification of this document took eleven years- i.e. it was not signed until March 2018 [1]. Early findings from the research are discussed. Additionally, potential solutions for improving accessibility outcomes in CH sites for visually impaired/blind people are investigated. Augmented 3D sound, ambisonics, and sound design are suggested as a facilitator for inclusive heritage experiences that elicit the same sense of "awe and wonder" [2, p.243] as seeing these sites does. Though there is a lack of sound design utilised in this way in Irish CH sites, more specifically rural sites, there is precedent for using sound design as an accessibility tool in other fields (e.g. gaming). This paper outlines the current state of CH and sound design in Ireland, and reflects on how the research following could potentially improve this.

[6432] *The Localizability of the Closest Wall with a Speaking Avatar at Increasing Distances in Three Rooms*

Annika Neidhardt (Technische Universität Ilmenau).

The presented study examines the maximum distance at which listeners can still localize the direction of a nearby wall if the own mouth is the sound source. For this investigation, oral binaural room impulse responses (OBRIRs) were measured with a KEMAR with mouth simulator at nine different distances to a wall in an anechoic chamber and two rooms with deviating reverberation properties. In a headphone-based dynamic auralization, the participants of the psychoacoustic experiment had to turn until they thought to be facing the wall. In a stair-case inspired procedure, the test always started with the closest position of 25cm. In case of a successful localization in at least two in three trials, the distance could be increased in intervals of 25cm up to about 2m. The results exhibit considerable differences in individual performance, which is in line with the observations of earlier works. At a distance of 25cm, all participants could localize the direction of the reflecting wall. From a distance of 50cm onwards, more and more participants found it difficult to determine the correct direction. In the anechoic room, four of the 22 participants succeeded in the localization at the 2m distance. In the reverberant rooms, the localizability decreased significantly.

[6668] *A small Opera theatre in the province of Cremona: the acoustics of the Social theatre of Soresina*

Ruoran Yan (University of Bologna), Giacomo Tentoni (Gruppo CSA SpA Rimini) and Alessandro Martinetti (Gruppo CSA SpA Rimini).

The Social Theatre of Soresina has been opened in 1840, a period determined by the debates between Francesco Milizia's theories, supporting the Greek amphitheater with the scenic building as realized by Palladio, and the new shapes characterized by the horseshoe and oval shapes. The reduced volume size of the Social Theatre of Soresina, composed of three orders of balconies, summarises the construction techniques very popular during the 19th century. The measured results, obtained by an acoustic survey carried out inside the theatre, have been compared with the acoustic simulations of a digital model realized with AutoCAD software. The main acoustic parameters have been assessed according to ISO 3382-1.

[6922] *Spherical Wave Diffraction for Microphone Arrays Operating in Near Field*

Daniel Pinardi (University of Parma).

Microphone arrays for spatial audio recording and reproduction became very popular in the last decade, due to the spread of virtual and augmented reality for entertainment, remote work, and teleconferencing. Several systems have been distributed on the market, and most of them are made of a rigid sphere. In this way, it is possible to rely on the theoretical solution of the plane wave diffraction to get the beamforming filters. Such filters are employed for synthesizing virtual microphones of arbitrary directivity from the pressure signals captured by the capsules of the array. However, the plane wave assumption corresponds to a far field condition, requiring the sound sources to be located far away from the microphone array to produce wave fronts with negligible curvature. As the sources are closer to the array, the curvature of the wave fronts becomes more significant, and the hypothesis of plane waves cannot be accepted. In this paper, the problem of spherical wave diffraction over a rigid sphere is investigated through numerical simulations based on Finite Elements Method and experimental measurements. It will be shown that the spatial performance is significantly degraded when theoretical filters are employed in near field conditions, while numerically calculated filters can provide a reliable improvement for near field beamforming.

[7027] *Binaural synthesis adjusting a simple gain and delay based on the two-channel optimal source distribution*

Motoki Yairi (Kajima Technical Research Institute), Tsuguto Hoshino (Kajima Technical Research Institute) and Takashi Takeuchi (OPSODIS Limited, University of Southampton, ISVR).

Two-channel optimal source distribution (OPSODIS) is a binaural synthesis technique over loudspeakers that utilizes a pair of conceptual sound sources whose azimuthal location varies continuously as a function of frequency. In practice, discretized sound sources with appropriate frequency ranges make it possible to realize practical systems based on the principle. As a result, the inverse matrix derived from a plant matrix of the discretized system including head-related transfer functions (HRTFs) is well conditioned to satisfy equalization of the reproduction transfer function and crosstalk cancellation simultaneously. In this paper, the authors propose a simplified technique using the discretized OPSODIS arrangement that emphasizes the crosstalk cancellation, minimizing dynamic range loss and distortion of the reproduction transfer function. This can be accomplished by adjusting only a relative gain and delay representative of each frequency range in the discretized sound sources without any inverse FIR filters. Theoretical considerations in a symmetrical free field model show that the proposed system can create some listening sweet spots equivalent to those from the previous one when the discretization is fine enough. One of the advantages of this system is that it can emphasize only the crosstalk cancellation and not equalize the reproduction transfer function. From numerical simulations using a plant matrix including HRTFs, it is confirmed that the HRTF spectrum of the straight component is well reproduced at almost all frequencies with keeping a sufficient crosstalk cancellation and a minimum dynamic range loss. The proposed system is likely to enhance an immersive spatial impression without any sense of in-head localization even for sound sources not having spatial information from HRTFs, such as stereo sound sources.

[7134] *Discussion of Acoustic and Perceptual Optimization Methods for Measuring Spatial Room Impulse Responses with a Mobile Robotic Platform*

Georg Stolz (Technische Universität Ilmenau), Florian Klein (Technische Universität Ilmenau), Stephan Werner (Technische Universität Ilmenau), Lukas Treybig (Technische Universität Ilmenau), Andreas Bley (MetraLabs GmbH) and Christian Martin (MetraLabs GmbH).

In the field of Auditory Augmented Reality (AAR), one aim is to provide a listening experience that is as close as possible to a real scenario. Measured Spatial Room Impulse Responses (SRIRs) describe the acoustics of a room and can serve as a reference for acoustic simulations or parametrization of room acoustics. In previous works, a measurement system for SRIRs using a mobile robotic platform was introduced. The system consists of a commercially available self-driving platform on which a microphone array is mounted, while the sound sources are distributed at fixed positions in the room. The system is able to conduct high spatial resolution measurements of SRIRs in a uniform grid. In applications where time is limited and/or the area to discover is large, however, a high-resolution measurement is not always feasible.

Therefore, the goal of this contribution is to compare different approaches for optimizing the measurement grid. One approach is to use mathematical optimization on acoustic parameters derived from a small set of initial measurements

to determine new measurement positions in a iterative manner. Another approach is to optimize the measurement grid in respect to human auditory perception, incorporating e.g. just-noticeable differences of distance and localization perception.

The results show that both approaches can achieve significant reductions in the number of measurements required for a adequate acoustic spatial reproduction, with different trade-offs depending on the application scenario and the available prior information.

[7181] *Autogenous Spatialization for Arbitrary Loudspeaker Setups*

Zeyu Yang (Technical University of Berlin) and Henrik von Coler (Technical University of Berlin).

Autogenous spatialization is an experimental approach for distributing audio signals to loudspeaker systems, based only on the signals' properties and the geometry of the loudspeaker setup. Time-domain and frequency-domain audio features of incoming audio streams are extracted and serve as control signals. The audio streams are then distributed to individual loudspeakers, based on a mapping between these signals and properties of the loudspeakers, such as position and orientation. Results of the approach can range from a localizable shifting between loudspeakers to alienating modulation effects and spatial sound synthesis. The proposed system is implemented as a C++ toolbox to allow the integration into various ecosystems for music production and sound design. External objects are provided for Pure Data, which allows the flexible testing and exploration of the spatialization approach.

[7389] *The workmanship of luthiers in the house of violin: The Auditorium G. Arvedi of Cremona and the acoustic features suggested by Toyota and Nagata*

Antonella Bevilacqua (University of Parma), Alessandro Martinetti (Gruppo CSA SpA Rimini) and Giacomo Tentoni (Gruppo CSA SpA Rimini).

The Auditorium Arvedi of Cremona has been realized inside a shoebox room adapted to the needs of musical performances. The project designed by the architects Palù & Bianchi has been shaped by the engineer Toyota and Nagata, who provided to perform the acoustics at a high-quality level. The height of the room has been increased above the stage by excavating the original floor. Further adjustments have been created by a suspended wooden structure to allocate the seats having the function of a big resonance box. The void created around the stage determines the shape of a sphere that seems to break the finite space of the shoebox, modeled with convex segments on the ceiling. Acoustic measurements have been carried out inside the Auditorium according to ISO 3382-1. Evaluation of the main acoustic parameters has been compared with a 3D model of this performing arts space which follows the soft profile of a violin in an original interior design.

[7612] *Effects of Modified Late Reverberation on Audio-Visual Plausibility and Externalization in AR*

Christian Schneiderwind (Technische Universität Ilmenau), Maike Richter (Technische Universität Ilmenau), Nils Merten (Brandenburg Labs GmbH) and Annika Neidhardt (Technische Universität Ilmenau).

Binaural synthesis systems can create virtual sound sources that are indistinguishable from reality. In Augmented Reality (AR) applications, virtual sound sources need to blend in with the real environment to create plausible illusions. However, in some scenarios, it may be desirable to enhance the natural acoustic properties of the virtual content to improve speech intelligibility, alleviate listener fatigue, or achieve a specific artistic effect. Previous research has shown that deviating from the original room acoustics can degrade the quality of the auditory illusion, often referred to as the room divergence effect. This study investigates whether it is possible to modify the auditory aesthetics of a room environment without compromising the plausibility of a sound event in AR. To accomplish this, the length of the reverberation tails of binaural room impulse responses are modified after the mixing time to change reverberance. A listening test was conducted to evaluate the externalization and audio-visual plausibility of an exemplary AR scene for different degrees of reverberation modification. The results indicate that externalization is unaffected even with more

extreme modifications (such as a stretch ratio of 1.8). However, audio-visual plausibility is only maintained for moderate modifications (such as stretch ratios of 0.8 and 1.2).

[7708] ***Software Tools for Flexible Control of Radiation Synthesis***

Thibaut Carpentier (STMS Lab, IRCAM, CNRS, Sorbonne Université), Olivier Warusfel (STMS Lab, IRCAM, CNRS, Sorbonne Université) and Jean-Marc Jot (Virtuel Works, LLC).

Compact spherical loudspeaker arrays are new tools for the creation of auditory objects in a space. Originally designed for musical and performance purposes – as a substitute for conventional cabinet loudspeakers lacking naturalness in playback – they intent to promote spatial presence and to emulate lifelike directional features of natural sources such as acoustic instruments and performers. They are typically used to reproduce the measured radiation pattern of sound sources, or to steer directional sound beams in order excite the acoustic environment. The directivity pattern of the sound beams can be adjusted, in orientation and shape, by modal beamforming techniques. In electroacoustic music, it is most often desirable to apply dynamic beamforming, i.e. to vary the directional attributes over time, in order to effectively enable a sense of spatial presence or immersion. This paper presents a software tool, intended for creative applications, for the flexible design and intuitive manipulation of spherical beampatterns. In particular, we propose a simple parametric model that allows to synthesize a variety of high-order radiation patterns, and to smoothly transition between shapes. The tool is implemented in the Max environment, making it easy to extend and to interface with remote controllers for interactive applications.

[7762] ***Treble Auralizer: a real time Web Audio Engine enabling 3DoF auralization of simulated room acoustics designs***

Alessia Milo (Treble Technologies), Jóhannes F. Einarsson (Treble Technologies), Úlfur Einarsson (Treble Technologies) and Finnur Pind (Treble Technologies).

We present with this paper an original system designed to experience spatial impulse responses generated by our acoustic simulation framework directly on the web. The user can set the simulation parameters (source-receiver positions, materials, simulation settings) and visualize the simulation results in the web application. Once the simulation data is obtained, the user can experience in 3DoF the dynamic binaural rendering of the Second Order Ambisonics Room Impulse Responses (SOA-RIRs) for each source-receiver pair, individually or with multiple sources at once. The original anechoic tracks from the database provided are convolved in real time with the SOA-RIRs and the source levels can be adjusted to create accurate soundscapes representing the acoustic design scenarios. The user can switch seamlessly between receiver positions and compare simulation results from different geometries or material configurations. In this paper we describe the system details and its technical limitations. Finally, we discuss its application for the acoustic engineering field, where perceptually comparing results from different room acoustic solutions can support the analysis of numerical or graphical reports, and thus shorten the time needed to find optimal design solutions.

[7787] ***Towards the evaluation of marine acoustic biodiversity through data-driven audio source separation***

Michele Mancusi (Sapienza - University of Rome), Emanuele Rodolà (Sapienza - University of Rome), Silvia Zuffi (IMATI-CNR) and Nicola Zonca (Studio Arki).

The marine ecosystem faces alarming changes, including biodiversity loss and the migration of tropical species to temperate regions. Monitoring underwater environments and their inhabitants is crucial, but challenging in vast and uncontrolled areas like oceans. Passive acoustics monitoring (PAM) has emerged as an effective method, using hydrophones to capture underwater sound. Soundscapes with rich sound spectra indicate high biodiversity, soniferous fish vocalizations can be detected to identify specific species. Our focus is on sound separation within underwater soundscapes, isolating fish vocalizations from background noise for accurate biodiversity assessment. To address the lack of suitable datasets, we collected fish vocalizations from online repositories and captured sea soundscapes at various locations. We propose an online generation of synthetic soundscapes to train two popular sound separation networks. Our study includes comprehensive evaluations on a synthetic test set, showing that these separation models can be effectively applied in our settings, yielding encouraging results. Qualitative results on real data showcase the model's generalization ability. Utilizing sound separation networks enables automatic extraction of fish vocalizations from PAM recordings, enhancing biodiversity monitoring and capturing animal sounds in their natural habitats.

[7792] ***Spatial Sampling in Mixed Reality: A Review of Ten Years of Research and Creation***

Gregory Beller (HfMT Hamburg).

Spatial Sampler XR is a new musical instrument linking gesture capture to sound processing. In the same way that a sampler is an empty keyboard filled with sounds, Spatial Sampler XR uses gesture capture to transform the surrounding physical space into an area of keys for, recording, indexing and playing back samples. Spatial Sampler XR let the musician arrange the sound around him or her through gesture, creating a spatialized and interactive soundstage. A virtual reality headset adds to the instrument the ability to visualize the layout of sounds. The 3D immersion greatly facilitates their organization and increases the precision of the interaction. Several modes of play are possible and the interaction modalities vary according to the type of performance and the number of performers. This article first introduces the Synekine project, a ten-year research project from which the concept of spatial sampling is derived. It presents the technical devices used, the instruments created, the different modes of play in performative situations. The instrument relies on movement to link time and space. Thus, the Spatial Sampler XR is particularly suitable for movement artists as well as for extra-musical applications.

[7975] ***Design of an Active Noise Reduction System for a Cogeneration Plant***

Emanuele Voltolini (University of Parma), Daniel Pinardi (University of Parma), Andrea Toscani (University of Parma), Marco Binelli (University of Parma), Angelo Farina (University of Parma), Jessica Ferrari (University of Parma), Stefano Maglia (AB Impianti Srl), Andrea Zenaro (AB Impianti Srl) and Enrico Calzavacca (AB Impianti Srl).

Active noise control (ANC) aims at reducing the amplitude of a primary sound wave, by emitting a controlling sound wave through one or more secondary sources, so that the two waves sum out of phase at the listening point. To generate such out of phase signal, the ANC system requires a specific Digital Signal Processing (DSP) algorithm. Applications of ANC systems cover a wide range of problems from noise reduction in industry or vehicle interiors to consumer products, such as headphones. In this paper, an application of structural ANC is investigated. It makes use of electro-dynamic shakers to reduce the noise generated by a cogeneration plant. The algorithm employed for generating the cancelling signal is an offline, feed-forward, filtered-X, Normalized Least Mean Square (FxNLMS). At first, primary and secondary path measurements are presented. Then, several algorithm configurations are evaluated, starting with a single reference, single-input, single-output (SISO) scheme, and ending up with a multiple reference, single-input, multiple-output (SIMO) scheme. Furthermore, a comparison between the theoretical model and the proposed FxNLMS algorithm is shown. The designed architecture demonstrated remarkable performance, with a noise reduction up to 4 dB(A) in the frequency range 50 Hz – 250 Hz.

[7976] ***Time-domain local wave field synthesis of virtual point sources***

Nara Hahn (University of Southampton), Frank Schultz (University of Rostock) and Sascha Spors (University of Rostock).

Wave field synthesis is a sound field synthesis technique that is based on high-frequency and far-field approximations of the Kirchhoff-Helmholtz integral equation. The loudspeaker driving function is given as the directional gradient of the sound field followed by a spatial windowing which ensures that the synthesized sound wave propagates in the desired direction. In practical systems, where the loudspeakers are placed on discrete positions, the synthesized sound fields commonly exhibit spatial aliasing artifacts, introducing errors at high frequencies. Recent studies showed that the synthesis accuracy can be improved inside a local area at the cost of stronger errors outside it. This can be achieved by using a spatial band limitation, where the target sound field is expanded in terms of spherical harmonics up to a finite order. In this paper, we consider the synthesis of a virtual point source whose modal impulse responses are described

by Legendre polynomials. The time-domain driving functions are obtained by performing a directional gradient to the modal impulse responses and applying a spatial windowing in the time domain. This yields FIR filters with coefficients given in closed forms. The proposed method is numerically evaluated for 2.5-dimensional scenarios, where the loudspeakers are arranged on a horizontal plane. It is demonstrated that the accuracy of the reproduced sound field is improved around a freely-chosen expansion center.

[8006] ***Acoustic exploration inside the mosque of Sidi Soufi in Bejaia, Algeria***

Feriel Saidane (Laboratory ETAP, university of Saad Dahleb, Blida 1, Algeria) and Gino Iannace (Department of Architecture and Industrial Design, University of Campania Luigi Vanvitelli, 81031 Aversa, Italy).

The majority of Algerian mosques, old or contemporaneous, have not been studied under an acoustic perspective. Sidi Soufi is one of the oldest mosques in the town of Bejaia, built in the 16th century following the liberation from Spanish rule, and rebuilt during the French colonisation. The prayer room has a rectangular plan, composed of six naves of 3,80m each facing the qibla wall, and three spans (one central and two lateral), with a height of 6m. The side walls are punctuated by twin windows composed of an overstepped semicircular horseshoe arch, which illuminate the room, in addition to the dome and the skylight. This paper deals with the determination of acoustic proprieties of the prayer room of Sidi Soufi's Mosque though acoustic measurements, which were carried out according to ISO 3382-1 standard, in unoccupied conditions, and by placing an omnidirectional sound source close to the Mihrab.

[8019] ***Characteristics of the Teatro India "Sala A" of Rome: new acoustic perspectives***

Ruoran Yan (University of Bologna), Cobi van Tonder (University of Bologna) and Lamberto Tronchin (Univerity of Bologna).

Construction of innovative theaters that give emerging performances the necessary acoustic conditions is a result of the fact that times are changing, and new performances are emerging. The conventional acoustic parameters of the Sala A of the Teatro India in Rome are first gathered and examined in this study using measurements that follow to ISO 3382-1's specifications. A combination of multi-channel microphones and a panoramic camera is also utilized to visualize the arrival direction and strength of sound reflections. The acoustic characteristics of Sala A were documented and its acoustic quality was assessed, and guidance was provided for the construction of different scenarios.

[8155] ***Designing for Spatial Sound in a Challenging Auditorium Renovation***

John O'Keefe (O'Keefe Acoustics).

The York Memorial Auditorium in Toronto, Canada burnt to the ground on 7 May 2019. The original building, erected in 1929, was typical of the early 20th century architectural style for auditoria in Canada – wide and flat. The spatial experience of the sound in this room, and so many others like it, suffered accordingly. Two key elements in the restoration design will address this concern. The coffered ceiling will be opened up to significantly to increase the overall height of the room and, in so doing, improve both the Reverberation Time and the Early Decay Time/Reverberation Time ratio. Lateral reflections will also be improved with the inclusion of reflectors inside the newly available ceiling space. The reflectors have been designed with the aid of a novel Genetic Algorithm (GA) routine that has been developed by the au-thor. Unlike previous routines, which optimise reflections from one point to another or from one point to a zone of points, the new GA optimises reflections from one zone of source points to another zone of receiver points.

[8180] ***Evaluation of Smart Glasses Rendering of Binaural Spatial Audio***

Huiyuan Sun (The University of Sydney), Craig Jin (The University of Sydney), Chin-Teng Lin (University of Technology, Sydney), Minh Nguyen (University of Technology, Sydney) and Howe Zhu (University of Technology, Sydney).

Smart glasses are an emerging technology that can offer an immersive augmented reality experience to users in their daily lives. While the high-quality rendering of binaural spatial audio using earphones/headphones has been well-studied using Head-Related Transfer Functions, the rendering of binaural spatial audio using smart glasses has been less well researched. Indeed, the quality of binaural spatial audio rendered using smart glasses varies widely. In this work, we explore the performance issues related to rendering high-fidelity binaural spatial audio using smart glasses. In particular, we explore the acoustic transfer function between the loudspeakers mounted in the smart glasses and the participant's ears and use signal processing methods to try to compensate for this acoustic path as best as possible. We evaluate the binaural spatial audio performance of off-the-shelf smart glasses using both numerical simulations and perceptual experiments and compare with industry standard earphones/headphones. Listening tests are conducted for a range of sound conditions. We report on the results of our numerical and psychophysical studies.

[8185] *On the relevance of acoustic measurements for creating realistic virtual acoustic environments*

Siegfried Gündert (Carl von Ossietzky Universität Oldenburg, Cluster of Excellence, Hearing4all), Stephan Ewert (Carl von Ossietzky Universität Oldenburg, Cluster of Excellence, Hearing4all) and Steven van de Par (Carl von Ossietzky Universität Oldenburg, Cluster of Excellence, Hearing4all).

Geometrical approaches for room acoustics simulation have the advantage of requiring limited computational resources while still achieving a high perceptual plausibility. A common approach is using the image source model for direct and early reflections in connection with further simplified models such as a feedback delay network for the diffuse reverberant tail. When recreating real spaces as virtual acoustic environments using room acoustics simulation, the perceptual relevance of individual parameters in the simulation is unclear. Here we investigate the importance of underlying acoustical measurements and technical evaluation methods to obtain high-quality room acoustics simulations in agreement with dummy-head recordings of a real space. We focus on the role of source directivity and the spatial distribution of absorption coefficients at the wall boundaries. A database of binaural room impulse response (BRIRs) has been recorded for various source distances and source orientations. The effect of including measured source-directivity patterns and wall absorption coefficients using the room acoustics simulator RAZR was assessed in comparison to the measured (reference) BRIRs. Technical evaluation strategies to verify and improve the accuracy of various elements in the simulation processing chain from source, room properties, to the receiver's head related transfer functions are presented.

[8270] *Physically accurate binaural reproductions from broadband wave-based room acoustics simulations, and comparison with measurements*

Matthias Cosnefroy (Treble Technologies), Steinar Guðjónsson (Treble Technologies) and Finnur Pind (Treble Technologies).

The ability to perform wave-based room acoustics simulations with a listener embedded into the scene can be valuable, for instance, to obtain accurate binaural renderings. Accounting for the listener directly in the simulations however remains challenging due to geometrical constraints, and is also impractical since the orientation is fixed and must be known in advance. We present a broadband wave-based acoustics simulation framework to estimate spatial room impulse responses with high accuracy, where the sound field is simulated with the discontinuous Galerkin method and sampled using an array of receivers designed to operate over a large frequency range. The data is then encoded into spherical harmonics contributions, resulting in a very high-order ambisonics formulation (e.g., up to order 16), which can be combined with free-field head-related transfer functions (HRTF) for binaural reproductions with a physical accuracy. The approach is validated against original binaural room impulse response measurements for different source-receiver configurations; an excellent agreement is found with the simulations.

[8274] *Acoustic evaluation of some churches located in south of Italy*

Ilaria Lombardi (Università della Campania Luigi Vanvitelli), Amelia Trematerra (Università della Campania Luigi Vanvitelli), Silvana Sukaj (University of Tirana (UET)) and Giovanni Amadasi (SCS-ControlSys - Vibro-Acoustic, Padova, Italy).

The churches represent a building type acoustically very complex given their dimensions, the reflecting finish materials of the envelope and the volume composition including domes and niches. This paper deals with the acoustic analysis of same churches located in Campania (south of Italy), under a reverberation perspective other than clarity and speech comprehension. The acoustic measurements have been undertaken according to ISO 3382-1. The speech definition has been assessed in line with the Vatican Council II guidance, based on a good verbal communication to be determined by a balance among all the acoustic parameters. A further evaluation has been made on the acoustic solutions that aim to adjust the measured values to be more suitable for speech comprehension

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[8365] ***New perspectives in virtual environments for opera music***

Andrea Bareggi (Opera Network; ESME School of Engineering; Conservatory "C.Monteverdi" of Cremona), Federico Bardazzi (Conservatory "G. Puccini" of La Spezia), Lamine Amour (ESME School of Engineering) and Michal Ostrowicki (Jagiellonian University).

Digitalisation in classical western music is an area of ongoing research, educational innovation and socio-political interest. Several Erasmus+ and Horizon projects are studying and experimenting new solutions for a philological and technological innovation in academic and performing environments of classical music. In this context, opera music is the crossroad of aesthetic, technological and performing ability of artists and artistic communities. As a blended art form of music, theatre, visual arts and literature, opera is one of the more complete art forms in live performance. Technology played an important role in the making of operas, so that networked technologies, XR and virtual environments cannot be ignored in the interpretation of the operas of the past and in the creative process of a new musical theatre. Virtual environments include visual and sound context of opera music; these contexts have to deal with technological issues and limitations - such as latency in the audio signal also give the opportunity to offer a more immersive experience in opera music, and the opportunity to reduce the carbon footprint by using virtual music and stage rehearsals. This paper presents the results of the Erasmus+ project Virtual Stage and the preliminary research perspectives of the Horizon project CAPHE, led by Jiagellonian University of Krakov, including modern techniques such as Networked Music Performance for music rehearsals and 3D graphic environments for stage rehearsals

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[8461] ***An investigation on the spatial adaptation of an artistic performance in contemporary churches***

Louena Shtrepi (Politecnico di Torino), Angela Guastamacchia (Politecnico di Torino), Arianna Astolfi (Politecnico di Torino) and Marco Masoero (Politecnico di Torino).

The contemporary church acoustics is affected by different aspects related to the creativity of the designer. Despite the importance of the role of acoustics in the execution of the main functions, the optimal acoustic conditions are not always verified, resulting in much more reverberant space. Very often this is due to the use of highly reflective materials and very large and complex volumes. This work presents the analysis of objective parameters in four contemporary churches located in the city of Milan based on ISO 3382-1 procedure. Moreover, further measurements of the directional properties of the sound fields in the four spaces has been performed using a spherical microphone array. The main focus is to investigate the acoustic conditions of four places chosen for the execution of a contemporary performance from six artists. The performance is modified and adapted to the places where it is performed in order to create the same perception and spatial sound experience while maintaining its artistic and aesthetic qualities unchanged. The results show significant differences of the objective acoustic parameters between the four spaces ranging for Reverberation Time 3÷5 s, Early Decay Time 2.8÷5 s, Definition 0.18÷0.29, Clarity -2.2÷-5.8 dB, and Center Time 188÷347 ms. The spherical microphone array measurements have been analyzed using beamforming techniques and processed for third-order Ambisonic reproduction in order to conduct listening tests involving the artists.

[8497] ***Noncontact Measurements of Sound Absorption Coefficient with a Pressure-velocity Probe, a Laser Doppler Vibrometer, and a Microphone Array***

Leonardo Saccenti (University of Parma), Jessica Ferrari (University of Parma), Daniel Pinardi (University of Parma) and Angelo Farina (University of Parma).

Kundt's tube and reverberant chamber are common methods for determining the sound absorption coefficient or acoustic impedance of materials. These measurement methodologies are well-known and standardized, albeit not being practicable in-situ and requiring the isolation of samples of the material under test. Furthermore, Kundt's tube results are affected by sample size, diameter, and length of the tube itself, while reverberant chamber ones by the room dimensions and diffusiveness. In literature, noncontact techniques for sound absorption coefficient and acoustic impedance measurement are widely debated. In this paper, three different noncontact systems for the measurement of the sound absorption coefficient have been investigated: a pressure-velocity probe, a Laser Doppler Vibrometer, and a spherical microphone array featuring 64 capsules. The three methods have been evaluated through in-situ measurements of materials with known acoustic characteristics: Basotect G+ and Expanded Polystyrene. Furthermore, the results obtained with the standard test signal, i.e., white noise, are compared with the exponential sine sweep technique, which provides an increased signal to noise ratio, and allows for removing nonlinear high order distortions and acoustic reflections. As a main contribution of this work, it will be shown that microphone arrays are an optimal solution for measuring the sound absorption coefficient.

[8583] ***Using response time to evaluate noise fluctuations and Lombard speech in auralizations***

Nicola Prodi (Dipartimento di Ingegneria, Università di Ferrara) and Chiara Visentin (Dipartimento di Ingegneria, Università di Ferrara).

This work presents developments on the usage of a viable measure of listening effort, that is the response time, in the framework of auralization. The approach is confirmed to be informative and suitable to evaluate the current means of acoustical renderings. In particular the manipulation of the noise signals and of the speakers' spectrum is introduced in the auralization process. It is shown that, even in conditions where speech intelligibility is kept constant, the response time can disclose the listener effort according to the required processing load to achieve the given accuracy of reception. Applications to the acoustical design of rooms for speech communication are discussed.

[8664] ***Acoustics and prayers: investigations on the Great Mosque of Tirana, Albania***

Silvana Sukaj (Department of Engineering and Architecture, European University of Tirana (UET), 1000 Tirana, Albania), Antonella Bevilacqua (Department of Industrial Engineering, University of Parma, 43124 Parma, Italy), Gino Iannace (Department of Architecture and Industrial Design, University of Campania Luigi Vanvitelli, 81031 Aversa, Italy) and Feriel Saidane (Laboratory ETAP, university of Saad Dahleb, Blida 1, Algeria).

The Great Mosque of Tirana, also called Namazgâh Mosque, is the largest mosque in the Balkans. After the end of communism, the mosque was requested by numerous Muslim citizens, as there was a lack of a religious building to pray in. The mosque is characterized by 50 m high minarets, crowned on the perimeter by a 35 m high central dome surmounted by six niches with half domes. The Great Mosque has an area of 600 m2, whose floor is covered with a precious carpet, and the interior is illuminated by large windows on the side walls. This paper deals with the acoustic characterization of the main prayer room of the Great Mosque and the comparison with the values measured in other mosques. A short digression of the historical background, as well as the measurement techniques according to ISO 3382, have been given.

[8724] ***Ambisonic room impulse responses extrapolation guided by single microphone measurements***

Yake Fan (Center of Intelligent Acoustics and Immersive Communications, Northwestern Polytechnical University) and Wen Zhang (Center of Intelligent Acoustics and Immersive Communications, Northwestern Polytechnical University).

Sound fields and room impulse responses represented using higher-order-Ambisonics (HOA) is an efficient way to generate six-degree-freedom (6DoF) binaural rendering. However, measuring the room impulse responses of the entire space is impractical in reality. While many methods have been proposed for Ambisonic room impulses response (ARIR) interpolation and extrapolation, in this work, we propose a robust ARIR extrapolation method by combing the ARIR, which is measured at a fixed point inside the room, with the monaural RIR measured at the extrapolation point. The ARIR is decomposed into a direct part, distinct early reflections, and the residual reverberation. For the direct and early reflection parts, extrapolation is performed by estimating the source positions based on a parametric analysis of the measured ARIR and monaural RIR. The extrapolation of the residual reverberation is achieved by manipulating the monaural RIR, where the weights are derived from ARIR measurements. Finally, objective and subjective evaluations are conducted to evaluate the accuracy and robustness of the proposed method. The results show that the proposed method approximates the true ARIR in terms of objective indicators and subjective perception.

[8834] *Towards a Data Driven Panning Algorithm for Visually Impaired Audiences*

Michael McLoughlin (University of York), Mariana López (University of York), Krisztián Hofstädter (University of York) and Gavin Kearney (University of York).

In stereo sound systems, the listener's ability to localise sound in the reproduced stereo field is affected by a variety of factors such as their listening position and the degree of separation between the loudspeakers. Understanding the effect of these factors is important for creating object-based mixes, where the panning position and volume of sounds can be adjusted to suit the user at the point of media consumption. Being able to control the panning and volume of different sources in the renderer is especially useful for visually impaired audience members, as it can allow for greater immersion and narrative understanding for film and television. For this reason, it is important to conduct listening tests on how these audiences perceive sound over stereo systems. These listening tests can then be used to inform renderer panning algorithms. However, before inviting visually impaired people to take part in listening tests, a robust methodology and control group comparison must first be established. Here, we present an algorithm that is driven by the results of a Minimum Audible Angle (MAA) staircase listening test for different listening positions and loudspeaker base width combinations. This consisted of 22 trials across 21 sighted participants. We calculated the mean MAA for these 22 trials and used these results to determine our panning angle when given a listener position and stereo base width. We then carried out a smaller listening test to determine the effectiveness of a perceptually informed rendering algorithm (8 tests across 8 participants). We achieved this by calculating the F-Score of our method against a control method that does not account for these factors. The mean F-score was 0.62 for our method against a mean of 0.36 for our control. Our results indicate that our algorithm has the potential to be used for object-based mixes where spatial separation for all audience members is more important than precise object placement. Future research will carry out the same experiment with visually impaired audience members, and compare the results with this pilot study.

[8942] *DUET combined with HRTF mask*

Ryota Shimokura (Graduate School of Engineering Science, Osaka University), Yuna Kitano (Graduate School of Engineering Science, Osaka University) and Youji Iiguni (Graduate School of Engineering Science, Osaka University).

Online meetings have become increasingly common following the COVID-19 pandemic, and several meeting places are now often connected remotely. This study proposes a digital signal processing method for controlling the sound localization of speech delivered from remote sites. To enable the control of simultaneous speech, our proposal introduces a blind source separation algorithm called the Degenerate Unmixing Estimation Technique (DUET), which takes stereo recordings obtained through a two-channel microphone as its input. During the process of DUET, we estimate the azimuth angles of the speakers using the amplitude attenuation and delay between the stereo recordings, and multiply the estimated binary mask and corresponding Head Related Transfer Function (HRTF) by the estimated azimuth angle (i.e., HRTF mask). Speaker localization tests verify that the HRTF mask can separate speech in the desired positions when two or three speakers are talking at the same time.

[9012] *User expectation of room acoustic parameters in VR environments*

Benjamin Burnett (University of Surrey), Annika Neidhardt (Technische Universität Ilmenau), Zoran Cvetkovic (King's College London), Huseyin Hacihabiboglu (Middle East Technical University) and Enzo De Sena (University of Surrey).

This paper explores how visual attributes of a VR scene affect user expectations of room reverberation. A psychoacoustic experiment was run wherein subjects wore a VR headset and adjusted two unlabelled sliders controlling the reverberation time (T60) and the acoustic room size until the reverberant response was closest to their expectation of how the room they were seeing should sound. Different visual characteristics, in particular, room type and size, surface material, and furnishing were modified to determine how these might affect their expectations of the reverberant response. Results showed that visual room size had a significant effect on both the expected T60, in agreement with previous literature, and on the expected acoustic room size. Both relations seem to be well-described by a simple sublinear power law model, which could be used, for instance, to design reverberation time (T60) and acoustic room size values that align well with listeners' expectation for a given visual room volume. Differences in visual surface materials were found to have a statistically significant effect on the expected T60. The level of visual furnishing, on the other hand, only had a marginally significant effect on the expected T60. The results also indicate considerable subjective differences in individual expectations.

[9048] *ADM-OSC: an industry initiative for communicating object-based audio data*

Michael Zbyszyński (L-Acoustics), Hervé Déjardin (Radio France), David Marston (BBC) and Guillaume Le Nost (L-Acoustics).

Spatial and immersive audio has become increasingly mainstream, presented in concert halls and more recently (Apple: June 2021) through music streaming services. There is a diverse ecosystem of hardware and software controllers and renderers in both live and studio settings that would benefit from a standardized communication protocol. Since 2021 a growing group of industry stakeholders has been working to develop ADM-OSC to fill this need. ADM-OSC proposes a standard for transmitting position data for object-based audio by implementing a namespace in parallel with the Audio Definition Model (ADM), a metadata standard developed in the broadcast industry. OpenSoundControl (OSC) is a well-established data transport protocol developed for flexible and accurate communication of realtime performance data. By leveraging these open standards, we have created a lightweight specification that can be easily implemented in audio software, plugins, game engines, consoles, and controllers.

This paper will discuss the design principals of ADM-OSC, developed to meet the needs of specific stakeholders and use cases. The core address space for position data is described, including a solution for representing both physics-based and cinematic distance. Proposed extensions to the next version of the specification will add comprehensive support for additional broadcast use cases, including different types of audio content (e.g., different languages or assistive descriptions), user interactivity, and cues for reducing channel count. We conclude with an overview of future ADM-OSC development, including next steps in bringing together ideas and discussion from multiple industry partners.

[9171] *Listening to Venice - A Thousand Echoes*

Cobi van Tonder (Bologna University), Giulia Vismara (Academy and Conservatory of Antwerp) and Angela McArthur (University of Greenwich).

The soundscapes of Venice are as mesmerizing as the architecture, history, art, and contemporary life in the lagoon city. Time, embedded in the marble over thousands of years, seems set in stone and simultaneously shifting with the tides. Both ephemeral and solid: a city in constant conversation and awareness of its past. We capture impulse responses during the city's quiet late nights. We take unusual perspectives, we record various points in time and space during the day when church bells ring in marvellous counterpoint across the grand canal, across big and smaller squares. We include field recordings of thousands of footsteps, everyday voices on the streets, and the water as it touches the surfaces of boats, docks, foundations, and the acoustics of the seabed. We end with a map, sonic impressions, and multiple perspectives on working with site-specific immersive audio. Next, we use this backdrop for three compositions:

Colonising Venice (sending water sounds into virtual churches), Time-Space Morphology (morphing through space via sound), and Erosa (weaving perspectives far and close).

[9187] *Outdoor Human Comfort Exploration: A Multi-Physical Approach with Sound, Light, and Heat Panoramic Measurement*

Jairo Acuña Paz Y Miño (Urban Physics Joint Laboratory), Inès de Bort (Urban Physics Joint Laboratory) and Benoit Beckers (Urban Physics Joint Laboratory).

Panoramic photography is a technique that allows images to be stitched together to extend the field of view of a scene, regardless of the camera's optics. In this paper, this technique is adapted to acoustic cameras to explore the extended acoustic field. This appears as an interesting alternative to measure an impulse response in a complex architectural scene, or to study a large urban scene. The temporal and frequential spatialization of sound reflections allows us to identify phenomena such as multiple reflections. These results can be compared to raytracing simulations. This same panoramic representation, extended to other physical fields such as light and heat, allows us to have a global and multi-physical vision of the comfort of a person. This method opens the way to immersive experiences in which the whole perceptible environment is taken into account.

[9220] *A Study on Accessibility and Sound in Cultural Heritage Contexts in Ireland*

Caitlin Kelly (Cork School of Music Munster Technological University, Advance CRT).

Ireland is a place with a colourful and expansive cultural heritage (CH). From our tangible heritage such as Newgrange, our native instruments, and the Book of Kells, to the intangible heritage of our myths, legends, and oral traditions, to name a few. However, many of the country's tangible CH sites remain if not partially, then completely inaccessible to the disabled community.

While Ireland has ratified the UN Convention on the Rights of a Person with Disabilities, and therefore acknowledges that equal access to Cultural and Leisure activities for disabled people is a right, change can be slow, and the ratification of this document took eleven years.

This paper discusses potential solutions for improving accessibility outcomes in CH sites. Specifically, it suggests using immersive augmented 3D sound, ambisonics, and Sound Design as a facilitator for inclusive heritage experiences that elicit the same sense of "wonder and awe" as seeing these sites does. Though there is a lack of sound design utilised in this way in Irish CH sites, more specifically rural sites, there is precedent for using sound design as an accessibility tool in other fields (e.g. gaming). This paper outlines the current state of CH and Sound Design in Ireland, and reflects on how the research following could potentially improve this.

[9267] *Multichannel mobile audio recordings for spatial enhancements and ambisonics rendering*

Nikolaos Vryzas (Aristotle University of Thessaloniki), Marina Eirini Stamatiadou (Aristotle University of Thessaloniki), Lazaros Vrysis (Aristotle University of Thessaloniki) and Charalampos Dimoulas (Aristotle University of Thessaloniki).

Ambisonics recording allows the playback of a captured 3D audio scene. In many cases, higher-order ambisonics or a set of first-order ambisonics arrays may be used for better results. Virtual ambisonics synthesis and reproduction allows the design and rendering of an audio scene by providing the sound source signals and positions. In this paper, an extended array is proposed, comprising a first-order ambisonics soundfield array, and an arbitrary non-calibrated spatial smartphone-microphone array. The main objective is to investigate whether this extended setup can enhance the spatial resolution of the first-order soundfield array without dramatically increasing the system complexity. The smartphone array provides additional information concerning the relative direction of arrival of different sources, through the calculation of a vector containing the time difference of arrival for all available microphones. A T-SNE visualization is used to show the separability of these vectors that expresses the system's spatial and temporal resolution. Resolution proves to be higher than that of a first-order soundfield array compared to an experiment

conducted in previous studies. This supports the idea that the proposed extended array can enhance the resolution of Ambisonic Energy-Based Localization methods.

[9290] *Nightports at Hull Minster: Physical, Hybrid and Virtualized Live Loudspeaker Array Spatialization of Electronic Music Performance*

Matt Barnard (University of Hull), Mark Slater (University of Hull) and Adam Martin (University of Huddersfield).

‘Nightports at Hull Minster’ is a musical project that harnesses spatialization techniques to present music composed of the sounds of Hull Minster, UK, in both the location itself and alternative performance spaces, whilst still expressing the spatiality of the location. The project involves a live electronic music performance by Nightports (The Leaf Label), using only sounds recorded in the Minster itself, spatialized in real-time by another performer across a 25-loudspeaker array. In total, three variant approaches are detailed: a physical acousmonium in-situ; a hybrid acousmonium and virtualmonium; and headphone-targeted virtualizations. To achieve virtualization, spatial room impulse response measurements were taken across the diffusion array to capture the characteristics of the loudspeaker arrangement implemented in the Minster. A scalable and adaptable spatialization system was devised to realize the different iterations and is detailed in this paper, alongside observations on the performative ramifications, compositional preoccupations, and potential developments of the project.

[9332] *Navigable reconstruction of reverberant sound fields using distributed microphone arrays*

Antonio Figueroa-Duran (Technical University of Denmark) and Efren Fernandez-Grande (Technical University of Denmark).

The reconstruction of sound fields over large domains is a challenging task that often requires a large number of measurements. To overcome this challenge, sound field reconstruction methods make use of additional knowledge to enable sound interpolation and extrapolation from a reduced number of measurements. In this study, we propose a method that leverages the generalisable time structure of a room impulse response. The approach accounts for the different structure of early and late parts of the room impulse response as well as reflection density and decay over time. In the early part, the strongest reflections are localised and modelled with a cluster of weighted monopoles. For the late part, kernel methods are used to reconstruct the sound field based on its statistical properties and temporal decay. The prediction ability of the method is compared with a classical plane wave reconstruction, based on experimental measurements in an auditorium. The results indicate that the approach is suited to reconstructing sound fields over large domains.

[9367] *A Two-Dimensional Threshold Test for Reverberation Time and Direct-to-Reverberant Ratio*

Nils Meyer-Kahlen (Reality Labs Research), Sebastià Amengual Garí (Reality Labs Research), Ishwarya Ananthabhotla (Reality Labs Research) and Paul Calamia (Reality Labs Research).

In audio for augmented reality (AR), virtual sources should be incorporated into a real acoustic environment, using reverberators that ideally match the properties of the real acoustic space. One way of assessing the closeness of this match is to compute differences in acoustical parameters between measurements and renderings and to compare them to just noticeable difference (JNDs) obtained from perceptual tests. However, JNDs for even the most common acoustical parameters are not well established and parameters that can be independently varied in the reverberator might not be perceptually independent. Here, we present a two-dimensional threshold test varying reverberation time (RT) and direct-to-reverberant ratio (DRR) of a spatial room impulse response (SRIR) rendered dynamically to headphones. Speech signals are used as stimuli. We show such a two-dimensional threshold test, using active stimulus selection. For this, a Gaussian Process Classification (GPC) model is fit to the listener response data throughout the test. For each trial, samples that are close to the 75% detection threshold currently predicted by the model are selected. We

show first results and discuss the shape of the estimated threshold curves and its variability between listeners. Finally, we discuss the implications for the broader context of sound in AR.

[9422] *Acoustic characterisation of hosiery factory waste materials*

Edoardo Alessio Piana (University of Brescia).

Hosiery factory by-products quite often cannot be recovered and are considered waste materials. Such elements are in the form of fabrics, wool having different densities or pellets. This research aims to characterise such materials from an acoustic point of view, determining their sound absorption and their sound transmission loss. The measurements were performed using a four-microphone impedance tube. Such a method allows us to determine the main acoustic and non-acoustic parameters of the material.

[9652] *Perceptually Motivated Spatial Audio Scene Description and Rendering for 6-DoF Immersive Music Experiences*

Jean-Marc Jot (Virtuel Works, LLC), Thibaut Carpentier (STMS Lab, IRCAM, CNRS, Sorbonne Université) and Olivier Warusfel (STMS Lab, IRCAM, CNRS, Sorbonne Université).

In recorded, live, virtual, or mixed-reality immersive musical experiences, the audio presentation may not be attached to the geometrical and physical model of a room or acoustic enclosure (as would often be expected, for instance, in architectural acoustic design and in video games or virtual reality). In previous work, a generic spatial audio rendering interface was proposed, seeking compatibility with both physically based and perceptually based application-level interactive audio scene representations. This parametric model builds and extends upon prior 6-DoF navigable audio rendering API standards such as OpenAL. It enables prioritizing auditory plausibility over simulation exactness, thereby facilitating low complexity implementation in power limited platforms. The rendering engine can be realized as a generalization of the familiar Digital Audio Workstation (DAW) processing topology, exploiting a shared multi-room acoustic reverberation and reflections processor to minimize per-object computation budget. In this article, we review prior research on the parametrization of the subjective acoustical quality of auditoria and the design of the perceptually based spatialization interface and reverberation engine implemented in IRCAM Spat. We describe a mapping scheme to calculate renderer parameters, enabling the control of important effects such as the percepts of source presence (or "depth") and running reverberance, accounting for the directivity index of a source, or simulating anisotropic reverberation sound fields.

[9671] *Spatial Audio Panning With Elevated Sources Using Horizontal-Only Reproduction Loudspeakers*

Jacob Hollebon (University of Southampton) and Filippo Fazi (University of Southampton).

To reproduce virtual sound sources with elevation often elevated reproduction loudspeakers are required (for example with Ambisonics), or alternatively complex inverse filtering with horizontal-only loudspeakers is needed (for example with crosstalk cancellation). It has recently been demonstrated that, at low frequencies, crosstalk cancellation approximates the stereo sine law, which is a simple frequency-independent panning approach. This paper reinforces this link and analyses it for the 3D scenario considering elevated virtual sources. The classic stereo sine law is adapted to include elevation, which is achieved by mapping elevated sources through a cone of confusion to positions in the horizontal plane with no elevation. Results considering localization cues reproduced by a stereo loudspeaker array using measured data corroborate the findings. The inclusion of dynamic cues, such as through the head-tracked sine law, is suggested to improve the perceptual performance.

[9757] *From Concert Halls to City Streets: Bridging the Gap Between Room Acoustics and Urban Acoustics*

Inès de Bort (Urban Physics Joint Laboratory), Jairo Acuña Paz Y Miño (Urban Physics Joint Laboratory) and Benoit Beckers (Urban Physics Joint Laboratory).

Room acoustics gives us the tools, both computational and technological, to identify the acoustic characteristics of urban designs. We propose to treat the design of a city as that of a concert hall, by studying the role of the shape of buildings on the propagation of sound reflections. This work is based on the case study of Pasaia, a small harbor city in the Spanish Basque Country that presents different urban patterns: a historical center, high towers, a harbor, narrow pedestrian streets, and large traffic lanes. Using devices developed for room acoustics, an acoustic camera measurement, and a ray-tracing simulation, we identify three main urban acoustic patterns that articulate with each other to form the city. The analysis of the results through spatial (panoramic images) and temporal (impulse response) representations of the sound field allows to propose different scenarios for the renovation of the city shape from an acoustic point of view, which is seldom taken into account in projects.

[9794] *Acoustic analysis of the Teatro Minimo of Atri*

Ruoran Yan (Department of Architecture, Bologna), Giacomo Tentoni (C.S.A. Group SpA), Cobi van Tonder (University of Bologna) and Alessandro Martinetti (C.S.A. Group SpA).

Teatro Minimo of Atri was designed by the architect Mazucelli and opened in 1881 with the performance of an Opera by Verdi. The main hall composed of 38 boxes on three orders of balconies and surmounted by a loggione of up to 200 seats reflects the characteristics of an Italian Opera theatre, with its horse-shoe shape in plan.

Acoustic measurements have been carried out in order to assess the acoustic response of the theatre. The analysis of the measured data has been compared with the optimal values of Opera theatres having similar volume sizes. A brief description of the history has been added in order to better understand the geometry and the material selected for the interior design.

[9805] *Multi-Physical Human Comfort Assessment through Integrated Sound, Light, and Heat Panoramic Measurements*

Jairo Acuña Paz Y Miño (Urban Physics Joint Laboratory), Inès de Bort (Urban Physics Joint Laboratory) and Benoit Beckers (Urban Physics Joint Laboratory).

Panoramic photography is a technique that allows images to be stitched together to extend the field of view of a scene, regardless of the camera's optics. In this paper, this technique is adapted to acoustic cameras to visualize the extended acoustic field. This appears as an interesting alternative to measure an impulse response in a complex architectural scene, or to study a large urban scene. The temporal and frequential spatialization of sound reflections allows us to identify phenomena such as eigenmodes and multiple reflections. These results can be compared to ray-tracing simulations. This same panoramic representation, extended to other physical fields such as light and heat, allows us to have a global and multi-physical vision of the comfort of a person. This method opens the way to immersive experiences in which the whole perceptible environment is taken into account.

[9822] *LISTENING IN ANCIENT SPACES: TOWARDS AN AURAL ARCHITECTURE IN THE PAST*

Angela Bellia (CNR, ISPC (National Reserach Council of Italy, Institute of Heritage Science)).

This paper aims to analyse recent studies which have raised new hypotheses concerning aural architecture as an emerging trend in humanities research, with a particular focus on the intersection of sacred space, rituals, and sound in the past. These studies have highlighted how sacred buildings not only defined a sacred place as a physical and symbolic expression of a specific form of worship, but also established the setting for performative and multisensorial ceremonies in which music, dance, and other sonic events played an important role. In this contribution, we investigate studies on aural architecture to explore if the location of sacred spaces indicates whether ancient people reacted to ritual and musical developments by modifying sanctuaries or by designing and constructing new buildings and spaces for performances. This overwiew also takes into consideration how digital technologies and virtual acoustics can help shape our understanding of the architecture-sound nexus.

[9878] ***Auralization of Three-Dimensional Sound Field in an Acoustic Scale Model Using Laser-Induced Sound Sources***

Kazunori Suzuki (Takenaka Research & Development Institute), Shinichiro Koyanagi (Takenaka Research & Development Institute) and Takayuki Hidaka (Takenaka Research & Development Institute).

Auralization of a three-dimensional sound field based on the measurement of room impulse response (RIR) in an acoustic scale model using a sonic impulse excited by a laser-induced air breakdown (LIB) as a sound source is proposed. The LIB as a sound source has a bandwidth of up to about 100 kHz and is an omnidirectional point source, making it suitable as a sound source for acoustic measurements in a scale model. Therefore, the LIB was excited on the stage of a 1/10-scale hall model. Various shapes of sound absorbers were placed on the stage to simulate the directivity of musical instruments and sound absorption by the performers on the stage. Auralization of the three-dimensional sound field was performed using the ambisonics technique. The RIRs were measured at multiple positions by moving a small microphone in small increments around the sound-receiving position, and the directivity of the spherical harmonic functions was synthesized by the superposition of multiple poles. The microphone arrangement was varied according to the target instrument to account for the lower dynamic range and upper-frequency limit when converting to an ambisonics signal. The sound sources of the instruments recorded in an anechoic room were convolved with this ambisonics signal, and a three-dimensional sound field was synthesized by decoding signal processing. A brief sound demonstration will be given at the end of the presentation.

[9891] ***The effect of an audio-video stimulation on emotions: a virtual reality study***

Chiara Burattini (Department of Astronautical Electric and Energy Engineering, Sapienza University), Giulia D'Aurizio (Department of Biotechnological and Applied Clinical Science University of L'Aquila), Giuseppe Curcio (Department of Biotechnological and Applied Clinical Science University of L'Aquila) and Fabio Bisegna (Department of Astronautical Electric and Energy Engineering, Sapienza University).

Working conditions are a frequent cause of stress in industrialized Countries and inducing positive emotions can have the effect to reduce stress level. Previous studies indicated that images, sounds and video can modulate human emotions, inducing positive or negative states. This work investigated the ability of an audio-video stimulation in virtual reality of inducing positive emotions and reducing human stress. A sequence of sounds, images and videos was administrated to ten participants in a virtual environment representing a single office and their emotional states was assessed using Positive and Negative Affect Scale (PANAS), Self-Assessment Manikin (SAM) and Global Vigor Affect Scale (GVAS). Results indicated that the stimulation could elicit positive emotions that affect some stress-related states.

[9981] ***Local Wave Field Synthesis by Temporal Bandlimitation***

Gergely Firtha (Laboratory of Acoustics and Studiotechnics, Budapest University of Technologies and Economics), Nara Hahn (Institute of Sound and Vibration Research, University of Southampton), Frank Schultz (Institute of Communications Engineering,University of Rostock) and Péter Fiala (Laboratory of Acoustics and Studiotechnics, Budapest University of Technologies and Economics).

Wave Field Synthesis (WFS) aims at the reproduction of a desired target wavefront by driving an ideally continuous loudspeaker distribution with properly chosen secondary source driving signals. In practical applications, using a discrete set of loudspeakers degrades the accuracy of reproduction heavily due to the violation of the theoretical requirements. As a result, spatial aliasing wavefronts emerge from the individual loudspeaker elements in addition to the intended virtual wavefront, perceived as strong colouration above the so-called spatial aliasing frequency. Local Wave Field Synthesis (LWFS) approaches improve the reproduction accuracy over a limited listening area by allowing stronger artefacts outside the control region. The present contribution discusses a novel LWFS approach, relying on the transformation of spatially defined antialiasing filters into an equivalent temporal filter bank. The resulting antialiased driving functions ensure aliasing-free synthesis at a predefined listening position at the cost of temporally bandlimited sound field at other listening regions. The results of the proposed approach are compared with a recent LWFS approach employing direct spatial bandlimitation.

[9987] ***Transducer Distribution on Spherical Arrays for Ambisonics Recording and Playback***

Daniel Pinardi (University of Parma), Angelo Farina (University of Parma) and Marco Binelli (University of Parma).

Microphone and loudspeaker arrays are nowadays more and more employed in several applications, such as automotive industry, entertainment, immersive teleconferencing, or remote assistance. The position of the transducers over the surface of the array has a great influence on the beamforming, and so on the spatial performance. In this paper, a recurring geometrical problem is discussed: choosing the optimal locations of transducers for spherical arrays, either microphones or loudspeakers. None of the existing systems is currently relying on a spherical design, or t-design, for the distribution of the sampling points over the sphere. It will be shown that such geometries are an optimal solution for the design of spherical arrays. They are the only known geometries ensuring a lossless transformation back and forth between the two most common spatial audio format: Ambisonics, which makes use of spherical harmonics, and Spatial PCM Sampling, which relies on unidirectional, high directivity virtual microphones.

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