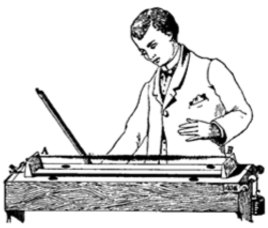
Programme

Abstracts



SMAC 2013

Stockholm Music Acoustics Conference

SMC 2013

Sound and Music Computing Conference

KTH Royal Institute of Technology
Sound and Music Computing Group

Programme

Abstracts

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Stockholm Music Acoustics Conference

SMC 2013

Sound and Music Computing Conference

July 30 – August 3, 2013

KTH Royal Institute of Technology
Sound and Music Computing Group

KTH Computer Science and Communication
Department of Speech, Music and Hearing
2013

Contents

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Preface

Welcome to the fourth Stockholm Music Acoustics Conference (SMAC 2013) and the tenth Sound and Music Computing Conference (SMC 2013). SMAC 2013 continues the series of music acoustics conferences in Stockholm, started 30 years ago. Following the tradition of SMAC 83, SMAC 93, and SMAC 03, SMAC 2013 covers the traditional fields of music acoustics, including musical instruments, singing voice, perception, and physical modeling. The fields of music perception and performance are this time covered by SMC 2013, as well as the rapidly growing research areas of sonic interaction design, sound processing and music information retrieval

A good reason for keeping the broad perspective from earlier SMACs is to offer the possibility of providing at least a partial overview of the many fascinating research areas which address the wonderful combination of performing arts, physics, creativity, and life experience called music. In order to give this perspective SMAC 2013 and SMC 2013 feature a number of invited presentations (the exact number being 13), in which outstanding researchers, old and young, will present overviews of their areas up to and including the research frontier.

SMAC 03 was run in two parallel sections. In the 2013 edition, one of the two sessions is the SMC conference. SMC is rapidly establishing itself as one of the most important conference series in the field of sound and music computing. This year SMC celebrates its 10th edition. SMC 2013 is jointly hosted by the Sound and Music Computing Research Group at the Royal Institute of Technology (KTH) and the Department of Composition, Conducting and Music Theory at the Royal College of Music (KMH) in Stockholm. KTH is responsible for the scientific part and KMH will host the music performances.

The theme for SMAC and SMC this year is “Sound Science, Sound Experience.” During the past five decades, the domain of music acoustics has widened from studies of the acoustics of musical instruments and voice, including basic elements of musical perception and performance, to investigations of how humans experience and interact with sounds and music. Increasingly, this knowledge is put into industrial, societal and psychological perspectives. The age-old dream of bridging science and art has found a new and bountiful ground in the field of Sound and Music Computing.

Besides all the scientific sessions, SMAC and SMC 2013 will include many memorable events, including three concerts organized by KMH, Rencon Performance Rendering Contest, an electroacoustic pub, a Swedish summer night banquet in the archipelago of Stockholm, and above all, numerous occasions to meet friends and colleagues, old and new.

WE WELCOME YOU ALL TO SMAC – SMC 2013!

Sten Ternström, Roberto Bresin, Anders Friberg, Anders Askenfelt

Sponsors

Rector

KTH Royal Institute of Technology

School of Computer Science and Communication

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Royal College of Music

European Acoustics Association (EAA)

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Programme

ORAL SESSIONS

Tuesday July 30

RENCON Piano Competition Hall F2, KTH	14.00 - 16.30	
Welcome Reception at KMH	17.00	Registration Desk open at KMH
Concert at KMH *(see separate programme)*	18.00 - 20.00	

Wednesday July 31

Session	Lecture Hall F2	Time	Alfvén Hall F1	Session
	Registration Desk open outside F2	08.30	Registration Desk open outside F2	
	Plenary session →	09.15	Welcome & Opening	**Opening** Chair: S. Ternström
		09.45	**KEYNOTE PRESENTATION** Exploiting domain knowledge in music information research **X. Serra**	
	COFFEE	10.30	***COFFEE***	
Woodwinds – reeds and flutes Chair: Murray Campbell	**INVITED PAPER** **Wo 1** The player–wind instrument interaction **J. Wolfe**	11.00	**PERC 1** Measuring the interaction between bassoon and horn players in achieving timbre blend **S-A. Lembke, S. Levine, M. de Francisco, S. McAdams**	**Perception** Chair: tba
		11.15	**PERC 2** A social network integrated game experiment to relate tapping to speed perception and explore rhythm reproduction **G. Bellec, A. Friberg, D. Wolff, A. Elowsson, T. Weyde**	
	Wo 2 Simulations of modal active control applied to the self-sustained oscillations of the clarinet **T. Meurisse, A. Mamou-Mani, R. Caussé, D. Sharp**	11.30	**PERC 3** Methods for real time harmonic excitation of acoustic signals **S. Enderby, Z. Baracskai, C. Athwal**	
	Wo 3 An attempt at predicting the variation in playing frequencies for clarinets **W. Coyle, J. Kergomard, P. Guillemain, C. Vergez, A. Guilloteau**	11.45	**PERC 4** Sensitivity to loudspeaker permutations during an eight-channel array reproduction of piano notes **F. Fontana, Y.i De Pra, A. Amendola**	
	Poster craze A	12.00	Poster craze 1	
	LUNCH	12.15	***LUNCH***	

Session	Track A	Time	Track B	Session
Woodwinds – reeds and flutes Chair: Murray Campbell	**Wo 4** Graph-based models for woodwinds **G. Le Vey**	13.30	**HMI 2** 3D gestural interaction with harmonic pitch space **T. Hedges, A. McPherson**	**Human – Machine Interaction 1** Chair: tba
	Wo 5 Sound characteristics of a new bassoon as compared to modern German bassoon **T. Grothe, P. Wolf**	13.45	**HMI 3** Audio-tactile feedback in musical gesture primitives: finger pressing **H. Järveläinen, S. Papetti, S. Schiesser, T. Grosshauser**	
	Wo 6 An experimental study of temperature variations inside a clarinet **D. Noreland**	14.00	**HMI 4** VocaRefiner: An interactive singing recording system with integration of multiple singing recordings **T. Nakano, M. Goto**	
	Wo 7 Vocal tract effects on the timbre of the saxophone **W. Li, J-M. Chen, J. Smith, J. Wolfe**	14.15	**INVITED PAPER** **HMI 1** Child/machine interaction in reflexive environment: The MIROR platform **A. R. Addessi**	
	Wo 8 “In vivo” and “in vitro” characterization of single cane reeds **A. Munoz, B. Gazengel, J-P. Dalmont**	14.30		
	Poster craze B	14.45	Poster craze 2	
	COFFEE	15.00	***COFFEE***	
Woodwinds – reeds and flutes Chair: Miko Hirschberg	**Wo 9** Study of the perceived quality of saxophone reeds by a panel of musicians **J-F. Petiot, P. Kersaudy, G. Scavone, S. McAdams**	15.30	**PERF 2** Human-computer music performance: from synchronized accompaniment to musical partner **R. Dannenberg, N. Gold, A. Robertson, Z. Jin, O-E. Sandu, P. Palliyaguru, A. Stark, R. Kleinberger**	**Performance 1** Chair: tba
	Wo 10 Influence of pad "resonators" on saxophone admittance **P. Eveno, M. Curtit, J-P. Dalmont, R. Caussé**	15.45	**PERF 3** Conducting a virtual ensemble with a kinect device **A. Rosa-Pujazón, I. Barbancho, L. Tardón, A.M. Barbancho**	
	Wo 11 Determination of 2D quasi incompressible flow around a recorder labium **R. Auvray, P-Y. Lagrée, B. Fabre**	16.00	**PERF 4** Virtual conductor for string quartet practice **R. Baez, A.M. Barbancho, A. Rosa-Pujazon, I. Barbancho, L.J. Tardon**	
	Wo 12 Is the jet-drive flute model able to produce modulated sounds like Flautas de Chinos ? **S.Terrien, C. Vergez, P. de La Cuadra, B. Fabre**	16.15	**PERF 5** Acoustic score following to musical performance with errors and arbitrary repeats and skips for automatic accompaniment **T. Nakamura, E. Nakamura, S. Sagayama**	
	Wo 13 The design of a chromatic quena: how can linear acoustic help? **C. Vauthrin, B. Fabre, P. de La Cuadra**	16.30	**PERF 6** A contour-based jazz walking bass generator **R. Dias, C. Guedes**	
	Wo 14 Numerical modeling of a recorder in three dimensions **N. Giordano**	16.45	**PERF 7** LANdini: A networking utility for wireless LAN-based laptop ensembles **J. Narveson, D. Trueman**	
	CONCERT KMH	18.00 – 20.00	CONCERT KMH	
	ELECTROACOUSTIC PUB KTH	21.00 – 23.00	ELECTROACOUSTIC PUB KTH	

Thursday August 1

Session	Lecture Hall F2	Time	Alfvén Hall F1	Session
Singing Chair: Johan Sundberg	**INVITED PAPER** **Si 1** Supranormal voices in singing **K-I. Sakakibara**	09.00	**HMI 5** Multi-scale design of interactive music systems: The libTuiles experiment **D. Janin, F. Berthaut, M. Desainte-Catherine**	**Human – Machine Interaction 2** Chair: tba
		09.15	**HMI 6** Real-time notation using brainwave control **J. Eaton, E. Miranda**	
	Si 2 Assessment of the acoustical impact of piriform sinuses in MRI based vocal tract replicas **B. Delvaux, D. Howard**	09.30	**HMI 7** Composing for cars **A. Parkinson, A.Tanaka**	
	Si 3 Interference-free observation of temporal and spectral features in "shout" singing voices and their perceptual roles **H. Kawahara, M. Morise, K-I. Sakakibara**	09.45	**HMI 8** Downy Oak: Rendering eco-physiological processes in plants audible **M. Maeder, R. Zweifel**	
	Si 4 Power control for the second harmonic at high pitches in soprano singing: A case study **H. Takemoto, S. Adachi, T. Saitou, K. Honda, E. Haneishi, H. Kishomoto**	10.00	**INVITED PAPER** **SID 1** Sonification and auditory displays in electronic devices **B. Walker**	**Sonic Interaction Design** Chair: tba
	Si 5 & 6 Formant frequencies of sung vowels intonated by six traditional Japanese Shigin singers. Part I & II. **M. Nakayama, K. Kato, M. Matsunaga**	10.15		
	COFFEE	10.30	***COFFEE***	
	Si 7 Experimental study of the frequency leap interval produced by the change of laryngeal vibratory mechanism during sustained notes **S. Lamesch, B. Doval, M. Castellengo**	11.00	**SID 2** Controlling a sound synthesizer using timbral attributes **A. Pošćić, G. Kreković**	
	Si 8 Acoustic characteristics of vibrato in different singing styles **N. Amir, I. Ronen, O. Amir**	11.15	**SID 3** A quantitative review of mappings in musical iOS applications **T. Kell, M. Wanderley**	
	Si 9 Diverse resonance tuning strategies for women singers **J. Smith, J. Wolfe, N. Henrich, M. Garnier**	11.30	**SID 4** Acoustics-like dynamics in signal-based synthesis through parameter mapping **B. Gaffney, T. Smyth**	
	Si 10 Glitch free FM vocal synthesis **C. Chafe**	11.45	**SID 5** Real, foley or synthetic? An evaluation of everyday walking sounds **A. de Götzen, E. Sikström, F. Grani, S. Serafin**	
	LUNCH	12:00	Poster craze 3	
		12:15	***LUNCH***	

Session	Paper	Time	Paper	Session
	Si 11 Testing a new protocol to measure tuning response behaviour in solo voice ensemble singing **H. Daffern, J. Brereton**	**13.30**	**SP 1** Spectral distortion using second-order allpass filters **G. Surges, T. Smyth**	**Sound Processing 1** Chair: tba
	Si 12 Temporal coordination in vocal duet performances of musical rounds **C. Palmer, F. Spidle, E. Koopmans, P. Schubert**	**13.45**	**SP 2** Multichannel control of spatial extent through sinusoidal partial modulation **A. Cabrera, G. Kendall**	
	Si 13 Parametrization of Byzantine chant ethos through acoustic analysis:from theory to praxis **A. Georgaki, A. Chaldaikis, T. Tzevelekos**	**14.00**	**SP 3** Real time digital audio processing using Arduino **A.J. Bianchi, M. Queiroz**	
Brass Instruments Chair: Benoit Fabre	**Br 1** The playing frequency of the trombone and the impedances of the upstream and downstream ducts **H. Boutin, N. Fletcher, J. Smith, J. Wolfe**	**14.15**	**MIR 2** Semi-automatic melody extraction using note position and pitch information from users **A. Laaksonen**	**MIR 1** Chair: tba
	Br 2 Trombone sound simulation under varying upstream coupling conditions **V. Fréour, G.P. Scavone**	**14.30**	**MIR 3** Joint F0 and inharmonicity estimation using second order optimization **H. Hahn, A. Röbel**	
	Br 3 Time domain simulation of standing waves in brass wind instruments taking non-linear wave steepening into account **W. Kausel, C.B. Geyer**	**14.45**	Poster craze 4	
	COFFEE	**15.00**	***COFFEE***	
	Br 4 Control of an artificial mouth playing a trombone **N. Lopes, T. Hélie, R. Caussé**	**15.30**	**PERF 8** Motion recurrence analysis in music performances **E. Teixeira, H. Yehia, M. Loureiro, M. Wanderley**	**Performance 2** Chair: tba
	Br 5 Muscle activity in playing trumpet **S. Matsukata, H.Terasawa, M. Matsubara, T. Kitahara**	**15.45**	**PERF 9** Observed differences in rhythm between performances of classical and jazz violin students **E. Guaus, O. Saña, Q. Llimona**	
	Br 6 Timpani-horn interactions at the player's lips **J-M. Chen, J.Smith, J. Wolfe**	**16.00**	**PERF 10** Tremolo technique on the acoustic guitar: Experimental setup and preliminary results on regularity **S. Freire, L. Nézio**	
	Br 7 Pitch bending techniques on early horns by manipulation of the embouchure **L. Norman, J. Kemp, J. Chick, M. Campbell**	**16.15**	**SP 4** Audio interpolation and morphing via structured-sparse linear regression **C. Kereliuk, P. Depalle, P. Pasquier**	**Sound Processing 2** Chair: tba
		16.30	**SP 5** Warped Frames: Dispersive vs. non-dispersive sampling **G. Evangelista**	
		16.45	**SP 6** Improved polynomial transition regions algorithm for alias-suppres-sed signal synthesis **D. Ambrits, B. Bank**	

RECEPTION STOCKHOLM CITY HALL	**18.00 – 20.00**	RECEPTION STOCKHOLM CITY HALL
CONCERT KMH	**21.00 – 23.00**	CONCERT KMH

Friday August 2

Session	Lecture Hall F2	Time	Alfvén Hall F1	Session
Knut Guettler Sessions Violin acoustics, making and evaluation Chair: Anders Askenfelt	Knut Guettler in memoriam **A. Askenfelt**	09.00	**SP 7** Towards a discrete electronic transmission line as a musical harmonic oscillator **K. Buys, R. Auvray**	**Sound Processing 3** Chair: tba
	INVITED PAPER **Vi 1** Playability of bowed-string instruments **J. Woodhouse**	09.15	**SP 8** Solving interactions between nonlinear resonators **J. Bensoam, D. Roze**	
		09.30	**SP 9** An energy conserving finite difference scheme for simulation of collisions **V. Chatziioannou, M. van Walstijn**	
	INVITED PAPER **Vi 2** On the effective material properties of violin plates **E. Davis**	09.45	**SP 10** On finite difference schemes for the 3D wave equation using non-cartesian grids **B. Hamilton, S. Bilbao**	
		10.00	**HMI 9** The influence of graphical user interface design on critical listening skills **J. Mycroft**	**HMI 3** Chair: tba
	Vi 5 Enhanced simulation of the bowed cello string **H. Mansour, J. Woodhouse, G. Scavone**	10.15	**HMI 10** Discrete isomorphic completeness and a unified isomorphic layout format **B. Park, D. Gerhard**	
	COFFEE	10.30	***COFFEE***	
	Vi 6 Vibrational modes of the violin family **C. Gough**	11.00	**MIR 4** Large data sets & recommender systems: Feasible approach to learning music **J. Gabriel**	**Music Information Retrieval 2** Chair: tba
	Vi 7 Digital modeling of bridge driving-point admittances from measurements on violin-family instruments **E. Maestre, G. Scavone, J.O. Smith**	11.15	**MIR 5** Comparing timbre-based features for musical genre classification **M. Hartmann, P. Saari, P.Toiviainen, O. Lartillot**	
	Vi 8 Effect of holding the violin and soundpost removal on violin radiation via the dynamic filter model **G. Bissinger**	11.30	**MIR 6** Similarity search of freesound environmental sound based on their enhanced multiscale fractal dimension **M. Sunouchi, Y.Tanaka**	
	Vi 9 Acoustic characterisation of violin family signature modes by internal cavity measurements **C. Gough**	11.45	**MIR 7** Using semantic layer projection for enhancing music mood prediction with audio features **P. Saari, T. Eerola, G. Fazekas, M. Sandler**	
LUNCH		12.00	***LUNCH***	

	Plenary session →	13.15	**KEYNOTE PRESENTATION** Music as the goal of training and means of rehabilitation: Evidence from brain science **M. Tervaniemi**	
Knut Guettler Sessions **Violin acoustics, making and evaluation** Chair: Matthias Demoucron	**INVITED PAPER** **Vi 3** Acoustic measurements in the workshop **G. Stoppani**	14.00	**PERF 11** A preliminary computational model of immanent accent salience in tonal music **R. Parncutt, E. Bisesi, A. Friberg**	**Performance 3** Chair: tba
		14.15	**PERF 12** Expressive production of piano timbre: Touch and playing techniques for timbre control in piano performance **M. Bernays, C.Traube**	
	Vi 10 The influence of plate arching and thickness on the second and fifth mode on violin tops **M. Tinnsten, P. Carlsson**	14.30	**PERF 13** Composing social interactions for an interactive-spatial performance system **A. Parkinson, K.Tahiroğlu**	
	Vi 11 The influence of different driving conditions on the frequency response of bowed-string instruments **A. Zhang, J. Woodhouse**	14.45	**PERF 14** How do people assess computer generated expressive music performances? **S. Canazza, G. De Poli, A. Rodà**	
	Vi 12 Analysis of bridge mobility of violins **B. Elie, F. Gauthier, B. David**	15.00	***COFFEE***	
	COFFEE	15.15		
		15.30	**MIR 8** Beat-station: A real-time rhythm annotation software **M. Miron, F. Gouyon, M.E.P. Davies, A. Holzapfel**	**MIR 3** Chair: tba
	INVITED PAPER **Vi 4** On perceptual evaluation of instruments: The case of the violin put into perspective **C. Fritz**	15.45	**MIR 9** Modelling perception of speed in music audio **A. Elowsson, A. Friberg**	
		16.00	**INVITED PAPER** **MIR 1** PHENICX: Performances as highly enriched and interactive concert experiences **E. Gomez, M. Grachten, A. Hanjalic, J. Janer, S. Jordà, C. Julià, C. Liem, A. Martorell, M. Schedl, G. Widmer**	**SMC Summer School** Chair: Roberto Bresin
	Vi 13 Evaluating violin quality: A comparison of player reliability in constrained vs unconstrained tasks. **C. Saitis, G. P. Scavone, C. Fritz, B. Giordano**	16.15		
	Vi 14 Violin quality evaluation: Examining the role of auditory, vibrotactile feedbacks **I. Wollman, C. Fritz, J. Poitevineau, S. McAdams**	16.30	**INVITED PAPER** **PERF 1** Modelling emotional effects of music: Key areas of improvement **T. Eerola**	
	Vi 15 Acoustical constraints and individual preference in the coordination of complex bowing patterns **E. Schoonderwaldt, M. Demoucron, E. Altenmüller**	16.45		
	BANQUET Waxholm Castle	18.00 - 23.00	BANQUET Waxholm Castle	

Saturday August 3

Session	Lecture Hall F2	Time	SMC Summer School
	Sleep in	9.00	
Physics-based Modeling Chair: Vesa Välimäki	**INVITED PAPER** **PM 1** Acoustics of pianos: Physical modeling, simulations and experiments **A. Chaigne**	9.30	
	PM 2 Large scale physical modeling sound synthesis **S. Bilbao, B. Hamilton, A.Torin, C. Webb, P. Graham, A. Gray, K. Kavoussanakis, J. Perry**	10.00	
	PM 3 Coupled modes and time-domain simulations of a twelve-string guitar with a movable bridge **M. Marques, J. Antunes, V. Debut**	10.15	
	COFFEE	10.30	
	PM 4 Modeling a vibrating string terminated against a bridge with arbitrary geometry **D. Kartofelev, A.Stulov, H-M. Lehtonen, V. Välimäki**	11.00	
	PM 5 Distributed piano soundboard modeling with common-pole parallel filters **S. Zambon**	11.15	
	PM 6 Simulated effects of combined control applied to an experimentally identified soundboard **S. Benacchio, B. Chomette, A. Mamou-Mani, F. Ollivier**	11.30	
	PM 7 Sound synthesis of gongs obtained from nonlinear thin plates vibrations **M.Ducceschi, C.Touzé, S. Bilbao**	11.45	
	LUNCH	12.00	
Percussions Chair: Gary Scavone	**Pe 1** Nonlinear vibrations of steelpans: Analysis of mode coupling in view of modal sound synthesis **M. Monteil, C.Touzé, O.Thomas**	13.15	
	Pe 2 Time-resolved interferometry and phase vocoder analysis of a Caribbean steelpan **A. Morrison, D. Zietlow, T.Moore**	13.30	
	Pe 3 The role of damping in steel pan manufacture. **C. Barlow, S. Maloney, J. Woodhouse**	13.45	
	Pe 4 Objective approach for assessing the tuning properties of historical carillons **V. Debut, M. Carvalho, J. Antunes**	14.00	
	Pe 5 Experimental study of coupled drumhead vibrations using electronic speckle-pattern interferometry **R. Worland**	14.15	
	Pe 6 Numerical experiments with non-linear double membrane drums **A. Torin, S. Bilbao**	14.30	
Adjourn		14.45	

POSTERS

All poster rooms will be open daily during session hours (09.00 - 17.00)

Introductions to the posters (1 min) are given during Poster Crazes

SMAC Hall F2

Poster Crase A
Wednesday 31 July
12:00 – 12:15

Woodwinds – reeds and flutes

Wo 1P Comparison of two methods of sound power measurements of flue organ pipes
J. Angster, K. Hoge, A. Miklos

Wo 2P Prediction of the dynamic oscillation threshold of a clarinet model
B. Bergeot, A. Almeida, C. Vergez, B. Gazengel

Wo 3P On reeds and resonators: Possible explanations for cyclic spectral envelopes in the case of double reed instruments
S. Carral, C. Reuter

Wo 4P External pipe resonators and harmonica acoustics
J. Cottingham, C. Brock

Wo 5P Investigation of bassoon directivity
T. Grothe, M. Kob

Wo 6P Evaluating a wavlet-based analysis of sensor reed signals for performance research
A. Hofmann, W. Goebl, M. Weilguni

Wo 7P The voice of the mechanical dragon
M. Hirschberg, O. Rudenko, G. Nakiboglu, A. Holten, J. Willems, A. Hirschberg

Wo 8P Embracing the digital in instrument making: Towards a musician-tailored mouthpiece by 3D printing
V. Lorenzoni, Z. Doubrovski, J. Verlinden

SMC Hall F1

Poster Crase 1
Wednesday 31 July
12:00 – 12:15

Perception

PERC 1P Reinforcement learning models for acquiring emotional musical modes
T. Tanaka, H. Ohmura, K.Furukawa

PERC 2P About the impact of audio quality on overall listening experience
M. Schoeffler, J. Herre

PERC 3P Effect of timbre on melody recognition in three-voice counterpoint music
S. Chon, K. Schwartzbach, B. Smith, S. McAdams

PERC 4P The importance of amplitude envelope: Surveying the temporal structure of sounds in perceptual research
J. Gillard, M. Schutz

PERC 5P Modeling of melodic rhythm based on entropy toward creating expectation and emotion
H. Ohmura, T. Shibayama, S. Shibuya, T. Takahashi, K. Okanoya, K. Furukawa

PERC 6P Design of an interactive earphone simulator and results from a perceptual experiment
PM. Lindborg, M. Lim

PERC 7P How predictable do we like our music? Eliciting aesthetic preferences with the melody triangle mobile app
H. Ekeus, S. Abdallah, P. W. Mcowan, M. Plumbley

PERC 8P A multipitch estimation algorithm based on fundamental frequencies and prime harmonics
A. Camacho, I. Kaver-Oreamuno

Wo 9P A digital bagpipe chanter system to assist in one-to-one piping tuition
D. Menzies, A. McPherson

Wo 10P Numerical analysis of the mean flow effect on the sound directivity pattern of cylindrical ducts
Y. Shi, A. Da Silva, G. Scavone

Singing

Si 1P Evaluation of pitch detection algorithms: Case of monophonic vocal performance
R. Budrys, R. Ambrazevičius

Si 2P Formant tuning in Byzantine chant
G. Chrysochoidis, G. Kouroupetroglou, D. Delviniotis, S. Theodoridis

Si 3P Vibrato analysis in Byzantine music
D. Delviniotis, G. Kouroupetroglou, S. Theodoridis

Si 4P A pilot study of vibration pattern measurement for facial surface during singing by using scanning vibrometer
T. Kitamura, H. Hatano, T. Saitou, Y. Shimokura, E. Haneishi

Si 5P Postural sway in vocal duets
E. Koopmans, C. Palmer, F. Spidles

Si 6P A study of speaker dependent formant space variations in karaoke singing
M. Mehrabani, J. Hansen

Si 7P An attempt to develop a singing synthesizer by collaborative creation
M. Morise

Si 8P A method of division of soprano ranges and confirmation of their voice transformation point based on harmonics analysis
G. Qu

Si 9P Generating singing voice expression contours based on unit selection
M. Umbert, J. Bonada, M. Blaauw

Music Information Retrieval

MIR 1P Global key extraction from classical music audio recordings based on the final chord
C. Weiss

MIR 2P PEVI: Interface for retrieving and analyzing expressive musical performances with scape plots
S. Miki, T. Baba, H. Katayose

MIR 3P Segmentation and timbre similarity in electronic dance music
B. Rocha, N. Bogaards, A. Honingh

MIR 4P Melodic outline extraction method for non-note-level melody editing
Y. Tsuchiya, T. Kitahara

MIR 5P SoundAnchoring: Content-based exploration of music collections with anchored self-organized maps
L. Collares, T.Tavares, J. Feliciano, S. Gao, G. Tzanetakis, A. Gooch

MIR 6P SmartDJ, An interactive music player for music discovery by similarity comparison
M. Aw, C. Lim, PM. Lindborg

MIR 7P Sound analysis based on phase information that connects time and frequency
P. Pabon, J. van Velthoven

Poster Crase 2
Wednesday 31 July
14:45 – 15:00

Human – Machine Interaction

HMI 1P Amarok Pikap: Interactive percussion playing automobile
S. Artut

HMI 2P Full automation of real-time processes in interactive compositions:Two related examples
J. Garavaglia

HMI 3P Mocap Toolbox - A MATLAB toolbox for computational analysis of movement data
B. Burger, P. Toiviainen

HMI 4P Relationships between spectral flux, perceived rhythmic strength and the propensity to move
B. Burger, R. Ahokas, A. Keipi, P. Toiviainen

HMI 5P Programming interactive music scores with INScore
D. Fober, S. Letz, Y. Orlarey, F. Bevilacqua

HMI 6P Real-time event sequencing without a visual interface
T. Tavares, A. Monteiro, J. Barbedo, R. Attux, J. Manzolli

HMI 7P Melody Bounce: Mobile rhythmic interaction for children
S. Baldan, S. Serafin, A. De Götzen

Poster Crase B
Wednesday 31 July
14:45 – 15:00

Violin acoustics, making and evaluation

Vi 1P The acoustics of the Hardanger fiddle
A. Buen

Vi 2P Acoustical features of complex bowing patterns in violin performance
M.Demoucron, E. Schoonderwaldt, M. Leman, E. Altenmüller

Vi 3P String 'after length' and the cello tailpiece: Acoustics and perception
E. Fouilhé, A. Houssay

Vi 4P Sensor based fatigue recognition in violin playing
T. Großhauser

Vi 5P New devices for testing the stiffness characteristics of free violin plates
R. Jia, A. Zhang, L. Fu

Vi 6P Conceptualizing the "good" violin in preference descriptions by experienced players
C. Saitis, C. Fritz, C. Guastavino, G. Scavone

Vi 7P Probabilistic modeling of bowing gestures for gesture-based violin sound synthesis
A. Thippur, H. Kjellström, A. Askenfelt

Brass Instruments

Br 1P The use of the input impedance for characterising historical serpents
P. Eveno, S. Le Conte

Br 2P Sensor based hand weight and pressure measurements in trombone playing
T. Großhauser

Physics-based Modeling

PM 1P A new method for the identification of the original modes of damped three-dimensional axi-symmetric structures subjected to constraining boundary conditions
V. Debut, M. Carvalho, J. Antunes

PM 2P A structured approach to using a rectangular brace to design a soundboard section for a desired natural frequency
P. Dumond, N. Baddour

PM 3P Computing virtual acoustics using the 3D finite difference time domain and Kepler architecture GPUs method
C. Webb

Human – Machine Interaction

HMI 8P Plucking Buttons: An alternate soft button input method on touch screens for musical interaction
E. Lee, W. Yeo

HMI 9P Robin: An algorithmic composer for interactive scenarios
F. Morreale, R. Masu, A. De Angeli

HMI 10P x-OSC: A versatile wireless I/O device for creative/music applications
S. Madgwick, T. Mitchell

HMI 11P The Airsticks: A new interface for electronic percussionists
A.Ilsar, M. Havryliv, A. Johnston

HMI 12P A computational method for exploring musical creativity development
A. Alexakis, A. Khatchatourov, A. Triantafyllaki, C. Anagnostopoulou

HMI 13P The actuated guitar: A platform enabling alternative interaction methods
J. Larsen, D. Overholt, T. Moeslund

HMI 14P Komeda: Framework for interactive algorithmic music on embedded systems
D. Jackowski, K. Baclawski

HMI 15P Sound Hunter – Developing a navigational HRTF-based audio game for people with visual impairments
S. Brieger

HMI 16P Energy harvesting power flower bell – A cybernetic sound installation driven by a dirt-battery
J. Schauer, W. Ritsch, L. Fickert

HMI 17P Artificial affective listening towards a machine learning tool for sound-based emotion therapy and control
A. Kirke, E. Miranda,S. Nasuto

Structured Poster Session
Hall F3
Thursday 1 Aug
12:15 – 13:15

Music Acoustics Education
Structured Poster Session
Chair: *Joe Wolfe*

MAE 1P Music acoustics education at the Erich Thienhaus Institut in Detmold
M. Kob

MAE 2P The musical acoustics research library (MARL): Fully digital & online
G. Scavone, J. McBride

MAE 3P Activities for a course of physics of bowed instruments
J. Torres

MAE 4P Teaching physics via the Web using music acoustics
J. Wolfe, G. Hatsidimitris, J. Smith, J.Tann

Structured Poster Session
Hall F3
Friday 2 Aug
12:15 – 13:15

Guitars and plucked instruments
Structured Poster Session
Chair: *Jean-Loïc Le Carrou*

Gi 1P Analysis of the harpsichord plectrum-string interaction
D. Chadefaux, J-L. Le Carrou, S. Le Conte, M. Castellengo

Gi 2P Bio-inspired robot to study stringed instruments: application to the harp
D. Chadefaux, A. Roy, J-L. Le Carrou, M-A. Vitrani, B. Fabre

Gi 3P Acoustic radiation of a carbon composite soudboard guitar compared to wooden classical guitars
S. Le Moyne, F. Ollivier, J. Frelat, C. Besnainou

Gi 4P Vibroacoustic behavior of a vihuela
S. Le Moyne, F. Ollivier, S. Le Conte

Gi 5P Eigenspectrum of the sarode membrane
M. Mishra, G. Suresh, R. Adhikari

Gi 6P Ebony vs. rosewood: Experimental investigation about the influence of the fingerboard on the sound of a solid body electric guitar
A. Paté, J-L. Le Carrou, B. Fabre

Poster Crase 3
Thursday 1 Aug
12:00 – 12:15

Music Performance

PERF 1P Towards computable procedures for deriving tree structures in music: Context dependency in GTTM and Schenkerian theory
A. Marsden, K. Hirata, S. Tojo

PERF 2P Situating the performer and the instrument in a rich social context with PESI extended system
C. Goddard, K. Tahiroğlu

PERF 3P
Refined spectral template models for score following
F. Korzeniowski, G. Widmer

PERF 4P A history of sequencers: Interfaces for organizing pattern-based music
R. Arar, A. Kapur

PERF 5P Tale following: Real-time speech recognition applied to live performance
J-L. Rouas, B. Mansencal, J. Larralde

PERF 6P Technical report on a short live-action film whose story with soundtrack is selected in real-time based on audience arousal during performance
A. Kirke, D. Williams, E. Miranda, A. Bluglass, C. Whyte, R. Pruthi, A. Eccleston

PERF 7P Improving the real-time performance of a causal audio drum transcription system
M. Miron, M. Davies, F. Gouyon

PERF 8P Creating expressive piano performance using a low-dimensional performance model
Y. Gu, C. Raphael

PERF 9P Skalldans, an audiovisual improvisation framework
PM. Lindborg

PERF 10P Network music with Medusa: A comparison of tempo alignment in existing MIDI APIs
F. Schiavoni, M. Queiroz, M. Wanderley

PERF 11P mono2eN: A multi-channel autospatialisation performance system
C. Goddard

Gi 7P High-speed camera displacement measurement (HCDM) technique of string vibrations
N. Plath

Gi 8P Finite element model of a kantele with improved sound radiation
H.Tahvanainenn, J. Pölkki, H. Penttinen, V. Välimäki

Gi 9P The Bolivian charango – An acoustic study
O. Woods, J. Woodhouse

Music Performance

PERF 12P Modeling and simulation: The Spectral CANON for CONLON Nancarrow by James Tenney
C. de Paiva Santana, J. Bresson, M. Andreatta

PERF 13P Capacitive left hand finger and bow sensors for synchronization and rhythmical regularity analysis in string ensembles
T Großhauser, S. Feese, G. Tröster

PERF 14P Acoustic retroreflectors for music performance monitoring
H. Tuominen, J. Rämö, V. Välimäki

PERF 15P Mixing symbolic and audio data in computer assisted music analysis: A case study from J. Harvey's Speakings (2008) for orchestra and live electronics
S. Schaub, I. Simurra, T. Tavares

PERF 16P Brazilian challenges on network music
J. Arango, M. Tomiyoshi, F. Iazzetta, M. Queiroz

Poster Crase 4
Thursday 1 Aug
14:45 – 15:00

Sound Processing

SP 1P Automatic tuning of the OP-1 synthesizer using a multi-objective genetic algorithm
M. Macret, P. Pasquier

SP 2P An open-source framework for time-domain simulations
C. Geyer, W. Kausel

SP 3P Auralization of coupled spaces based on a diffusion equation model
P. Luizard, J-D. Polack, B. Katz

SP 4P Warped low-order modeling of musical tones
R. Mignot, H-M. Lehtonen, V. Välimäki

SP 5P Four-part harmonization using probabilistic models: comparison of models with and without chord nodes
S. Suzuki, T. Kitahara

SP 6P A versatile toolkit for controlling dynamic stochastic synthesis
G. Kreković, D. Petrinović

SP 7P Visions of sound: The Centro di Sonologia Computazionale, from computer music to sound and music computing
S. Canazza, G. De Poli, A. Vidolin

SP 8P Smoothness under parameter changes: Derivatives and total variation
R. Holopainen

SP 9P Audio restoration of solo guitar excerpts using a excitation-filter instrument model
J. Parras-Moral, F. Canadas, P. Candeas, N. Reyes

SP 10P Spatium, tools for sound spatialization **R. Penha, J. Oliveira**	
SP 11P Dynamic FM synthesis using a network of complex resonator filters **J. Parker, T. Bovermann**	
SP 12P Reconfigurable autonomous novel guitar effects (RANGE) **D. MacConnell, S. Trail, G.Tzanetakis, P. Driessen, W. Page**	

SID 1P Urb: Urban sound analysis and storage project **J. Gomes, D. Tudela**	Sonic Interaction Design
SID 2P Non-realtime sonification of motiongrams **A. Refsum Jensenius**	
SID 3P Impulse response estimation for the auralisation of vehicle engine sounds using dual channel FFT analysis **S. Shelley, D. Murphy, S. Goodwin**	
SID 4P Image sonification using local keypoint features **K. Kim, W. Yeo**	
SID 5P Real-time hallucination sonification and simulation through user-led development of an iPad augmented reality system **A. Kirke, J. Eaton, E. Miranda**	

Abstracts

Keynotes

Exploiting domain knowledge in music information research

Xavier Serra

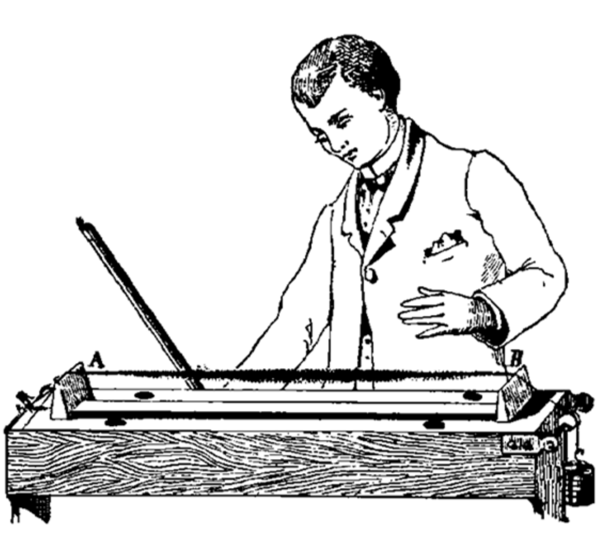
Music Information Research (MIR) is a discipline that aims to understand and model music from an information processing perspective, but the successful approaches used in MIR are going beyond the traditional data processing methodologies. Most of the great advancements have been the result of combining engineering disciplines such as audio signal processing and machine learning with non-engineering disciplines such as music perception and music theory. One of the challenges in MIR is to automatically describe music audio signals, thus to develop methodologies to extract musically useful information from audio recordings. In this paper we claim that if we want to advance in this direction we should maximize the use of musical knowledge in all the steps of our research tasks. To support this claim we overview some of the work being carried out in CompMusic, a project that aims to analyze and automatically describe the music of several non-western music traditions.

Music as the goal of training and means of rehabilitation: Evidence from brain science

Mari Tervaniemi

During the past three decades, our knowledge about brain functions and its structures underlying music perception, performance, and emotions has accumulated relatively quickly. Cortical and subcortical brain areas involved in these musical functions have been identified using various techniques and paradigms. In the present talk, I will introduce recent findings revealing enhanced brain mechanisms during long-term musical training, as well as by informal music activities at home. Furthermore, I will present examples of how casual music activities, such as music listening and singing, can be used in neurological rehabilitation to promote health and well-being in patients and their family members. In sum, these findings promote the use of music in formal and informal settings across the whole life span in healthy participants, as well as with individuals with special needs.

SMAC 2013



In memory of Knut Guettler (1943 – 2013)

Chairs: *Anders Askenfelt* & *Matthias Demoucron*

INVITED Vi 1

Playability of bowed-string instruments

Jim Woodhouse

Theory and measurements relating to the motion of a bowed string are surveyed critically, to see where they may shed light on judgments by players about the relative “ease of playing” of different violins. Particular attention is paid to what a player can do to achieve a range of tone colour, and how that perception of tonal range might vary between instruments. Some current lines of research are reviewed, and suggestions made for future physical and psychophysical investigation.

INVITED Vi 2

On the effective material properties of violin plates

Evan Davis

The graceful arch and thickness variations of the violin family plates are replaced by a rectangular block of an effective material. The effective material properties are estimated from the violin’s free plate modal frequencies. The dynamics of a rectangular block of the effective material will match the dynamics of the violin plate in the low frequencies. The violin free plate mode shapes have the characteristic ring and X shapes of a dynamically-square plate. Dynamically square plates balance the bending stiffness in two directions with the plate dimensions. The Poisson’s ratio of the effective material is determined by the ratio of the frequency of the ring mode to the X mode. The observed tuning ratio of 2.3 (M5:M2) yields a Poisson’s ratio of one. Violin free plate data from old masters and modern makers cluster near the 2.3 tuning ratio. A Poisson’s ratio of one implies that the free plate has dynamic properties similar to a spherical cap. A spherical cap model can be used to gain insights into roles of arch height and plate thickness in plate tuning.

INVITED Vi 3

Acoustic measurements in the workshop

George Stoppani

Violin makers in the past have been remarkably successful with minimal knowledge of sound and vibration. We know some things about their woodworking methods but nothing about how they thought about the relationship between the structure and the sound it produced. The trade has tended to insulate itself from science, preferring to refine the empirical and intuitive methods and build on the success or failure in their own work and of previous generations. The intention of incorporating acoustic knowledge and measurements into the making process is to add to the existing body of expertise rather than to replace it. Structural vibration and sound radiation measurements can be helpful in making decisions about the initial design, during the making process and after the instrument is completed. Measurements can also be useful for diagnosing problems with older instruments that are performing below expectation and even for a decision whether or not to buy an instrument. Acoustic measurements are often disappointing when the intention is to explore the effect of usual workshop adjustments such as soundpost moves or bridge trimming. However, for a more global view of where a particular instrument lies on a map of mass, stiffness and frequency characteristics or to examine the relationship between substructures such as neck/fingerboard or tailpiece modal analysis provides insight that cannot be obtained by traditional methods.

INVITED Vi 4

On perceptual evaluation of instruments: The case of the violin put into perspective

Claudia Fritz

An object of study in mechanics for more than three hundred years, the violin has only recently been scientifically studied from a perceptual point of view. The range of experiments which have been conducted since 2005, complemented by a few studies of other instruments, offers a nice illustration of possible methodologies, and serves as a basis for discussing their respective advantages, as well as their limitations and issues. While there is no general recipe (sincemethodological choices depend largely on the goals of the study), ecological validity is certainly one of the key prerequisites for a perceptual experiment seeking to provide meaningful results, in the sense that they can be generalised from laboratory settings to "real life" situations.

Vi 5

Enhanced simulation of the bowed cello string

Hossein Mansour, Jim Woodhouse and Gary P. Scavone

A detailed time-domain simulation is implemented to model the bowed cello string. Building on earlier simulation models, several new features have been added to make the model more realistic. In particular, a large number of body modes, both transverse polarizations of the string motion, the longitudinal vibrations of the bow hair and the effect of the sympathetic strings are included. These additional features can be turned on and off in the model to evaluate their relative importance. To the best of our knowledge this is the first time that the second polarization of the string and the effect of the sympathetic strings have been included in a bowed-string simulation. The compliance of the bow hair was accounted for in previous studies but without considering its own vibration properties. Different features of the model are turned on and the classic Schelleng minimum bow force is calculated for combinations of bow-bridge distance and different notes being played on the string. The main finding is that all features reduce the minimum bow force to some extent. This reduction is almost frequency independent for the case of the second polarization and the longitudinal bow hair vibration, but clearly frequency dependent for the sympathetic strings case.

Vi 6

Vibrational modes of the violin family

Colin Gough

The generic wave-mechanical properties of violin-shaped instruments are described by considering their bodies as simplified, shallow, thin-walled, guitar-shaped, shell structures with the arched plates connected around their edges by the ribs. COMSOL finite element software is used to illustrate the strong dependence of the shapes and frequencies of the low frequency *A1*, *CBR*, *B1-* and *B1+* signature modes on the rib coupling strength, the island area between the *f*-holes, coupling to the internal cavity pressure fluctuations via the Helmholtz *f*-hole resonance, and the soundpost position and strength. The model illustrates the relationship between the free pate modes and those of the fully assembled instrument. It also identifies the important *B1-* and *B1+* signature modes as normal modes involving the in- and out-of-phase combinations of a bending and breathing mode of the shell, with the breathing component responsible for both the directly and indirectly (via the *A0* mode) radiated sound. The model describes the vibrational modes over the whole playing range of the violin and can be used to predict both the admittance at the bridge and the radiated sound.

Vi 7

Digital modeling of bridge driving-point admittances from measurements on violin-family instruments

Esteban Maestre, Gary P. Scavone and Julius O. Smith

We present a methodology for digital modeling of D-dimensional driving-point bridge admittances from vibration measurements on instruments of the violin family. Our study, centered around the two-dimensional case for violin, viola, and cello, is based on using the modal framework to construct an admittance formulation providing physically meaningful and effective control over model parameters. In a first stage, mode frequencies and bandwidths are estimated in the frequency domain via solving a non-convex, constrained optimization problem. Then, mode amplitudes are estimated via semidefinite programming while enforcing passivity. We obtain accurate, low-order digital admittance models suited for real-time sound synthesis via physical models.

Vi 8

Effect of holding the violin and soundpost removal on violin radiation via the dynamic filter model

George Bissinger

The dynamic filter model, a deterministic-statistical, net-work-structural acoustics parameterization of violin radiativity, was developed to simulate violin radiativity trends due to plate and bridge tuning for a violin with a soundpost, suspended free-free. The model formalism is quite general however and this generality was tested by successful application to two decades-old experiments: 1) vibration measurements on a violin held (as in playing) vs. free-free and 2) a soundpost removal experiment with pre- and post- removal: a) modal analyses, b) boundary element method (no f-hole) radiation efficiency calculations) and c) room-averaged acoustic measurements.

Vi 9

Acoustic characterization of violin family signature modes by internal cavity measurements

Colin Gough

The sound radiated by the signature modes of stringed instruments in their lowest two octaves is shown to be directly related to the sound pressure at the acoustic centre of their hollow bodies, where the nodal lines of the longitudinal and transverse *a1* and *a2* internal dipole modes of the air cavity intersect. Pressure measurements inside the instrument itself can therefore be used to characterise the acoustic radiating properties of the signature modes of instruments of any size without contamination from resonances of the surrounding space or interference from external noise. This is illustrated by high resolution, acoustic location independent, measurements on violins, violas, cellos and double basses, which could all have been made in the luthier’s workshop without any sophisticated equipment.

Vi 10

The influence of plate arching and thickness on the second and fifth mode on violin tops

Mats Tinnsten and Peter Carlsson

The objective of the work presented is to numerically investigate the influence of different arching and thickness on the resonance frequencies and mode shapes of the top plate. Here the focus has been on the second and fifth resonance frequency, the so called X and ring mode. The use of numerical analyses makes it possible to vary only one parameter while all other are held constant. The geometrical model was generated by use of the ideas of Robert Zuger and the general 3D-modelling software Rhinoceros. For each arch–height value, a new geometric model has been generated and exported to the finite element program Abaqus. The geometric model was generated in the same systematic way in all analyses. The results show surprising linearity both when varying the thickness and the arching.

Vi 11

The influence of different driving conditions on the frequency response of bowed-string instruments

Ailin Zhang and Jim Woodhouse

A well-established measurement on the bodies of bowed-string instruments is the input admittance at the bridge. The commonest method for measuring the input admittance is hammer testing on the bridge corner. However, this method has been questioned, due especially to differences between human bowing and hammer impact. In this paper an attempt has been made to survey the influence of different driving methods on the frequency response of a bowed-string instrument, as well as confirm the reliability of the classic hammer method. A series of experiments are carried out with three different driving conditions in the case of a cello: hammer, normal bowing of a string, and step excitation by a breaking wire. The results suggest that there is nothing fundamentally different about the hammer method, compared to other kinds of excitation methods. Some possible differences between the hammer method and normal bowing are also discussed.

Vi 12

Analysis of bridge mobility of violins

Benjamin Elie, François Gautier and Bertrand David

This paper focuses on the bridge mobility of violins. The mobility, or mechanical admittance, quantifies the efficiency of the instrument body to vibrate when a force is applied to the structure. The computation of the mean mobility, after the Skudrzyk's mean-value theorem, enables a global characterization of the bridge mobility. The choices made by the luthier, when he builds, restores, or adjusts an instrument, modify the mobility and the mean mobility: this is the signature of the instrument. This study shows that the bridge mobility measurement may be helpful for luthiers to objectively characterize an instrument. Two experimental applications on the violin is presented in the paper: the first one studies the characterization of the same violin in several configurations, corresponding to different positions of the soundpost. The second application studies the effect of a violin mute on both the bridge mobility and the spectral characteristics of the produced sound. This study is a part of the PAFI project, which aims to develop a set of tools dedicated to instrument makers.

Vi 13

Evaluating violin quality: A comparison of player reliability in constrained vs. unconstrained tasks

Charalampos Saitis, Gary P. Scavone, Claudia Fritz and Bruno L. Giordano

The overall goal of the research presented here is to better understand how players evaluate different qualities of the violin. To this end, we investigated intra- and inter-individual consistency in preference judgements by experienced violinists. Results from two previous studies that involved free-playing evaluative tasks showed that players are self-consistent in their preference for violins and tend to agree of what particular qualities they look for in an instrument (in this case, “richness” and “dynamic range”). However, the perception of the same attributes widely varies across individuals, thus likely resulting in large inter-individual differences in the preference for violins. A third study was conducted to further investigate the perceptual evaluation of richness and dynamic range in constrained vs. unconstrained playing tasks. Results indicated that specifying the musical material and technique removes a significant amount of inter-individual variability: the more focused the task, the more self-consistent violinists are and the more they agree with each other.

Vi 14

Violin quality evaluation: Examining the role of auditory and vibrotactile feedbacks

Indiana Wollman, Claudia Fritz, Jacques Poitevineau and Stephen McAdams

The present paper explores the role of auditory and vibrotactile feedbacks involved in violin playing and evaluation. Twenty professional violinists took part in a perceptual experiment employing a near-blind violin evaluation task under three different playing conditions: i) under normal playing conditions, ii) with auditory masking, and iii) with vibrotactile masking. Under each condition, the violinists evaluated five violins according to criteria related to violin playing and sound characteristics, rated their overall quality and relative preference. Both auditory and tactile feedbacks appeared important in the violinists' evaluations; if on average, auditory masking had a greater effect on criteria evaluation and preference than did tactile masking, their relative importance was found to depend on the violinist, the violin and the type of evaluation (criteria or preference). The overall quality ratings were accurately predicted by the rating criteria, which also proved to be perceptually relevant to violinists, but were poorly correlated with the preference ratings, suggesting that the two types of ratings may stem from different strategic decision processes.

Vi 15

Acoustical constraints and individual preference in the coordination of complex bowing patterns

Erwin Schoonderwaldt, Matthias Demoucron and Eckart Altenmüller

A series of recent studies have shed light on coordination in complex bowing gestures involving string crossings and bow changes, both in performance and perception. However, significant individual differences in coordination behavior were found, which puts into question the strictness of acoustical constraints. In this paper the inter- and intra-individual variance will be examined more closely. It is suggested that the acoustical constraints are (within certain limits) not very strict, and leave room for the performer to manipulate subtle quality aspects of the note transitions.

Vi 1P

The acoustics of the Hardanger fiddle

Anders Buen

The Hardanger fiddle (HF) is a highly decorated, baroque-like Norwegian folk music instrument with four or five sympathetic strings. Compared to a violin, it has shorter and lighter gut, G, D and A and E steel strings, a flatter bridge and fingerboard, longer *f*-holes and the top has a flatter cross arch, mainly between and above the *f*-holes. Acoustically it is closely related to the violin. Its “ringing qualities” relate it to other bowed string instruments with free vibrating string designs, like the Swedish folk drone fiddle and the ‘nyckelharpa,’ and also to the more “distant relative” the viola d’amore. Acoustic properties and the construction of the HF are compared to the violin, showing close similarities in the lower frequency range. The bridge design and the tonal ideals are different, especially for the higher frequencies. A HF will in general be a little quieter than a violin, but may sound more “intense”. Scordatura is often used and the HF is traditionally played solo. The tuning of the A string can be any frequency between A4 - D5 (440 - 587 Hz), with B4 (494 Hz) being the most typical pitch.

Vi 2P

Acoustical features of complex bowing patterns in violin performance

Matthias Demoucron, Erwin Schoonderwaldt and Marc Leman

Music performance relies on fine motor skills and subtle judgements of the resulting sound in order to adjust body movements involved in playing. In this respect, fast repetitive bowing patterns in which notes are alternatively played on two adjacent strings offer an interesting case study of subtle coordination of bow movements. In recent experiments, it was shown that a particular timing relation between bow velocity and string crossing was preferred both in real performance and in perceptual tests, the bow change occurring consistently after the string crossing. The aim of the present study was to identify and quantify acoustical features on which violinists may base this preference. For that purpose, we performed simulations over a wide range of coordination parameters. The results of perceptual tests were examined under the light of the simulations in order to determine significant acoustical features and the acceptance limits associated with them.

Vi 3P

String ‘after length’ and the cello tailpiece: Acoustics and perception

Eric Fouilhé and Anne Houssay

In a long term study of cello tailpieces, we have first identified the vibrating modes of a cello tailpiece mounted on a dead rig, and have worked on the possible influence of the wood on theses modes. The influence of the position of the tailpiece on the modes and on the sound has also been explored, by varying the “after-length”, i.e. the distance of the tailpiece to the bridge which leaves a small length of vibrating string. A historical study on iconography and texts on this “after-length" and the changes in the history of the cello was carried out, and the few texts of the 19th century mentioning its role on the sound of the cello were studied. Here we describe the experiments on the parameters involved, and the perception of sound changes is exposed.

Vi 4P

Sensor based fatigue recognition in violin playing

Tobias Großhauser

Fatigue in daily instrumental exercising is a common problem in musical instrument learning. Musicians practice long and pause concepts are underrepresented resulting into performance shortcomings or even work-related injuries. The basic idea of the following investigation was to detect fatigue based on sensor data with sensors fixed on the musical instrument. Reliable fatigue detection would increase practicing efficiency, which results in faster progress in musical instrument learning while minimizing pain or preventing aftereffects due to too much physical stress. In this paper, we tested a sensor setup that is unobtrusively mounted to a violin and the frog of the violin bow. The sensors were 9 degree of freedom (9 DOF) acceleration sensors and gyroscopes. In two studies, amateur and professional violinists played defined music excerpts and scales at different speeds during sensor recordings. A one-hour uninterrupted playing regime with interspersed subjective evaluations of playing effort and tiredness was used. To test for changes in fatigue level, a scale and a segment of a music piece were intermittently played every 10 minutes. Sensor data were recorded and specific movement patterns recognized. Results show that the present setup can uncover features associated to individual fatigue during violin playing for amateurs, potentially helping to prevent injuries and to optimize practice regimes in these musicians.

Vi 5P

New devices for testing the stiffness characteristics of free violin plates

Ruiqing Jia, Ailin Zhang and Lei Fu

The mode tuning of violin plates has been demonstrating its importance to violin makers and researchers. As an aid to tune free plates to right resonance frequencies, modern measuring techniques such as the Chladni-pattern method and holographic interferometry show luthiers how the eigenmodes and frequencies change with adjusting the stiffness and mass of the plate intuitively. However, such patterns cannot be recorded quantitatively and used efficiently by luthiers. Thus, the art of tuning is still dominated by empirical knowledge of violinmakers. In this article, three new devices have been developed to test the bending and torsional stiffness of free plates instead of traditional manual tests. Experiments are performed on four pairs of violin plates to show that these devices could reliably produce recordable data related to stiffness characteristics. In addition to a detailed calculation of the stiffness of tested plates, potential directions for future investigation are also discussed.

Vi 6P

Conceptualizing the "good" violin in preference descriptions by experienced players

Charalampos Saitis, Claudia Fritz, Catherine Guastavino and Gary P. Scavone

This paper explores how violin quality is conceptualized in spontaneous preference descriptions by experienced performers collected in a playing-based perceptual evaluation experiment. Upon ordering a set of different violins in terms of preference, players were asked to explain their choices via an open questionnaire. The constant comparison technique from grounded theory was employed to develop a classification scheme of concepts and the attributes that embody them. A quantitative analysis, based on the number of occurrences for each attribute and concept, provided a hierarchy of violin preference criteria/quality concepts: The response of the violin to the various techniques and musical intentions in direct association with the quantity and quality of the produced sound as well as the emotions and values of the player influence how the “good” violin is conceptualized.

Vi 7P

Probabilistic modeling of bowing gestures for gesture-based violin sound synthesis

Akshaya Thippur, Anders Askenfelt and Hedvig Kjellström

We present a probabilistic approach to modeling violin bowing gestures, for the purpose of synthesizing violin sound from a musical score. The gesture models are based on Gaussian processes, a principled probabilistic framework. Models for bow velocity, bow-bridge distance and bow force during a stroke are learned from training data of recorded bowing motion. From the models of bow motion during a stroke, slightly novel bow motion can be synthesized, varying in a random manner along the main modes of variation learned from the data. Such synthesized bow strokes can be stitched together to form a continuous bowing motion, which can drive a physical violin model, producing naturalistic violin sound. Listening tests show that the sound produced from the synthetic bowing motion is perceived as very similar to sound produced from real bowing motion, recorded with motion capture. Even more importantly, the Gaussian process framework allows modeling short and long range temporal dependencies, as well as learning latent style parameters from the training data in an unsupervised manner.

Guitars & plucked instruments Poster session

Chair: *Jean-Loïc Le Carrou*

Gi 1P

Analysis of the harpsichord plectrum-string interaction

Delphine Chadefaux, Jean-Loïc Le Carrou, Sandie Le Conte and Michèle Castellengo

This paper describes a study of string plucking for the harpsichord. Its aim is to provide an experimentally-based analysis of the plectrum-string interaction and to propose some refinements of an existing model. An experimental setup has been designed using a high-speed camera combined with a laser doppler vibrometer and classical audio recordings. This provides accurate estimations of jack and plectrum motion throughout the harpsichord plucking in a realistic musical context. The set of descriptors extracted from these measurements provides typical orders of magnitude of plucking parameters required to feed and validate the investigated model. Results highlight estimations of the instrumentalist's control parameters as well as of the intrinsic plucking parameters according to the performed sequence tempo. Besides, a model of the plectrum-string interaction, which takes into account the section variation at the plectrum tip, gives results close to the experimental ones. This model will be of great interest to enquire about harpsichord plectrum harmonization.

Gi 2P

Bio-inspired robot to study stringed instruments: Application to the harp

Delphine Chadefaux, Alexandre Roy, Jean-Loic Le Carrou, Marie-Aude Vitrani and Benoit Fabre

This paper is a review of a previous musician/harp interaction study leading to the specifications of a configurable excitatory mechanism. Such robotic tools can be valuable to investigate the musician/instrument interaction and the mechanical behavior of an instrument in a repeatable and realistic musical context. To design a robotised excitatory mechanism for the concert harp, the mechanical descriptors' typical orders of magnitude defining a musical performance have to be highlighted. For this purpose, two experimental setups have been designed. The first one focuses on the sound-producing gesture (i.e. on the plucking action). A high-speed camera has been used to accurately measure the finger and string motions in a realistic musical context. The second measurement set-up consists in capturing the whole harpist's body motion through infrared cameras during a performance. The set of mechanical parameters extracted from these measurements is of great interest to get an insight on an ideal robotic tool to pluck harp strings. Based on these considerations, a highly-configurable and repeatable robotic finger has been designed.

Gi 3P

Acoustic radiation of a carbon composite soudboard guitar compared to wooden classical guitars

Sylvie Le Moyne, François Ollivier, Joël Frelat and Charles Besnainou

A new design carbon composite soundboard guitar (2012) is compared to traditional wooden guitars: a Antonio Torrès (1887), its copy by Thomas Norwood (2013) and a Ignatio Fléta (1981). Comparison leads to the absolute power radiation and the spectrum shapes. Globally, the composite guitar is the most powerful and its spectrum is better balanced.

Gi 4P

Vibroacoustic behavior of a vihuela

Sylvie Le Moyne, François Ollivier and Sandie Le Conte

The music museum of Paris possesses one of the three existing instruments certified to be vihuelas. None of these instruments is playable, and therefore no one can tell with certainty what the vihuela sounded like. The objective of the museum is to improve our knowledge of this unknown instrument and allow the general public to hear it. In this objective a copy was ordered and realized. An Impact

Near Field acoustic holography (INAH) measurement of the copied soundboard was performed. The sound radiated by the back of the copied instrument was measured with the same technique. For both the soundboard and the back, the spectrum of the radiated sound is investigated, and a modal analysis is proposed. The influence of the string tension on the soundboard acoustic radiation is also studied. Measurement on the copy, as it is complete and playable have the principal objective of improving our knowledge of this instrument.

Gi 5P

Eigenspectrum of the sarode membrane

Manaswi Mishra, Gowtham Suresh and Ronojoy Adhikari

The sarode is a multi-stringed musical instrument widely used in performances in the classical music of northern India. Its main parts consist of a solid body of carved wood, a fretless metal fingerboard, and a goatskin membrane radiator. While the instrument is second only in popularity to the sitar in north Indian music, to the best of our knowledge, no mathematical modeling of the physical instrument has been attempted to characterize its sound. Here, as a first step towards a complete mathematical model of the sarode, we compute the eigenvalues and eigenfunctions of the sarode membrane, the main radiating element of the instrument. We obtain the shape of the membrane from laser scans of a real instrument and then project it onto a computational mesh for finite element eigenanalysis. While the eigenfunctions can be characterised by the number of open and closed nodal contours, as is customary for highly symmetric circular or elliptic membranes, they are totally non-degenerate and have no symmetry about the nodal lines. The enhanced number of distinct eigenmodes that appear due to the peculiarity of the membrane shape adds to the timbre of the instrument. We also compute the eigenspectrum including radiation loss in a phenomenological manner to aid possible numerical modal sound synthesis.

Gi 6P

Ebony vs. rosewood: Experimental investigation about the influence of the fingerboard on the sound of a solid body electric guitar

Arthur Paté, Jean-Loïc Le Carrou and Benoît Fabre

Beyond electronics, lutherie also has something to do with the sound of the solid body electric guitar. The basis of its sound is indeed the conversion of the string vibration to an electrical signal. The string vibration is altered by coupling with the guitar at the neck. Electric guitar lutherie being a huge topic, this paper focuses on the influence of the fingerboard on the string vibration. An experimental study is carried out on two guitars whose only intentional difference is the fingerboard wood: ebony or rosewood. The well-known "dead spot" phenomenon is observed, where a frequency coincidence of string and structure at the coupling point leads to an abnormal damping of the note. Striking is the different behaviour of each fingerboard wood about dead spots: affected notes, as well as how much they are affected, differ with the wood.

Gi 7P

High-speed camera displacement measurement (HCDM) technique of string vibrations

Niko Plath

In this paper we describe a method for string vibration measurement through analysing high-speed camera recordings (HCDM). A short introduction into most common string motion detection methods is given and discussed. The experimental design for HCDM is described, covering the necessary settings needed to produce quantifiable recordings. The applied method of data analysis can be summarized as: String motions are converted to time series by tracking of string positions in every discrete time step through searching for high contrasts of RGB values in a row of recorded pixels. Examples of measurements of string motion are presented: The formation of the Helmholtz motion in a bowed violin string, the transversal versus the torsional motion of a plucked vihuela string, the transversal versus the longitudinal motion of a struck piano string and the effect of the una corda pedal for the transversal motion of three unison piano strings. Benefits and drawbacks of the presented method are discussed followed by possible improvements for future research.

Gi 8P

Finite element model of a kantele with improved sound radiation

Henna Tahvanainen, Jyrki Pölkki, Henri Penttinen and Vesa Välimäki

In this paper, a plucked string instrument called the *kantele* is modelled with the finite element method. The aim is to compare two traditional body structures and a modified body structure in terms of vibrational modes and radiation efficiency. The two traditional body structures are the closed box kantele and the top-plate kantele. In the modified structure, the top plate and the back plate are separated with an air gap. The modified structure has more vibrational modes than the traditional body structures, because it incorporates a freely vibrating top plate coupled with enclosed air. The simulations show that when the air gap is between 1-3 mm, the radiation efficiency of the modified kantele is higher than that of the traditional kanteles. This result supports previous research that concluded the modified kantele to be louder than the traditional top-plate kantele.

Gi 9P

The Bolivian charango – An acoustic study

Owen Woods and Jim Woodhouse

The charango, the subject of this paper, is a small stringed instrument from the Andes region of South America. The bridge admittance, mode shapes and impedance ratio were found for a range of charangos, a ukulele and a timple. A graphical method was developed in order to directly compare the results. It was found that the charangos tested were recognisably different from the other instruments. It was also found that the charangos tested were recognisably similar to one another. We can therefore conclude that it is meaningful to consider the charango as a distinct and homogenous instrument. The acoustic characteristics measured were explained using the social circumstances behind their construction, particularly the difference between campesino (peasant) and mestizo (urban) instruments. The most characteristic feature of the results, the ‘wasted’ Helmholtz resonance, was also explained in this way.

Singing Oral

Chair: *Johan Sundberg*

INVITED Si 1

Supranormal voices in singing

Ken-Ichi Sakakibara

Various supranormal voices are surveyed and synthesis methods for such voices based on physiological observations are discussed. In particular, physiological aspects on abrupt register changes, vocal-ventricular phonation and vocal-arytenoid phonation, which are widely found in ethno music or primitive music and recently also found in pop music, are described. Two synthesis methods for supranormal voices are proposed: physical modeling and signal modelling.

Si 2

Assessment of the acoustical impact of piriform sinuses in MRI based vocal tract replicas

Bertrand Delvaux and David Howard

Mixed soft/solid models of the vocal tract were molded with a 3D rapid prototyping technique based on MRI data obtained from two male singers during the phonation of five English vowels as in *hard*, *stern*, *neap*, *port* and *food*. The replicas are used to assess the spectral role of the piriform sinuses. In an anechoic chamber, a sound source producing a sinesweep is connected to the tracts. The acoustical response is then used to compute the transfer functions. In one case, the piriform sinuses remain empty, in the other, they are filled with water. The spectral impact of the piriform sinuses is discussed.

Si 3

Interference-free observation of temporal and spectral features in "shout" singing voices and their perceptual roles

Hideki Kawahara, Masanori Morise and Ken-Ichi Sakakibara

A new set of algorithms are introduced to represent and manipulate physical features found in expressive singing performances. The proposed representations are simple in both conceptual as well as computational aspects. This simplicity makes interpretation and control of the representations straightforward. First, a new fundamental frequency (F0) extractor revealed rapid and strong F0 modulation around 70Hz (in terms of modulation frequency) in a "shout" performance of a Japanese POP song. The F0 extractor is based on higher-order waveform symmetry and has finer temporal resolution than conventional methods. It is followed by a refinement procedure based on an interference-free representation of instantaneous frequency. Second, in interference-free representation of power spectrum revealed a synchronized spectral modulation to the rapid F0 modulation in "shout" performance. Third, one third octave level differences between plain performances and "shout" performances were observed. A set of manipulated singing voices were synthesized by manipulation of these features using an auto-regressive (AR) model-based notch-filtering in the modulation frequency domain and one-third octave FIR equalizers. The stimuli were synthesized using an extended version of a speech analysis, modification and resynthesis framework, TANDEM-STRAIGHT. Subjective tests illustrated that all these three features are effective contributing factors of "shout" impression. Physiological mechanism and possible applications are also discussed.

Si 4

Power control for the second harmonic at high pitches in soprano singing: A case study

Hironori Takemoto, Seiji Adachi, Takeshi Saitou, Kiyoshi Honda, Eri Haneishi and Hiroko Kishimoto

Sopranos have many singing skills relating to control of the vocal tract shape, especially at high pitches. In the present study, one of the skills called girare in Italian opera was examined. This skill is said to be used for producing a mild voice at high pitches. The midsagittal images of the vocal tract of a soprano were scanned by magnetic resonance imaging during singing of the vowel /a/ at G5 (784 Hz) and A5 (880 Hz) with and without girare. Acoustic analysis showed that girare selectively depressed the power of the second harmonic by more than 10 dB. This result implied that the power reduction could make the auditory impression of singing voice mild. Image analysis indicated that girare elongated the vocal tract, widened the oral cavity, and slightly constricted the lips. According to acoustic sensitivity analysis, all these changes tend to decrease the frequency of the second resonance of the vocal tract. Acoustic simulation confirmed that these changes in the vocal tract successfully reduced the power of the second harmonic by more than 10 dB.

Si 5

Formant frequencies of sung vowels intonated by six traditional Japanese Shigin singers. Part I: Dataset construction and analysis method

Masashi Nakayama, Kosuke Kato and Masaru Matsunaga

Shigin, a type of traditional Japanese singing, is a performance of reciting Japanese or Chinese poetry in Japanese. In order to understand the production mechanisms of Shigin, and thereby progress toward synthesized singing and singer training systems, the acoustic features of Shigin need to be clarified in relation to the interpretation styles of its practitioners. This study investigates each of the four lowest formant frequencies of sung vowels in relation to the singer's gender, the singer’s vocal part, intonated vowel (/u/, /o/, /a/, /e/, or /i/), the loudness level, and phonation frequency. Anechoic recordings of the steady-state portions of vowels intonated by six trained Shigin singers using different volume levels, pitches, and vowels, were used for the acoustic analysis. This paper (Part I) describes the method to construct the dataset for analysis and the method for analyzing the constructed dataset. Results of the analysis and the statistical investigation will be reported in the following paper (Part II).

Si 6

Formant frequencies of sung vowels intonated by six traditional Japanese Shigin singers. Part II: Results of analysis and statistical investigations

Kosuke Kato, Masashi Nakayama and Masaru Matsunaga

This study aims to clarify the acoustic characteristics of Shigin, a type of traditional Japanese singing. On the basis of the materials and method as described in Part I, this study investigates the four lowest formant frequencies of sung vowels in relation to the singer's gender, intonated vowel, the loudness level, and the phonation frequency. An analysis showed that: (1) The third formant frequency was about 400−500 Hz higher and the fourth formant frequency was about 1,000 Hz higher than those for operatic singing. (2) The mean of the second formant frequency of the /u/ vowel was similar to that for normal speech mode in Japanese but about 400 Hz higher than that for Noh singing and about 800 Hz higher than that for operatic singing. (3) 92.1% of the variance in the second formant frequency and 83.0% of the variance in the first formant frequency was described by a linear model that employs the four variables: intonated vowel, the singer's gender, the loudness level, and the phonation frequency. (4) Each of the first formant frequency and the second formant frequency was significantly increased with the phonation frequency. These findings can be considered as the major acoustic characteristics of Shigin.

Si 7

Experimental study of the frequency leap interval produced by the change of laryngeal vibratory mechanism during sustained notes

Sylvain Lamesch, Boris Doval and Michèle Castellengo

The transitions between two different laryngeal vibratory mechanisms are often characterised by frequency jumps. The leap interval of these frequency jumps is studied for the transitions from M1 to M2 and conversely from M2 to M1. Its correlation with the starting fundamental frequency, the vocal intensity and the vowel is investigated. Seven singers have produced sustained notes with laryngeal mechanisms transitions occurring during the production. The sound and the electroglottographic signals were recorded. The leap intervals values depend on the subject. However global tendencies can be observed for most of them: the leap interval rises with the musical dynamics for the M1->M2 transition, and decreases with the frequency for some subjects. Concerning the M2->M1 transition, no tendency was observed. The frequency leap interval does not depend on the vowel; however the results show individual strategies. The subglottal pressure at the beginning of the jump could play a role in the leap interval variation. Results show that the relation between the fundamental frequency and the subglottal pressure could be different in M1 and in M2.

Si 8

Acoustic characteristics of vibrato in different singing styles

Noam Amir, Irit Ronen and Ofer Amir

Vibrato is a commonly found characteristic of the singing voice, adding a sense of flexibility and richness. Acoustically, it is a periodic variation of frequency and amplitude, characterized by several parameters: rate – the numbers of cycles per second; extent – the degree of modulation of F0; amplitude – the degree of modulation in intensity. The purpose of the present study was to compare characteristics of vibrato across different singing styles. 429 samples were taken from commercial CDs, sung by thirty singers in three different singing styles: opera, jazz and pop. Vibrato rate tended to be slightly higher in the opera style, with borderline significance. Vibrato extent was significantly greater in the opera style than in the other two. These results imply that vibrato varies with musical style and support the idea that vibrato is a learned vocal skill rather than a natural vocal behavior.

Si 9

Diverse resonance tuning strategies for women singers

John Smith, Joe Wolfe, Nathalie Henrich and Maëva Garnier

Over the range 200 to 2000 Hz, the fundamental frequency f0 of women's singing voices covers the range of the first two resonances (R1 and R2) of the vocal tract. This allows diverse techniques of resonance tuning. Resonances were measured using broadband excitation at their lips. A commonly noted strategy, used by sopranos, and some altos, is to tune R1 close to the fundamental frequency f0 (R1:f0 tuning) once f0 approached the value of R1 of that vowel in speech. At extremely high pitch, sopranos could no longer increase R1 sufficiently and switched from R1:f0 to R2:f0 tuning. At lower pitch many singers of various singing styles found it advantageous to use R1:2f0 tuning Additionally, many sopranos employed R2:2f0 tuning over some of their range, often simultaneously with R1:f0 tuning.

Si 10

Glitch free FM vocal synthesis

Chris Chafe

Frequency Modulation (FM) and other audio rate non-linear modulation techniques like Waveshaping Digital Synthesis, Amplitude Modulation (AM) and their variants are well-known techniques for generating complex sound spectra. Kleimola (2013) provides a comprehensive and up-to-date description of the entire family. One shared trait is that synthesizing vocal sounds and other harmonically-structured sounds comprised of formants can be problematic because of an obstacle which causes distortions when cranking up time-varying controls. Large deflections of pitch or phoneme parameters cause jumps in the required integer approximations of formant center frequencies. Trying to imitate human vocal behavior with its often wide prosodic and expressive excursions causes audible clicks. A partial solution lay buried in some code from the 80's. This, combined with a new kind of oscillator bank described in Lazzarini and Timoney (2010) which produces uniform phase harmonic components ensures artifact-free, exact formant spectra even under the most extreme dynamic conditions. The paper revisits singing and speech synthesis using the classic FM single modulator / multiple-carrier structure pioneered by Chowning (1980). The revised method has been demonstrated as software written in Faust and is as efficient as its predecessor technique. FM for singing synthesis can now be 'abused' with radical time-varying controls. Applications are discussed, including an analysis - resynthesis speech coder.

Si 11

Testing a new protocol to measure tuning response behaviour in solo voice ensemble singing

Helena Daffern and Jude Brereton

Tuning within choirs when performing a capella continues to be a subject of interest to researchers and practitioners in terms of theoretical preferred tuning systems and the instinctive tuning behaviour of choral singers. Although more studies are exploring the issues surrounding tuning within choirs, most research on tuning in choirs focuses on the mean fundamental frequency of tones, either through analysis of performance, or perceptual listening tests. This paper considers a specific protocol and tools for analysis to assess the tuning characteristics of individuals singing in a choral texture. The results illustrate the tuning features throughout single notes and the extent to which singers adapt their tuning to the surrounding chord in equal temperament and just tuning. To test the effectiveness of this new methodology, data collected in a previous experiment is reconsidered in which singers were recorded performing a specially written choral exercise where the other parts were synthesised and heard over headphones. Overall the analysis proved a successful and efficient process. The results of the analysis suggest that further research into the tuning tendencies of choral singers, particularly in terms of fundamental frequency trajectories would prove valuable as a research area. Modifications to the protocol are considered for an upcoming study, which is planned for next year, and a revised hypothesis for the tuning strategies in the set exercise is presented.

Si 12

Temporal coordination in vocal duet performances of musical rounds

Caroline Palmer, Frances Spidle, Erik Koopmans and Peter Schubert

How do vocalists coordinate their timing with others in choral ensembles? Performers in many musical ensembles take on musical roles, such as a conductor or leader of the group; these roles may influence their synchronization. Pitch relationships between the singers' musical parts may also influence synchronization, such as the comparison between Unison singing (singing the same pitches) and musical Rounds (singing the same part, delayed in time). We compared the temporal coordination of duet performance in Unison and Round performances of a familiar melody. Vocalists with experience in ensemble performance sang the melody in duet performances; one participant was assigned the role of Leader and one of Follower, in Unison and Round performances. The tone onsets of the Leader preceded those of the Follower by a small but consistent amount. In addition, Rounds were performed at a slower rate than Unison performances, suggesting the increased difficulty of maintaining a different part. Finally, the pattern of correlations across the Leader and Follower's beat durations indicated strong similarity in tempo changes between the singers within each performance. The correlations were reduced in the Rounds relative to Unison performances, but Round performances showed similar tempo changes when the 2-bar delay between parts was taken into account.

Parametrization of Byzantine chant ethos through acoustic analysis: From theory to praxis

Anastasia Georgaki, Achilleas Chaldaikis and Takis Tzevelekos

In this paper we will analyze the notion of ethos in Byzantine music theory through a systematic parameterization of a vocal performance. In this direction we focus on the analysis of ethos from special pitch contours, melismatic microintervalic variations, intensity curves, timbral analysis, spectral variations and energy rates which indicate the emotional state of the performer (respecting the music notation and the meaning of the sacred text) trying to interpret the indicated mood by a concrete melismatic approach. More precisely in our approach we will attempt both structural and performance acoustic analysis as a first step in order to understand the different changements of the ethos in the macrostructure of the piece and by isolated phrases which indicate the the functionality of the sang mode (echos).

Si 1P

Evaluation of pitch detection algorithms: Case of monophonic vocal performance

Robertas Budrys and Rytis Ambrazevičius

Evaluation of musical scale in monophonic vocal performance is a complicated task, and some issues causing inaccurate results might occur. For instance, one issue is the selection of algorithm which detects the fundamental frequency of performance as precisely as possible. If precision of the algorithm is worse than jnd or if the algorithm detects unsteady pitch track inaccurately, its data is often not suitable for further analysis. In this paper, three different pitch detection algorithms are discussed: autocorrelation, YIN and SWIPE'. The algorithms were applied to generate pitch tracks of synthesized sine tones and examples of natural voice, and the results were compared.

Si 2P

Formant tuning in Byzantine chant

Georgios Chrysochoidis, Georgios Kouroupetroglou, Dimitrios Delviniotis and Sergios Theodoridis

We present an investigation of formant tuning in the context of the Byzantine Ecclesiastic chant voice. The recordings selected for the analysis are part of the DAMASKINOS prototype acoustic corpus of Byzantine Ecclesiastic voice. More specifically, we analyzed recordings from 10 different chanters in ascending musical scales of the diatonic genre, for the /a/ vowel. The method of analysis included a semi-automatic segmentation of the audio material, extraction of the measurements in PRAAT and the final post-processing in MATLAB. Results show clear evidence of formant tuning in at least six of the chanters, proving that the technique is in use by the modern Byzantine music performers.

Si 3P

Vibrato analysis in Byzantine music

Dimitrios Delviniotis, Georgios Kouroupetroglou and Sergios Theodoridis

In this work we study the vibrato rate, extent and intonation in Byzantine Music. Two methods of analysis have been applied: the first based on the analytical signal and the second on the crests and troughs of the waveform of the vibrato signal. Tones - samples were extracted from ascending and descending music scales, chanted by five famous singers for all the Greek vowels. The two analysis methods produced identical results in the level of significance, a=0.05, concerning the mean extent, the mean standard deviation of the rate and the mean intonation, while they differed in the rate (1.9 - 6.6%), the mean standard deviation of the extent (4-6 cents) and the standard deviation of intonation (0.46 - 1.20 Hz). Typical values of the average rate within a tone were found to be 5.35 Hz (SD: 0.96 Hz) and 5.13 Hz (SD: 0.95 Hz), while the most frequent values were 4.8 and 4.5 Hz, for the first and second method, respectively. The average extent within a tone was 50 cents (SD: 18 cents). Moreover, the dependence of the vibrato parameters on pitch and sound intensity has been studied; there was no systematic relationship between them.

Si 4P

A pilot study of vibration pattern measurement for facial surface during singing by using scanning vibrometer

Tatsuya Kitamura, Hiroaki Hatano, Takeshi Saitou, Yui Shimokura, Eri Haneishi and Hiroko Kishimoto

We attempted to measure the vibration velocity patterns of the facial surface during sustained singing using a scanning laser Doppler vibrometer. The measurement system used allowed laser-based, noncontact, and multipoint measurements of the vibration of objects. Three Japanese female professional singers (A, B, and C) participated in three experiments. In the first experiment, the vibration velocity patterns for the Japanese vowels /a/ and /i/ were obtained for the singers. The patterns for the vowels demonstrated a clear contrast, especially around the nose and cheek. In the second experiment, the differences in the vibration velocity patterns due to the pitch frequency were compared for Singer A. The results show that the amplitude of the vibration velocity of the forehead for a higher pitch frequency (F5=698.5 Hz) is larger than that for a lower pitch frequency (A4=440 Hz). In the third experiment, the effects of methods of vocalization on the vibration velocity pattern were measured while Singer A sang in falsetto and modal voice. A significant difference in the amplitude of the volume velocity of the cheek was observed.

Si 5P

Postural sway in vocal duets

Erik Koopmans, Caroline Palmer and Frances Spidle

Postural control is important for singing, and is constrained by the demands of breath support. We examined postural sway in vocal duets and the influence of visual and auditory cues on posture during vocal performance. Experienced vocalists performed simple melodies in Solo, Unison (same musical part sung by two vocalists simultaneously), and Round conditions (same musical part sung at a temporal delay) while standing on force plates. Visual feedback about the partners' movements was manipulated by facing the singers Inward (with full view of each other) and Outward (with no view of each other), while normal acoustic feedback was maintained across all conditions. Measures of ground reaction forces and center of pressure both indicated increased variability in sway in Rounds compared with Unison performances, and when the vocalists had full visual cues in Inward-facing conditions. Correlations of vertical ground reaction force measures indicated more agreement between singers during production of Unison singing than Round singing. Although visual cues about one's partner had some influence on the variability of vocalists' posture during duet performance, only the type of performance (Unison/Round) influenced the degree of correspondence in posture between vocal duet members.

Si 6P

A study of speaker dependent formant space variations in karaoke singing

Mahnoosh Mehrabani and John H.L. Hansen

It has been shown that trained and professional singers vary their formant frequencies due to vowel modification while singing. In this study we analyze the formant space of vowels for Hindi amateur singers, singing Karaoke. The first and second formants are examined for each speaker, and the deviation of formant space configuration from speaking to singing is compared for different speakers. It is shown that while singing vowel space varies form one speaker to the other, for all speakers the average distance between vowels in F2/F1 plane decreases from speaking to singing. Correlation of F1 and F2 with F0 is also studied for different vowels, and speaker dependent variations of formant tuning are analyzed. In addition to the scientific value, the results of this study can be applied to singer identification, singing skill evaluation, and singing quality assessment.

Si 7P

An attempt to develop a singing synthesizer by collaborative creation

Masanori Morise

This paper presents singing synthesizers collaboratively designed by several developers. In the video-sharing Web site Nico Nico Douga, many creators jointly create songs with a singing synthesis system called Vocaloid. To synthesize various styles of singer, another singing system UTAU which is a free software, is being developed and used by many creators. However, the sound quality of this system is not yet as good as Vocaloid. The purpose of this study is to develop a singing synthesizer for UTAU by collaborative creation. Developers were encouraged to design a singing synthesizer by using a high-quality speech synthesis system named WORLD that can synthesize a singing voice that sounds as natural as a human voice. We released WORLD and a singing synthesizer for UTAU as free software with C language source code and attempted to encourage collaborative creation. As a result of our attempt, six singing synthesizers for UTAU and two original singing synthesis systems were developed and released. These were used to create many songs that were evaluated as high-quality singing by audiences on a video-sharing Web site Nico Nico Douga.

Si 8P

A method of division of soprano ranges and confirmation of their voice transformation point based on harmonics analysis

Ge Qu

This study raises a new method to divide the soprano ranges and to find the sound transformation point with the analysis of the fundamental frequency of every sound in the whole range, the numbers of visible harmonics, the intensity of basic sounds, the difference of intensity among the first, second and third harmonic and the types of envelope, etc, besides the judgments of instructor's subjective hearing and the singer's self-sense when the three sopranos singing the vowel /a/.

Si 9P

Generating singing voice expression contours based on unit selection

Marti Umbert, Jordi Bonada and Merlijn Blaauw

A common problem of many current singing voice synthesizers is that obtaining a natural-sounding and expressive performance requires a lot of manual user input. This thus becomes a time-consuming and difficult task. In this paper we introduce a unit selection-based approach for the generation of expression parameters that control the synthesizer. Given the notes of a target score, the system is able to automatically generate pitch and dynamics contours. These are derived from a database of singer recordings containing expressive excerpts. In our experiments the database contained a small set of songs belonging to a single singer and style. The basic length of units is set to three consecutive notes or silences, representing a local expression context. To generate the contours, first an optimal sequence of overlapping units is selected according to a minimum cost criteria. Then, these are time scaled and pitch shifted to match the target score. Finally, the overlapping, transformed units are crossfaded to produce the output contours. In the transformation process, special care is taken with respect to the attacks and releases of notes. A parametric model of vibratos is used to allow transformation without affecting vibrato properties such as rate, depth or underlying baseline pitch. The results of a perceptual evaluation show that the proposed approach is comparable to parameters that are manually tuned by expert users and outperforms a baseline system based on heuristic rules.

Woodwinds – reeds and flutes Oral

Chairs: *Murray Campbell* & *Miko Hirschberg*

INVITED Wo 1

The player–wind instrument interaction

Joe Wolfe, André Almeida, Jer Ming Chen, David George, Noel Hanna and John Smith

Players control a range of parameters in the playerinstrument system. First we show how loudness and pitch vary over the plane of mouth pressure and force on the reed of a clarinet, and thus how these parameters can be used in compensation to produce trajectories in this plane that have varying loudness and timbre but constant pitch. Next we present impedance spectra for several different types of musical instruments and for the vocal tract, to allow general observations. We report different ways in which the acoustic properties of the player's tract interact with those of the instrument bore to control the frequency of reed vibration in some wind instruments. We also show how vocal tract resonances can influence timbre.

Wo 2

Simulations of modal active control applied to the self-sustained oscillations of the clarinet

Thibaut Meurisse, Adrien Mamou-Mani, René Caussé and David Sharp

Modal active control enables modifications of the damping and the frequencies of the different resonances of a system. A self-sustained oscillating wind instrument is modeled as a disturbance coupled to a resonator through a non-linear coupling. The aim of this study is to present simulations of modal active control applied to a modeled simplified self-sustained oscillating wind instrument (e.g. a cylindrical tube coupled to a reed, which is considered to approximate a simplified clarinet), incorporating collocated speaker, microphone and a reed. The next goal will be to apply this control experimentally and to test it with musicians.

Wo 3

An attempt at predicting the variation in playing frequencies for clarinets

Whitney Coyle, Jean Kergomard, Philippe Guillemain, Christophe Vergez and Alexis Guilloteau

The input impedance measurement is today a standard method used by several wind instruments makers for the design of modifications. For small modifications, the knowledge of this quantity, and especially the resonance frequencies, is often sufficient. However for a complete design, it is much better to know the playing frequencies themselves, which depend on several control parameters, such as the mouth pressure or the reed opening. Starting from values of these parameters, numerical computation (either in time of frequency domains) allows us to determine the playing frequencies. This paper presents an attempt at deducing, analytically, these frequencies from the different control parameters, and the knowledge of the input impedance curve. Three effects are examined separately: the flow rate due to the reed motion, the reed dynamics, and the inharmonicity of the resonator. An example of results is given for a clarinet.

Wo 4

Graph-based models for woodwinds

Georges Le Vey

A model for woodwinds with tone and register holes is presented. It is inspired by the original idea of A. H. Benade considering the set of toneholes as a sequence of 'T-shaped' sections. This idea can be deepened thanks to more recent works on mathematical modelling and analysis of repetitive structures such as networks of strings, beams, membranes, pipes or canals. An essential feature of the model is that it keeps at the one dimensional level while mode matching is automatically satisfied. The purpose of this work is to build upon the idea of Benade, inside a precise mathematical framework using concepts and methods from graph theory, for modelling the bore of woodwinds together with its holes in order to address questions like length corrections due to the lattice of (closed or open) toneholes or toneholes interactions. The long term objective is to use this type of model for simulation, characterization of the natural frequencies of woodwinds and, in a control theoretical setting, for musical acoustics questions such as design as well as for theoretical purposes. Applications of the approach are exposed as a program for future research.

Wo 5

A study of sound characteristics of a new bassoon as compared to the modern German bassoon

Timo Grothe and Peter Wolf

An interdisciplinary team of instrument makers, engineers and musicians has recently developed a new bassoon-like instrument. Projected fields of use for this instrument are ensembles, in which bassoonists have to compete with the brasses in terms of sound power. In order to overcome dynamic limitations of the modern bassoon and to achieve a brighter tone-color, a drastically re-designed air column is introduced. Compared to the bassoon, the new instrument has a larger taper which is constant along the main bore, as well as wider and shorter tone holes, which are entirely operated by keys. A chromatically playable prototype with full keywork has been built. The new instrument has powerful bass, an enriched overtone spectrum and can be played significantly louder than the modern bassoon, so we call it Bassoforte. This comparative study presents sound recordings of bassoon and Bassoforte in a reverberation chamber and in a recording studio, as well as some considerations on the acoustical design based on input impedance curves. Our results indicate that the cutoff-frequency of the aircolumn is correlated with formant frequencies in the sound spectrum. This suggests an optimization target for reconsiderations of the tonehole concept, to overcome an observed unevenness in timbre.

Wo 6

An experimental study of temperature variations inside a clarinet

Daniel Noreland

A method for measuring the air column temperature of woodwind instruments using a medical infrared thermometer is devised and applied on a clarinet. It is found that the temperature varies roughly linearly along the bore, but with a considerable temperature swing according to playing conditions. If the temperature is approximated by an average along the bore, computations erroneously predict the octave to be compressed by 8.5 and 5.7 cents in the fundamental and second registers, respectively. The importance of considering humidity in accurate woodwind modelling is pointed out as well.

Wo 7

Vocal tract effects on the timbre of the saxophone

Weicong Li, Jer-Ming Chen, John Smith and Joe Wolfe

For notes sounded over the normal and altissimo playing range, experienced saxophonists can produce changes in the spectral envelope of the radiated sound by adjusting their vocal tract configuration. Measurements of the vocal tract acoustic impedance, Z_{mouth} during performance showed that when Z_{mouth} was comparable with the input impedance of the bore, Z_{bore}, i.e. several MPa·s·m^{-3} or more, harmonics of the radiated sound falling near these peaks in Z_{mouth} were substantially enhanced. In contrast, the broad band noise in the radiated sound produced by upstream turbulence was attenuated in the frequency range over which the magnitude of Z_{mouth} was large.

Wo 8

“In vivo” and “in vitro” characterization of single cane reeds

Alberto Munoz, Bruno Gazengel and Jean Pierre Dalmont

The aim of this paper is to estimate the characteristic of the reed defined by the relation between the pressure drop accross the reed channel and the displacement of the reed tip. Two differential pressure sensors are used to measure the pressure inside the mouth and the mouthpiece. A photointerruptor placed in the mouthpiece is calibrated and used in order to measure the reed tip displacement. In vivo measurements are performed when a musician plays a simplified clarinet. Using the measured characteristic in vivo, a phenomenological model is derived and enables to estimate some parameters which describe the reed behaviour. In vitro measurements are also performed using a vacuum pump to make the reed bend against the mouthpiece lay without any artificial lip. Using a physical model of the reed mechanics, parameters are also deduced to characterise the reed behaviour.

Wo 9

Study of the perceived quality of saxophone reeds by a panel of musicians

Jean-Francois Petiot, Pierric Kersaudy, Gary Scavone and Stephen McAdams

The subjective quality of cane reeds used on saxophones or clarinets may be very different from one reed to another even though the reeds have the same shape and strength. The aim of this work is to study the differences in the subjective quality of reeds, assessed by a panel of musicians. The work focuses mainly on the agreement of the panel of musicians, the reliability of the evaluations and the discrimination power of the panel. A subjective study, involving 10 skilled musicians, was conducted on a set of 20 reeds of the same strength. Three descriptors were assessed: Brightness, Softness, and Global quality. The ratings of the musicians were analyzed using sensory data analysis methods to estimate the agreement between them and the main consensual differences between the reeds. Results show that for Softness and Brightness, the agreement between the musicians is important and that significant differences between the reeds can be observed. For Global quality, the inter-individual differences are more important. The performance of the panel in providing reliable assessments opens the potential for an objectification of the perceived quality.

Wo 10

Influence of pad "resonators" on saxophone

Pauline Eveno, Marthe Curtit, Jean-Pierre Dalmont and René Caussé

Toneholes have an important role in the acoustics of woodwind instruments. In the saxophone, the toneholes are surmounted by a key provided with a pad and a "resonator", as called by musicians and craftsmen. "Resonators" are flat disks made of metal or plastic fixed in the middle of the pad. In order to understand their role, measurements of the input impedance of a cylinder topped by a key with interchangeable pads (with and without "resonators") are performed. For closed holes, pads with "resonators" have a low absorption coefficient and effects of the "resonators" on the radiation are highlighted for small key heights. A study of the pad vibrations shows that these effects can be explained by the high mobility of the pad without "resonator", which seems to be a "stiffener". Then, the input impedance measurement of a whole saxophone confirms that when the holes are closed, the effect of a pad without "resonator" is to increase the damping. The effect on open holes is negligible. Finally, measurements in playing situation show that saxophones without "resonator" have a higher HSC and require a higher mouth pressure.

Wo 11

Determination of 2D quasi incompressible flow around a recorder labium: A comparison between different methods

Roman Auvray, Pierre-Yves Lagrée and Benoît Fabre

The shape of the labium has important consequences on the sound produced by flute-like instrument. This statement is well known by instrument makers who take extremely care to this precise part. The sharp edge of the labium modifies both acoustic and hydrodynamic properties. Non-linear acoustic phenomena and intricate vortex structures might strongly depend on the shape of the labium. A first step in the study of the labium is to consider the acoustic part only. This paper presents a comparison of different numerical methods to estimate the linear part of the acoustic flow around the labium. Results are discussed with respect to the sharpness and the angle of the labium, two main features of the labium that have already been studied experimentally. Finally, one of the methods is proposed as a good candidate to include in a sound production model.

Wo 12

Is the jet-drive flute model able to produce modulated sounds like Flautas de Chinos?

Soizic Terrien, Christophe Vergez, Patricio de La Cuadra and Benoît Fabre

"Flautas de chinos" - prehispanic Chilean flutes played during ritual celebrations in central Chile - are known to produce very particular beating sounds, the so-called "sonido rajado". Some previous works have focused on the spectral analysis of these sounds, and on the input impedance of the complex resonator. However, the beating sounds origin remains to be investigated. Throughout this paper, a comparison is provided between the characteristics of both the sound produced by "flautas de chinos" and a synthesis sound obtained through time-domain simulation of the jet-drive model for flute-like instruments. Jet-drive model appears to be able to produce quasiperiodic sounds similar to "sonido rajado." Finally, the analysis of the system dynamics through numerical continuation methods allows to explore the production mechanism of these quasiperiodic regimes.

Wo 13

The design of a chromatic quena: How can linear acoustic help?

Camille Vauthrin, Benoit Fabre and Patricio de La Cuadra

While traditional instrument making relies on a trial and error process, acoustics may help in designing a new instrument or evolution of existing instruments; resulting in a faster/more efficient designing process and/or better quality instrument. Because of complex and intricate relations between the different building parameters, it also may focus work on the specific parameters. This work can also help to grasp the global influence of each geometric parameter on the note's pitch, harmonicity between the registers, the timbre or the field of freedom in play. The definition of the requirements provides the objectives and the limitations of the design process. The requirements are established from linear acoustics, discussions with the flute maker. The prototype of the chromatic quena should correspond to these; especially of the crossover between Andean sounding aesthetics and modern flute playing techniques. Linear acoustic allows for relatively correct details on resonance frequencies, however the timbre is more difficult to studied. In the aim to approach the diatonic quena's timbre, their distinguished elements are preserved on the chromatic quena as the notch, the knot. Moreover, large holes allow to obtain a regular timbre. This work shows that this design process can help the flute maker in realization of chromatic quena, especially in reducing the number of prototypes.

Wo 14

Numerical modeling of a recorder in three dimensions

Nicholas Giordano

A modeling study of the recorder using a direct numerical solution of the Navier-Stokes equations is described. Results for the spectrum as a function of the blowing speed u are presented. An increase in the mode frequencies and switching between modes are observed as u is increased, in qualitative agreement with the behavior found for recorders and flue organ pipes.

Wo 1P

Comparison of two methods of sound power measurements of flue organ pipes

Judit Angster, Katrin Hoge and Andras Miklos

The Fraunhofer IBP (Stuttgart, Germany) realized with 10 organ builder companies from 9 countries a research project, which was supported by the European Union and dealt with the matching of the pipe organ to the room. The aim was to assist organ builders already during the planning phase of an instrument. An important parameter for acoustic design is the radiated sound power of the instrument, and thus of the pipes. With regard to the specific features of the sound of an open flue pipe the reverberation room procedure was selected from possible measurement methods to determine the sound power. Measurements were carried out to show the influence of wind pressure, pitch and geometry (scaling) of the flue organ pipes on the sound power. The measurements were significantly influenced by the strong interaction of the harmonic spectrum of the pipe sound with the eigenmodes of the reverberation room and by the coherent radiation of both point-like sound sources at the mouth and open end of the flue organ pipe. In a second measuring procedure the measurements were performed by a microphone-array system (NS-STSF/SONAH) in the anechoic room. The results as well as the advantages and disadvantages of the measuring methods and a view of other possibilities for the detection of the sound power of flue organ pipes are discussed.

Wo 2P

Prediction of the dynamic oscillation threshold of a clarinet model: Comparison between analytical predictions and simulation results

Baptiste Bergeot, André Almeida, Christophe Vergez and Bruno Gazengel

Simple models of clarinet instruments based on iterated maps have been used in the past to successfully estimate the threshold of oscillation of this instrument as a function of a constant blowing pressure. However, when the blowing pressure gradually increases through time, the oscillations appear at a much higher value than what is predicted in the static case. This is known as bifurcation delay, a phenomenon studied for a clarinet model. In numerical simulations the bifurcation delay showed a strong sensitivity to numerical precision. This paper presents an analytical estimation of the bifurcation delay of the simplified clarinet model taking into account the numerical precision of the computer. The model is then shown to correctly predict the bifurcation delay in numerical simulations.

Wo 3P

On reeds and resonators: Possible explanations for cyclic spectral envelopes in the case of double reed instruments

Sandra Carral and Christoph Reuter

Proponents of the Pulse Forming Theory claim that the reed closing time of wind instruments remains approximately constant over their playing range, causing constant spectral gaps and therefore formants in the spectra of wind instruments. One of the latest measurements of the oboe reed closing time of 0.4 to 0.5 ms dates back to the 1970's, and is often quoted thereafter in other articles. They also claim that the reed closing time depends only on the characteristics of the reed, and disregard the influence of the geometry of the instrument. In this article a modern, minimally intrusive and accurate method for measuring the closing time of the oboe reed is presented. Other possible theories that could explain the approximately constant closing time of the reed in woodwind instruments are also discussed.

Wo 4P

External pipe resonators and harmonica acoustics

James Cottingham and Casey Brock

Measurements of reed motion and sound field have been made on a diatonic harmonica mounted on a fixed volume wind chamber. These include variation of sounding frequency with blowing pressure, and the degree to which the sounding frequency and sound spectrum can be altered by attaching external pipe resonators. The current results are compared with the results of measurements made in earlier studies. Differences were observed between the behavior of blow and draw reeds, as well as the dependence of the results on whether the secondary reed in the reed chamber is allowed to vibrate. For frequencies bent down with the pipe resonators, the sound and reed motion spectra displayed interesting changes in timbre and increased high frequency activity, some of which may be a related to the second transverse mode of reed vibration possibly involving coupling to the reed chamber resonance.

Wo 5P

Investigation of bassoon directivity

Timo Grothe and Malte Kob

Due to the distribution of tone holes across a long bent corpus, the bassoon has an irregular directivity. This is well-known for Tonmeister, who whish to record the sound of a bassoon with a small number of spot microphones. This study presents the directivity patterns of the bassoon obtained in an anechoic chamber with different methods. Using an artificial mouth with adjustable embouchure, the instrument was excited at different pressure levels that were held constant for a time span of minutes. Several fingerings, covering most of the playing range of the bassoon ($f0 = 58 - 591$ Hz) have been sounded while the bassoon was mounted vertically and horizontally with respect to the rotary axis of the turntable. The use of the artificial mouth allowed studying influences of the fingering as well as of the playing dynamics on the pressure radiation pattern. As a second method an acoustic camera was used to evaluate the sound radiation characteristics.

Wo 6P

Evaluating a wavelet-based analysis of sensor reed signals for performance research

Alex Hofmann, Werner Goebl and Michael Weilguni

Empirical investigations of saxophone and clarinet performance are important for a thorough understanding of the human motor skills required to musically operate woodwind instruments. In this paper, we discuss two methods of detecting tonguing related landmarks in a sensor saxophone reed signal. We give detail about the reed signal's characteristics under three typical playing instructions (legato, portato and staccato articulation) and define detection tasks for physical tone onsets and tone offsets. When the player's tongue contacts the reed, the oscillations are dampened and the reed is bent towards the mouthpiece (tongue-reed contact). Removing the tongue from the reed returns it to its equilibrium position (tongue-reed release). From these observations we derive two landmark detection functions: a heuristic for peak detection, based on thresholding the smoothed reed signal, and a wavelet-based analysis, operating on specific sub-bands of the reed signal. When evaluating both methods, the wavelet analysis revealed better results using our current test dataset.

Wo 7P

The voice of the mechanical dragon

Michel Hirschberg, Oleksii Rudenko, Gunes Nakiboglu, Ad Holten, Jan Willems and Avraham Hirschberg

The sound radiation of a swinging corrugated tube (Hummer) has been measured under anechoic and semi-anechoic conditions. The whistling is induced by synchronized vortex shedding at each corrugation coupled to an acoustic standing longitudinal wave. In an earlier paper the Hummer was hand-driven. In order to eliminate this human factor, the instrument is driven mechanically. This considerably enhances the agreement between measured sound radiation and prediction by a model assuming at the open ends acoustic velocity fluctuations of 5\% of the main flow velocity through the tube. These pipe terminations act as monopole sound sources. Significant deviations from theory still remain. In particular the amplitude modulation of the sound, due to interference between the two sources, is deeper than predicted. Experiments also show the possible coexistence of two acoustic modes, which is not considered in the theory. The possible elongation of the tube by centrifugal force and flutter due to vortex shedding appears to be negligible.

Wo 8P

Embracing the digital in instrument making: Towards a musician-tailored mouthpiece by 3D printing

Valerio Lorenzoni, Zjenja Doubrovski and Jouke Verlinden

At present, the manufacturing of musical instruments still strongly relies on the tacit knowledge of experienced handcrafts while is commonly based on standard machining or casting techniques. This limits the musician-tailoredness to a small group of players, while others take compromises by employing stock parts. The present article describes a new methodology for the design and production of woodwind instruments mouthpieces. The aim of the presented methodology is to link the geometry of the mouthpiece to tone properties. Based on 3D printing, the inside geometry can be altered to complex and reproducible detail to obtain the desired acoustic features - eventually leading to mouthpiece geometries tailored to the player's sound and playability requirements. The results of aerodynamic investigations together with the subjective experience of saxophone players have been used to design mouthpieces with modified inside geometries of both baffle and chamber. Future work includes additional measurements and developing a parameterized database of 3D models.

Wo 9P

A digital bagpipe chanter system to assist in one-to-one piping tuition

Duncan Menzies and Andrew McPherson

This paper describes an electronic bagpipe chanter interface and software system, developed to assist in the practice of one-to-one Highland piping tuition. The chanter employs infrared reflectance sensors to detect the continuous movements of the player's fingers, and incorporates an air pressure sensor in place of the chanter reed, allowing it to be connected to a traditional acoustic set of pipes. The software is intended to assist the instructor in communicating feedback to the student by providing facilities for recording, playback, visualisation and comparison of teacher and pupil performances. A user study of the system was carried out with an experienced piping instructor and seven students at a school in NE Scotland. The sessions yielded encouraging and constructive feedback from both the students and instructor, and produced promising avenues for further work.

Wo 10P

Numerical analysis of the mean flow effect on the sound directivity pattern of cylindrical ducts

Yong Shi, Andrey Da Silva and Gary Scavone

This paper presents a numerical investigation of the sound directivity pattern of normal modes radiated from the open end of a cylindrical pipe. A good agreement is found between the numerical results and the analytical predictions of the directivity pattern for an unfledged pipe in the absence and in the presence of a low Mach-number mean flow. The investigations are conducted by using an axisymmetric two-dimensional lattice Boltzmann model. The numerical model is first validated by comparing its directivity with the established analytical model by Levine and Schwinger for the case of zero mean flow. Then the numerical results under the condition of mean flow with two different Mach numbers are compared with the analytical model by Gabard and Astley and recent experiment observations by Gorazd et al. The effects of the so-called zone of relative silence are observed in the results even for very low Mach number (M=0.036).

Brass instruments Oral

Chair: *Benoît Fabre*

Br 1

Lip motion, the playing frequency of the trombone and the upstream and downstream impedances

Henri Boutin, Neville Fletcher, John Smith and Joe Wolfe

We report the motion of trombone players' lips, its phase with respect to the mouthpiece pressure, the impedances of the bore and the player's vocal tract, and the frequency difference between the bore resonance and played note. The bore resonance frequency shifts very little with playing and often decreases somewhat: the effect of CO_2 can exceed that of temperature and humidity. The bore impedance is usually compliant for the note Bb2 and inertive for Bb3. The vocal tract impedance measured at the player's mouth is inertive for both notes. In terms of Fletcher's simple model for regeneration (JASA, 93, 2172), the results are consistent with a (+1,−1) valve for Bb2 and (−1,+1) for Bb3. The pressure in the mouthpiece in both cases rises before the lips separate. For Bb3, where the lip motion is mainly transverse, this is consistent with the inertive load. For Bb2, the substantial motion of the lips in the direction of flow provides a sweeping motion which produces the current into the bore that precedes lip opening.

Br 2

Trombone sound simulation under varying upstream coupling conditions

Vincent Fréour and Gary P. Scavone

The acoustical influence of the upstream airways is an important issue in brass performance. Analyzing the modalities of upstream resonance adjustments around the playing frequency will improve our understanding of vocal-tract tuning and lip-valve mechanics in brass instrument playing. In this study, different conditions of upstream coupling are simulated at the fundamental frequency using a simple one-mass model of the lips coupled to a trombone resonator. Maintaining a constant amplitude of the upstream acoustic pressure, variations of the phase of the upstream relative to the downstream input impedance at f_0 result in changes in playing frequency, as well as in the downstream acoustic energy produced. Further analysis shows that this upstream acoustic control can displace the playing frequency near the lip natural frequency, allowing optimal efficiency of the mechanical lip-valve system. These results highlight the possible importance of upstream phase tuning as part of a vocal-tract tuning strategy in brass performance. It further suggests a new experimental method for the estimation of lip natural frequencies on artificial player systems.

Br 3

Time domain simulation of standing waves in brass wind instruments taking non-linear wave steepening into account

Wilfried Kausel and Clemens Bernhard Geyer

Nonlinear wave steepening up to the degree of shock wave formation is commonly associated with the observed spectral enrichment of brass wind instrument sounds with increasing dynamic level. By modulating fractional delay stages - similar to a method known for producing arbitrary non-linear audio effects - the known physical relationships between local sound pressure, temperature, fluid velocity and wave propagation speed can be enforced in time domain simulations. This way a realistic model of bi-directional non-linear wave propagation can be established. A chain of such non-linear propagation elements combined with traditional Digital-Wave-Guide (DWG) scattering elements and loss filters can be used for closed loop simulations including vibrating lips and realistic radiation conditions. Arbitrary acoustical ducts, like real brass wind instruments defined by their accurate bore profiles, can be simulated and their characteristic sound synthesized. For obtaining the numerical results presented below, ART (Acoustic Research Tool) has been used, which is an OpenSource simulation framework and model library for acoustical simulations in the frequency and time domain. The simulated shock formation distance is in good agreement with analytic results, eg. 4.3 - 8.6 m for sinusoidal stimulus at 170 dB between 87 and 175 Hz (the low octave of a tenor trombone).

Br 4

Control of an artificial mouth playing a trombone and analysis of sound descriptors on experimental data

Nicolas Lopes, Thomas Hélie and René Caussé

This paper deals with a robotized artificial mouth adapted to brass instruments. A technical description of the robotic platform is drawn, including calibrations, initialization processes, and modes of control. An experimental protocol is proposed and the repeatability is checked. Then, experiments are conducted on a trombone for several types of quasi-static controls. Sound descriptors (fundamental frequency, roughness, energy) of measured acoustic signals are estimated and used to build cartographies indexed by the control inputs. An analysis reveals that several stable notes can easily be reached using a basic mapping with respect to these control inputs. However, the histogram of fundamental frequencies shows that input impedance peaks in the high range of the instrument do not correspond to notes that can be played by the artificial mouth, contrarily to musicians. It also reveals that some notes are difficult to play in the middle range. This exploration suggests some possible improvements of the machine that are finally discussed.

Br 5

Muscle activity in playing trumpet: the dependence on the playable pitch region and the experience of a non-trumpet brass instrument player

Shogo Matsukata, Hiroko Terasawa, Masaki Matsubara and Tetsuro Kitahara

In this paper, we investigated the relationship between the muscle activities around the lips and the sounds of playing the trumpet. While one is playing the trumpet, it is important but also difficult to keep an embouchure. Previous studies have investigated the difference between muscle activities for register and proficiency, but these studies have not considered the player's playable register and the influence of experience playing other instruments. Thus, in this study we aimed to solve these problems by examining the surface electromyograms (EMGs) around the lips: the orbicularis oris superioris, orbicularis oris inferioris, depressor anguli oris, and levator anguli oris. As a result, we found that (1) muscle activity is greater in the high register than the low register, but muscle activities in the high register are not significantly greater than those in the low register for the wide range of the playable register in the fixed register (FR); and (2) a novice trumpeter has greater muscle activity in the upper lip than in the lower lip, while an expert shows no difference. The muscle activity is the same as for the novice player when a non-trumpet brass player plays the trumpet.

Br 6

Timpani-horn interactions at the player's lips

Jer-Ming Chen, John Smith and Joe Wolfe

This study investigates the observation by some horn players that a timpani sounding nearby can interfere with their playing. By determining the horn’s transfer function and measuring the pressure response in the bell and mouthpiece during moderate to loud timpani strokes, the horn is found to behave as an acoustic impedance-matching device capable of transmitting an overall impulse gain response of at least ~16 dB from the bell to the mouthpiece, while some non-linear propagation in the bore is also observed. Further resonance interactions between the bore of the horn and the timpani stroke show gain responses of up to ~26 dB, which depend on the timpani’s tuning. Lastly, pressure measurements in the mouthpiece made during horn playing show that timpani strokes played near the bell can affect the amplitude, periodicity and frequency of the pressure signal generated at the horn player’s lips, and may be large enough to perturb the player’s musical performance.

Pitch bending techniques on early horns by manipulation of the embouchure: A comparison between measured and predicted data.

Lisa Norman, Jonathan Kemp, John Chick and Murray Campbell

Brass players sometimes adopt a technique whereby they adjust their embouchure in order to alter or bend the pitch of a note away from the centre of the resonance. The ease and control with which this can be achieved is an important factor in assessing the playability of a brass instrument. A good instrument will have well defined resonances, but experienced players do not like instruments with notes that are too `stiff', and which lack sufficient flexibility for musical expression. The need for the ability to bend the pitch of a note is particularly important for natural trumpets and horns used in the baroque era, when instruments did not have valves and players were required to bend the pitch of some resonances (e.g. the 11th and 13th) by a significant fraction of a semitone. The instrument and player form a complex and closely coupled system. Using experimental data from playing tests on early orchestral horns, and comparing these results with those from a recently developed time domain model, it is possible to begin to identify features of an instrument and its interaction with a player which make it more or less susceptible to this type of manipulation.

Br 1P

The use of the input impedance for characterising historical serpents

Pauline Eveno and Sandie Le Conte

This article describes how the input impedance is used as a descriptor to classify and characterise the corpus of serpents from the Musée de la Musique. The study focuses on the production of a family of craftsmen (Baudouin father and son) with the objective to find a trace of the know-how transmission. Input impedance measurements highlight some differences between serpents made by Baudoin and a serpent from another craftsman but not allow discriminating between instruments made by the father or the son.

Br 2P

Sensor based hand and lip weight and pressure measurements in trombone playing

Tobias Großhauser

Trombone players are exposed to high physical stress while playing. This paper describes a first step towards a physical stress measurement setup. It is based on flexible force sensitive resistors (FSRs) fixed to the trombone. The load of the left hand holding the instrument, the right hand and the overall pressure on the lips is observed. The results show the pressure distribution between the fingers of different players and the overall pressure on the lips. An evaluation of measurements made with six trombone players while playing was conducted and a questionnaire based evaluation about the influence of the sensors while playing was carried out. In addition to audio and video recordings, this approach allows real-world measurements out of lab into the daily playing and exercising environment of the musicians. The setup also provides further possibilities because it can be used in the sense of augmented instruments, allowing new and additional forms of expression, if the sensor data are used for real-time sound synthesis or effects. Further supportive applications in the field of pedagogy, practicing, physical stress avoidance and choosing the right instrument are possible.

Chair: *Gary Scavone*

Pe 1

Nonlinear vibrations of steelpans: Analysis of mode coupling in view of modal sound synthesis.

Mélodie Monteil, Cyril Touzé and Olivier Thomas

Steelpans are musical percussions made from steel barrels. During the manufacturing, the metal is stretched and bended, to produce a set of thin shells that are the differents notes of the instrument. In normal playing, each note is struck, and the sound reveals some nonlinear characteristics which give its peculiar tone to the instrument. In this paper, an experimental approach is first presented in order to show the complex dynamics existing in steelpan's vibrations. Then two models, based on typical modal interactions, are proposed to quantify these nonlinearities. Finally, one of them is observed in free oscillations simulations, in order to compare the internal resonance model to the steelpan vibrations behaviour in normal playing. The aim is to identify the important modes participating in the vibrations in view of building reduced-order models for modal sound synthesis.

Pe 2

Time-resolved interferometry and phase vocoder analysis of a Caribbean steelpan

Andrew Morrison, Daniel Zietlow and Thomas Moore

The Caribbean steelpan is one of the most recently developed tuned percussion instruments and has been the subject of much scientific study in recent years. Electronic speckle pattern interferometry (ESPI) is a useful method for characterizing the operating deflection shapes (ODS) and modes of vibration of musical instruments, and previous studies have used time averaged ESPI to characterize the ODS of resonances of the notes on steelpans. Using ESPI in conjunction with a high speed camera, capable of capturing images at rates of several thousand frames per second, allows for time-resolved examinations of transient motion of the note strike and the subsequent decay. High speed ESPI movies of note strikes of a low tenor steelpan were acquired while simultaneously recording the sound of the strike. The comparison of the time-resolved interferometry data with the analysis of the sound recordings allows for insights into the evolution of coupling between note areas.

Pe 3

The role of damping in steel pan manufacture

Claire Barlow, Soren Maloney and Jim Woodhouse

Results are presented of systematic studies of vibration damping in steel of a type, and processed by a route, relevant to Caribbean steel pans. Damping is likely to be a significant factor in the variation of sound quality between different pans. The main stages in pan manufacture are simulated in a controlled manner using sheet steel, cold-rolled to a prescribed level of thickness reduction then annealed at a chosen temperature in a laboratory furnace. Small test strips were cut from the resulting material, and tested in free-free beam bending to deduce the Young's modulus and its associated loss factor. It is shown that the steel type, the degree of cold working and the annealing temperature all have a significant influence on damping. Furthermore, for each individual specimen damping is found to decrease with rising frequency, approximately following a power law. Comparison with samples cut from a real pan show that there are further influences from the pan's geometrical details.

Pe 4

An objective approach for assessing the tuning properties of historical carillons

Vincent Debut, Miguel Carvalho and Jose Antunes

The carillons of the Mafra National Palace are undergoing a restoration project. Together, the pair of carillons represent the largest surviving 18th century carillons in Europe. To garantee the historical significance of these outstanding musical instruments, a detailed diagnosis of their current tuning state was achieved and results were analyzed with respect to historical, acoustical and musical concerns. In a first stage, we developed a suitable polyreference modal identification technique to infer the tuning status of bells from their modal parameters and we then systematically performed modal testing experiments on the historical bells of the Mafra carillons. For each carillon bell which plays a separate note of the instrument, tuning charts displaying the frequency relationships between its most important partials were obtained, as well as the modeshapes, decay times and beating frequencies between modal-doublets for every single musical partial of the bell. In a second part, since carillon bells also must be tuned very accuratly one relative to the others, the important topic of estimating the reference pitch and musical temperament of the musical instrument was addressed by developing optimization techniques. After presenting the modal identification procedure and optimal strategies devised for this work, the feasability and interests of this instrumental approach are finally illustrated for one of the Mafra carillons.

Pe 5

Experimental study of coupled drumhead vibrations using electronic speckle-pattern interferometry

Randy Worland

The coupled vibrations of a two-headed musical snare drum were investigated experimentally using electronic speckle-pattern interferometry. A dual interferometer system was used to record images of both vibrating heads simultaneously. Operating deflection shapes and frequencies are reported for the first several coupled modes, along with their relative amplitudes, orientations, and phase relations. Previously reported results for coupled drumhead modes are verified and extended to include the effects of doubly degenerate mode pairs that are split due to non-uniform tension in the drumheads. The (1,1) mode shapes are found to create four sets of coupled vibrations, with clear angular orientations and phase relations between the two heads in each case. The higher frequency (2,1) mode shapes are less strongly coupled, but do exhibit three of the four possible coupled pairs with this particular drum and tuning.

Pe 6

Numerical experiments with non-linear double membrane drums

Alberto Torin and Stefan Bilbao

Drums with two membranes are very common; snare drums, tom-toms and bass drums can be found in Western music, but there are examples in Eastern music, as well (the Indian mridangam, the Japanese taiko, etc.) These instruments can have considerable physical dimensions; bass drum heads can sometimes reach a radius of half a meter. Given the size, the low tension at which the membranes are generally tuned and the amplitude of vibrations, it is unlikely that a linear model could capture the most salient features of the sound of these instruments. Pitch glide effects and an increase of high-frequency energy have been observed at high excitation amplitudes for the bass drum and more recently for tom-toms. Similar phenomena have been observed, for example, in strings and plates, and are often related to the presence of non-linearities in the system. In this paper we present a finite difference time domain model of double membrane drums (i.e. tom-toms and bass drums) with air coupling and with non-linear terms (due to von Karman) in the equations of motion for the two membranes. Some of the computational difficulties stemming from this particular choice will be discussed. Simulation results and sound examples will be presented.

Physics-based modeling Oral

Chair: *Vesa Välimäki*

INVITED PM 1

Acoustics of pianos: Physical modeling, simulations and experiments

Antoine Chaigne

The outlines of a recently developed model of a grand piano are summarized. Using dedicated numerical methods, the main vibratory and acoustic variables of each constitutive part of the instrument (strings, bridge, soundboard, sound pressure) are simulated in the time-domain. The obtained waveforms are analyzed and compared with experimental data derived from measurements on a Steinway D grand piano. This comparison yields valuable insight into the physics of the instrument. It shows, in particular, that a nonlinear string model is necessary to account for the observed richness of piano spectra. The model is able to reproduce important features of piano sounds, such as the presence of soundboard modes in the transients, precursors and phantom partials. However, one important limitation of the model, in its present state, is that it does not account for the change of polarization observed on piano strings. Experimental observations of this phenomenon are discussed and a preliminary model for explaining the possible role of the zig-zag end condition in string polarization change is presented.

PM 2

Large scale physical modeling sound synthesis

Stefan Bilbao, Brian Hamilton, Alberto Torin, Craig Webb, Paul Graham, Alan Gray, Kostas Kavoussanakis and James Perry

Sound synthesis based on physical models of musical instruments is, ultimately, an exercise in numerical simulation. As such, for complex systems of the type seen in musical acoustics, simulation can be a computationally costly undertaking, particularly if simplifying hypotheses, such as those of traveling wave or mode decompositions are not employed. In this paper, large scale time stepping methods, such as the finite difference time domain and finite volume time domain methods are explored for a variety of systems of interest in musical acoustics, including brass instruments, percussion instruments based on thin plate and shell vibration, and also their embeddings in 3D acoustic spaces. Attention is paid here to implementation issues, particularly on parallel hardware, which is well-suited to time stepping methods operating over regular grids. Sound examples are presented.

PM 3

Coupled modes and time-domain simulations of a twelve-string guitar with a movable bridge

Miguel Marques, José Antunes and Vincent Debut

Coupling between the different vibrating sub-systems of a musical instrument is an important feature in music acoustics. It is the reason why instruments of similar families have such different and characteristic sounds. In this work, we propose a model for a twelve strings (six pairs) guitar, such that the strings are coupled with the instrument body through the moving bridge, which is the relevant component for energy transmission from the strings to the guitar body and back. In this preliminary study, the guitar body is modelled as a simple plate, strings being assumed to display planar-only vertical motions. However, the coupled equations thus obtained can be readily extended to cope with real guitar body modes and orbital string motions. After obtaining the coupled modes of the instrument, we illustrate the instrument time-domain coupled dynamics, by considering the characteristic modal frequencies typical of a Portuguese guitar. In particular we show how, when only one string alone is plucked, energy is transmitted to all other strings, causing sympathetic vibrations, which contribute to give this guitar its own characteristic sound identity.

PM 4

Modeling a vibrating string terminated against a bridge with arbitrary geometry

Dmitri Kartofelev, Anatoli Stulov, Heidi-Maria Lehtonen and Vesa Välimäki

This paper considers dynamic string motion in which the displacement is unilaterally constrained by the termination condition with an arbitrarily chosen geometry. A digital waveguide model is proposed for simulating the nonlinearity inducing interactions between the vibrating string and the contact condition at the point of string termination. The current work analyzes the resulting string motion influenced by the contact conditions with mostly flat but slightly curved geometries. The effect of a minute imperfection of the termination condition on the string vibration is investigated. It is shown that the lossless string vibrates in two distinct vibration regimes. In the beginning the string starts to interact in a nonlinear fashion with the bridge, and the resulting string motion is nonperiodic. The duration of that vibration regime depends on the geometry of the bridge. After some time of nonperiodic vibration, the string vibration settles in a periodic regime. Presented results are applicable for example in the physics-based sound synthesis of stringed musical instruments, such as the shamisen, biwa, sitar, tambura, veena or even the bray harp and the grand piano.

PM 5

Distributed piano soundboard modeling with common-pole parallel filters

Stefano Zambon

The soundboard plays a major role in defining the peculiar spectral and temporal characteristics of piano tones. Within the context of physics-based sound synthesis, it is customary to model the radiation effects of the soundboard as a common post-processing block for all the notes. In this paper, a computationally efficient technique is proposed for the simultaneous computation of multiple responses, corresponding to distributed excitation positions along the bridge. The method employs an approximation of measured impulse responses with several sets of parallel second-order resonators sharing the same poles, followed by a common FIR part. Details about experimental setup, parameter estimation and computational cost are covered and sound examples are provided.

PM 6

Simulated effects of combined control applied to an experimentally identified soundboard

Simon Benacchio, Baptiste Chomette, Adrien Mamou-Mani and François Ollivier

This paper presents an approach using combined state and derivative state active control applied to an experimentally identified model of a vibrating structure. Time simulations are made and discussed on the simplified soundboard of a string instrument to study the effects of this control.

PM 7

Sound synthesis of gongs obtained from nonlinear thin plates vibrations: Comparison between a modal approach and a finite difference scheme

Michele Ducceschi, Cyril Touzé and Stefan Bilbao

The sound of a gong is simulated through the vibrations of thin elastic plates. The dynamical equations are necessarily nonlinear, crashing and shimmering being typical nonlinear effects. In this work two methods are used to simulate the nonlinear plates: a finite difference scheme and a modal approach. The striking force is approximated to the first order by a raised cosine of varying amplitude and contact duration acting on one point of the surface. It will be seen that for linear and moderately nonlinear vibrations the modal approach is particularly appealing as it allows the implementation of a rich damping mechanism by introducing a damping coefficient for each mode. In this way, the frequency-dependent decay rates can be tuned to get a very realistic sound. However, in many cases cymbal vibrations are found in strongly nonlinear regimes, where an energy cascade through lengthscales brings energy up to high-frequency modes. Hence, the number of modes retained in the truncation becomes a crucial parameter of the simulation. In this sense the finite difference scheme is usually better suited for reproducing crash and gong-like sounds, because this scheme retains all the modes up to (almost) Nyquist.

PM 1P

A new method for the identification of the original modes of damped three-dimensional axi-symmetric structures subjected to constraining boundary conditions

Vincent Debut, Miguel Carvalho and Jose Antunes

Difficulties often enforce the need for performing modal identifications on structures subjected to support conditions different from those for which it was designed. Our work addresses the case of axisymmetric structures for which the modal properties under free boundary conditions must be identified, when available dynamical data pertains to supported conditions. It was motivated by the need to diagnose the tuning of large historical carillon bells temporarily fitted with supports at several locations of their rim. In a recent paper we proposed an inverse method for identifying the modal frequencies and transfer functions of the original unconstrained system, as well as for identifying the point mass and stiffness constraints. The method was applied to the 1D oscillations of a simple discretized ring-shaped structure. Results highlighted the robustness of the proposed technique, which is not prone to disturbing effects from modal identification and truncation errors. In this paper we extend our method to continuous 3D axisymmetric structures. Also we address the case of dissipative point constraints. After recalling dynamical formulations for structural modifications, we present our technique for identifying the unconstrained system modes. The method is illustrated on a simulated realistic axisymmetric structure.

PM 2P

A structured approach to using a rectangular brace to design a soundboard section for a desired natural frequency

Patrick Dumond and Natalie Baddour

The manufacture of acoustically consistent wooden musical instruments remains economically demanding and can lead to a great deal of material waste. To address this, the problem of design-for-frequency of braced plates is considered in this paper. The theory of inverse eigenvalue problems seeks to address the problem by creating representative system matrices directly from the desired natural frequencies of the system. The goal of this paper is to demonstrate how the generalized Cayley-Hamilton theorem can be used to find the system matrices. In particular, a simple rectangular brace-plate system is analyzed. The radial stiffness of the plate is varied in order to model variations typically found in wood which is quartersawn. The corresponding thickness of the brace required to keep the fundamental natural frequency of the brace-plate system at a desired value is then calculated with the proposed method. It is shown that the method works well for such a system and demonstrates the potential of using this technique for more complex systems, including soundboards of wooden musical instruments.

PM 3P

Computing virtual acoustics using the 3D finite difference time domain method and Kepler architecture GPUs

Craig Webb

The computation of virtual acoustics for physical modelling synthesis using the finite difference time domain is a computationally expensive process, especially at audio rates such as 44.1kHz. However, the high level of data-independence is well suited to parallel architectures such as those provided by graphics processing units. This paper describes the use of the latest Nvidia Kepler cards to accelerate the computation of three-dimensional schemes. The CUDA language and hardware architecture allow many possible approaches to computing even a basic model. Various techniques are considered, such as full tiling, iteration slicing, and the use of shared memory. A standard simulation was used to measure the performance of these different approaches. Benchmark times were compared for the latest Nvidia Tesla K20 GPU against the previous generation cards. Results show the continuing maturity of the hardware, especially in terms of data caching, which allows basic code designs to perform as well as more complex shared memory versions.

Chair: *Joe Wolfe*

MAE 1P

Music acoustics education at the Erich Thienhaus Institut in Detmold

Malte Kob

The education in Music Acoustics has been one of the main objectives of the founders of the first Institute for Tonmeister education, Erich Thienhaus. Since 1949 the Erich Thienhaus Institute (ETI) has offered a unique education that is dedicated to the technical and musical aspects of music production. Within the Bologna process the education at ETI has been structured into a Bachelor of Music, three flavors of Master programs and one Ph.D. course program. Two education programs are dedicated to music acoustics: one Master of Science and the Ph.D. program. The presentation presents the structure and content of these programs and potential links to related course programs inside and outside Europe.

MAE 2P

The Musical Acoustics Research Library (MARL): Fully digital & online

Gary Scavone and Jerry McBride

The Musical Acoustics Research Library (MARL) is a collection of research materials assembled by distinguished groups or individuals in the field of musical acoustics research. MARL was established at the Center for Computer Research in Music and Acoustics, Stanford University in the mid-1990s. A catalogue of the MARL contents was made available online and individual items were digitized and linked to the site upon request when resources allowed. In 2009, an agreement was reached between the various MARL representatives and the Stanford University Library for the transfer and digitization of the entire collection. The new MARL website is now officially online and its contents are freely available to the musical acoustics community in digital form.

MAE 3P

Activities for a course of physics of bowed instruments

Jesus A. Torres

Simple experiments, simulations and theoretical procedures are proposed as complement of a one-year course of acoustics, mainly focused in physics of the violin. Covered topics include string and body vibrations, the bridge, sound radiation, the wolf tone, and tonal quality. Majorly, explained procedures do not require expensive equipment or strong background in science. Therefore, the confluence of physics, music and fun allows interacting with scientific applications in musical instruments which could result a daunting experience by traditional methods.

MAE 4P

Teaching physics via the Web using music acoustics

Joe Wolfe, George Hatsidimitris, John Smith and John Tann

The UNSW Music Acoustics site provides a learning experience for its users, but it has also provided one for its makers. This paper describes how it was made and some of what we learned in making it. It also describes a new, larger project, called Physclips, which is being made with a consistent philosophy, in the light of our experience. We describe these ideas as well as some of the principles from the educational literature.

SMC 2013

PERC 1

Measuring the interaction between bassoon and horn players in achieving timbre blend

Sven-Amin Lembke, Scott Levine, Martha de Francisco and Stephen McAdams

Our study investigates the interactive relationship between bassoon and horn players in achieving timbre blend during musical performance. The interaction is studied in a behavioral experiment, measuring the timbral adjustments performers employ. Several timbre descriptors serve as acoustic measures, quantifying global and formant-based spectral-envelope properties. Furthermore, musicians' self-assessment of their performances is measured through behavioral ratings. The performances are investigated across four factors, i.e., room acoustics, communication directivity, musical voicing, and leading vs. accompanying roles. Findings from ANOVAs suggest that differences in role assignments and communication directivity between performers lead to timbral adjustments. These effects are more pronounced for horn than for bassoon and performer interdependencies appear to be most important for unison voicing.

PERC 2

A social network integrated game experiment to relate tapping to speed perception and explore rhythm reproduction

Guillaume Bellec, Anders Friberg, Daniel Wolff, Anders Elowsson and Tillman Weyde

During recent years, games with a purpose (GWAPs) have become increasingly popular for studying human behaviour. However, there is no standardised method for webbased game experiments so far. We present here an approach comprising an extended version of the CaSimIR social game framework for data collection, a mini-game for tempo and rhythm based on CaSimIR, and an initial analysis of the first dataset collected from this mini-game. The mini-game presented here are part of the Spot The Odd Song Out game that also contains a mini-game on music similarity. The Spot The Odd Song Out game is freely available for use on Facebook and the Web 1. We focus here on the GWAP method and on preliminary results of a tempo tapping and a rhythm tapping experiment. We relate the tapping data to perceptual ratings obtained in previous work. The results suggest that the centroid of the tapped tempo distribution offers prediction of perceived speed comparable to a tempo value by an expert. When averaging the rhythmic performances of a group of 10 players in the second experiment, the tapping frequency shows a pattern that corresponds to the time signature of the played music. The GWAP continues to run and collect data, and has proven to be a very effective approach in terms of data quantity, but our experience shows that is requires more effort to design prepare and run, compared to a traditional experiment.

PERC 3

Methods for real time harmonic excitation of acoustic signals

Sean Enderby, Zlatko Baracskai and Cham Athwal

In this paper three methods for the introduction of new harmonic content to an acoustic signal are assessed. Each method extracts the amplitude envelope of the fundamental frequency in a signal and applies it to a newly generated harmonic. In one method this is achieved in the frequency domain through use of the short time Fourier transform. The other two methods process audio in the time domain using either instantaneous amplitude and phase measurements or single side band automodulation. The results from a set of preliminary listening tests are discussed and compared against objective measurements based on psychoacoustic models. It is suggested that frequency domain processing is too inaccurate where low latency is required and a time domain approach is preferential. The two time domain approaches show similar levels of accuracy, however it is considered that extracting the amplitude envelope of harmonics other than the fundamental could increase accuracy. It is noted that the instantaneous amplitude and phase method provides more flexibility in order to achieve this.

Sensitivity to loudspeaker permutations during an eight-channel array reproduction of piano notes

Federico Fontana, Yuri De Pra and Alberto Amendola

An experiment has been conducted, in which ten pianists with different skill rated the sound realism and scene accuracy of a sequence of piano notes reproduced by a linear loudspeaker array, whose channel positions were changed during the test so to define different spatial patterns for the same sequence. Only exaggerated channel permutations produced significant downgrade of both qualities, furthermore without introducing appreciable changes of the apparent listening position. These results suggest that an accurate multi-channel reproduction of the frontal waves may not be crucial for determining the perceived quality of a digital piano.

PERC 1P

Reinforcement learning models for acquiring emotional musical modes

Tsubasa Tanaka, Hidefumi Ohmura and Kiyoshi Furukawa

Music is deeply related to emotions. The relationships between musical modes and emotions are especially strong. This has been recognized since the age of ancient Greece. However, finding a mode that represents a specific emotion well by psychological experiments is not easy because there are so many modes mathematically. To deal with this problem, we propose a method to generate modes that represent emotions with an engineering approach that uses reinforcement learning rather than a psychological approach. Since this method gradually adapts a mode to a target emotion, we can expect to obtain a desirable mode without enumerating all the possible modes one by one. However, this method needs a human evaluator who trains the mode. In consideration of reducing the burden on the evaluator, we have designed four function approximation models of the action-value function. As a result of a pilot experiment, the best model could acquire modes that represent "high" representational power of happiness, sadness and tenderness and "a little high" representational power of fear. Additionally, we propose a musicological concept "interval scale" that is derived from the second model and show a possibility of applying it to compose music.

PERC 2P

About the impact of audio quality on overall listening experience

Michael Schoeffler and Jürgen Herre

When listening to music, rating the overall listening experience takes many different aspects into account, e. g. the provided audio quality, the listener's mood, the song that is played back etc. Music that is distributed over the Internet is usually encoded into a compressed audio format. Compressed audio formats are evaluated by expert listeners who rate these audio formats according to the perceived audio quality. Much effort is put into researching techniques for encoding music by having better audio quality at lower bit rates. Nevertheless, the beneficial effect that the audio quality has on the overall listening experience is not fully known. This paper presents the results of an experiment that was carried out to examine the influence that a song and audio quality have on the overall listening experience. The 27 participants rated their personal overall listening experience of music items which were played back in different levels of audio quality. Since listeners have different preferences when rating overall listening experience, the participants were divided into two groups of listeners according to their responses: song likers and audio quality likers. For both types of listeners, the effect of the audio quality on the rating of overall listening experience is shown.

PERC 3P

Effect of timbre on melody recognition in three-voice counterpoint music

Song Hui Chon, Kevin Schwartzbach, Bennett Smith and Stephen McAdams

Timbre saliency refers to the attention-capturing quality of timbre. Can we make one musical line stand out of multiple concurrent lines using a highly salient timbre on the line? This is the question we ask in this paper using a voice recognition task in counterpoint music. Three-voice stimuli were generated using instrument timbres that were chosen following specific conditions of timbre saliency and timbre dissimilarity. A listening experiment was carried out with 36 musicians without absolute pitch. No effect of gender was found in the recognition data. Although a strong difference was observed on the middle voice from mono-timbre to multi-timbre conditions, timbre saliency and timbre dissimilarity conditions did not appear to have systematic effects on the average recognition rate as we hypothesized. This could be due to the variability in the excerpts used for certain conditions, or more fundamentally, because the context effect of each voice position might have been much bigger than the effects of timbre conditions we were trying to measure. A further discussion is presented on possible context effects.

PERC 4P

The importance of amplitude envelope: Surveying the temporal structure of sounds in perceptual research

Jessica Gillard and Michael Schutz

Our lab's research has repeatedly documented significant differences in the outcomes of perception experiments using flat (i.e. sustained) vs. percussive (i.e. decaying) tones. Some of these findings contrast with well-established theories and models, and we suspect this discrepancy stems from a traditional focus on flat tones in psychophysical research on auditory perception. To explore this issue, we surveyed 94 articles published in Attention, Perception & Psychophysics, classifying the temporal structure (i.e. amplitude envelope) of each sound using five categories: flat (i.e. sustained with abruptly ending offsets), percussive (i.e. naturally decaying offsets), click train (i.e. a series of rapid sound-bursts), other, and not specified (i.e. insufficient specification with respect to temporal structure). The use of flat tones (31%) clearly outnumbered percussive (4.5%). This under-utilization of percussive sounds is intriguing, given their ecological prevalence outside the lab. Interestingly, 55% of the tones encountered fell within the not specified category. This is not indicative of general neglect, as these articles frequently specified other details such as spectral envelope, headphone model, and model of computer/synthesizer. This suggests that temporal structure's full importance has not traditionally been recognized, and that it represents a rich area for future research and exploration.

PERC 5P

Modeling of melodic rhythm based on entropy toward creating expectation and emotion

Hidefumi Ohmura, Takuro Shibayama, Satoshi Shibuya, Tatsuji Takahashi, Kazuo Okanoya and Kiyoshi Furukawa

The act of listening to music can be regarded as a sequence of expectations about the nature of the next segment in the musical piece. While listening to music, the listener infers how the next section of a musical piece would sound based on whether or not the previous inferences were confirmed. However, if the listener's expectations continue to be satisfied, the listener will gradually want a change in the music. Therefore, the pleasant betrayal of the listener's expectations is important to evoke emotion in music. The increase and decrease of local complexities in the music structure are deeply involved in the betrayal of expectation. Nevertheless, no quantitative research has been conducted in this area of study. We already validated that entropy in sets of note pitches are closely related to the listeners' feeling of complexity. Therefore, in this paper, we propose a model that is able to generate a melodic rhythm based on entropy in sets of note values, and then we validate the suitability of the model in terms of complexities of rhythm through a psychological experiment.

PERC 6P

Design of an interactive earphone simulator and results from a perceptual experiment

PerMagnus Lindborg and Miracle Jia Yi Lim

The article outlines a psychoacoustically founded method to describe the acoustic performance of earphones in two dimensions, Spectral Shape and Stereo Image Coherence. In a test set of 14 typical earphones, these dimensions explained 66.2% of total variability in 11 acoustic features based on Bark band energy distribution. We designed an interactive Earphone Simulator software that allows smooth interpolation between measured earphones, and employed it in a controlled experiment (N=30). Results showed that the preferred 'virtual earphone' sound was different between two test conditions, silence and commuter noise, both in terms of gain level and spectral shape. We discuss possible development of the simulator design for use in perceptual research as well as in commercial applications.

PERC 7P

How predictable do we like our music? Eliciting aesthetic preferences with the melody triangle mobile app

Henrik Ekeus, Samer Abdallah, Peter W. Mcowan and Mark Plumbley

The Melody Triangle is a smartphone application for Android that lets users easily create musical patterns and textures. The user creates melodies by specifying positions within a triangle, and these positions correspond to the information theoretic properties of generated musical sequences. A model of human expectation and surprise in the perception of music, information dynamics, is used to ‘map out’ a musical generative system’s parameter space, in this case Markov chains. This enables a user to explore the possibilities afforded by Markov chains, not by directly selecting their parameters, but by specifying the subjective predictability of the output sequence. As users of the app find melodies and patterns they like, they are encouraged to press a ‘like’ button, where their setting are uploaded to our servers for analysis. Collecting the ‘liked’ settings of many users worldwide allows us to elicit trends and commonalities in aesthetic preferences with relation to the information-dynamic model of human expectation and surprise. We outline some of the relevant ideas from information dynamics and how the Melody Triangle is defined in terms of these. We then describe the Melody Triangle mobile application, and how it is being used to collect research data.

PERC 8P

A multipitch estimation algorithm based on fundamental frequencies and prime harmonics

Arturo Camacho and Iosef Kaver-Oreamuno

An algorithm named Prime-multiF0 for the estimation of multiple pitches in a signal is proposed. Unlike other algorithms that consider all harmonics of every pitch candidate, our algorithm considers only on the fundamental frequency and prime harmonics. This approach is shown to work extremely well with chords made of intervals no smaller than a minor third. A test suite was created using synthetic signals of sawtooth, square, and triangle waves; major, minor, diminished and augmented triads in fundamental and first and second inversion, and spanning a bass range of three octaves. Experimental results show that our algorithm was able to detect the correct notes (after rounding to the closest semitone) for all the sawtooth and square waves in the test set, and for 99.3% of the triangle waves, failing only on very high pitch notes.

INVITED HMI 1

Child/machine interaction in reflexive environment: The MIROR platform

Anna Rita Addessi

This paper introduces the MIROR Platform, an innovative adaptive device for music and dance education, proposed in the framework of the EU-ICT project MIROR-Musical Interaction Relying On Reflexion. In concluding the MIROR project, 3 software applications (MIROR-Impro, MIROR-Compo and MIROR-Body Gesture) and the draft version of the User's and Teacher's Guides have been accomplished. In this paper, the technological and pedagogical principles of the MIROR platform, notably the "reflexive interaction" paradigm, the 3 applications and related experiments will be introduced. Finally, the draft of the full architecture of the platform is presented.

HMI 2

3D gestural interaction with harmonic pitch space

Thomas Hedges and Andrew McPherson

This paper presents an interface allowing users to intuitively interact with harmonic pitch space through gestures in physical space. Although harmonic pitch spaces are a well-defined concept within the circles of academic musicology, they often fail to engage with non-musicians or musicians outside academia. A three-dimensional tonnetz founded on root progression theories is conceived, and a graphical representation rendered for visual feedback. Users navigate the tonnetz with two-handed gestures captured in three-dimensional space with a purpose built video colour-tracking system. Root transitions and pivot tone triads are used to navigate the tonnetz, and trigger audio feedback generated with MIDI.

HMI 3

Audio-tactile feedback in musical gesture primitives: Finger pressing

Hanna Järveläinen, Stefano Papetti, Sébastien Schiesser and Tobias Grosshauser

We present a study on the effect of auditory and vibrotactile cues in a finger-pressing task. During a training phase subjects learned three target forces, and had to reproduce them during an experiment, under different feedback conditions. Results show that audio-tactile augmentation allowed subjects to achieve memorized target forces with improved accuracy. A tabletop device capable of recording normal force and displaying vibrotactile feedback was implemented to run several experiments. This study is first in a series of planned investigations on the role of audio-haptic feedback and perception in relation to musical gestures primitives.

HMI 4

VocaRefiner: An interactive singing recording system with integration of multiple singing recordings

Tomoyasu Nakano and Masataka Goto

This paper presents a singing recording system, VocaRefiner, that enables a singer to make a better singing recording by integrating multiple recordings of a song he or she has sung repeatedly. It features a function called clickable lyrics, with which the singer can click a word in the displayed lyrics to start recording from that word. Clickable lyrics facilitate efficient multiple recordings because the singer can easily and quickly repeat recordings of a phrase until satisfied. Each of the recordings is automatically aligned to the music-synchronized lyrics for comparison by using a phonetic alignment technique. Our system also features a function, called three-element decomposition, that analyzes each recording to decompose it into three essential elements: F0, power, and spectral envelope. This enables the singer to select good elements from different recordings and use them to synthesize a better recording by taking full advantage of the singer's ability. Pitch correction and time stretching are also supported so that singers can overcome limitations in their singing skills. VocaRefiner was implemented by combining existing signal processing methods with new estimation methods for achieving high-accuracy robust F0 and group delay.

HMI 5

Multi-scale design of interactive music systems: The libTuiles experiment

David Janin, Florent Berthaut and Myriam Desainte-Catherine

The design and implementation of an interactive music system is a difficult task. It necessitates the description of complex interplays between two design layers at least: the real time synchronous layer for audio processing, and the symbolic event-based layer for interaction handling. Tiled programming is a recent proposal that aims at combining in a single metaphor: tiled signals, the distinct programmatic features that are used in these two layers. The libTuiles experiment presented in this paper is a first experimental implementation of such a new design principle.

HMI 6

Real-time notation using brainwave control

Joel Eaton and Eduardo Miranda

We present a significant extension to our work in the field of Brain-Computer Music Interfacing (BCMI) through providing brainwave control over a musical score in real time. This new approach combines measuring Electroencephalogram (EEG) data elicited via generating Steady State Visual Evoked Potentials (SSVEP), with mappings that allow a user to influence a score presented to a musician in a compositional and/or performance setting. Mind Trio is a generative BCMI composition based upon a musical game of 18th century origin. It is designed to respond to the subjective decisions of a user allowing them to affect control over elements of notation, ultimately directing parameters that can influence musical dramaturgy and expression via the brain. We present the design of this piece alongside the practicalities of using such a system on low-cost and accessible equipment. Our work further demonstrates how such an approach can be used by multiple users and musicians, and provides a sound foundation for our upcoming work involving four BCMI subjects and a string quartet.

HMI 7

Composing for cars

Adam Parkinson and Atau Tanaka

The authors report on composing a piece for RoadMusic, an interactive music project which generates and manipulates music for the passengers and driver in a car, using sensor information gathered from the surroundings and from the movements of the car. We present a literature review which brings together related works in the diverse fields of Automotive UI, musical mappings, generative music and sonification. We then describe our strategies for composing for this novel system, and the unique challenges it presented. We describe how the process of constructing mappings is an essential part of composing a piece of this nature, and we discuss the crucial role of mapping in defining RoadMusic as either a new musical instrument, a sonification system or generative music. We then consider briefly the extent to which the RoadMusic performance was as we anticipated, and the relative success of our composition strategies, along with suggestions for future adaptations when composing for such an environment.

HMI 8

Downy oak: Rendering ecophysiological processes in plants audible

Marcus Maeder and Roman Zweifel

In our research project trees: Rendering Ecophysiological Processes Audible, we are working on the acoustic recording, analysis and representation of ecophysiological processes in plants and studying the acoustic and aesthetic requirements for making them perceptible. Measurements of acoustic emissions in plants are only interpretable in relation to climatic and physiological dynamics such as microclimatic conditions, sap flow, and changes in trunk radius and water potentials within the plants—all measurement data that is not auditory per se. Therefore, our work involves sonifying ecophysiological data, on one hand, and analysing the acoustic emissions mathematically on the other. How can phenomena that are beyond our normal perception be made directly observable, creating new experiences and opening a new window on the processes of nature? The sound installation trees: Downy Oak, exhibited at swissnex in San Francisco in summer 2012, is a first approach to a spatial audio sonification and research system. Our experiments how that immediate and intuitive access to measurement data through sounds and their spatial positioning is very promising in terms of new forms of data display as well as generative art works.

HMI 9

The influence of graphical user interface design on critical listening skills

Josh Mycroft, Joshua D. Reiss and Tony Stockman

Current Digital Audio Workstations include increasingly complex visual interfaces which have been criticised for focusing user's attention on visual rather than aural modalities. This study aims to investigate whether visual interface complexity has an influence on critical listening skills. Participants with experience mixing audio on computers were given critical listening tests while manipulating Graphical User interfaces of varying complexity. Results from the study suggest that interfaces requiring the use of a scroll bar have a significant negative effect on critical listening reaction times. We conclude that the use of scrolling interfaces, by requiring users to hold information in working memory, can interfere with simultaneous critical listening tasks. These results have implications for the design of Digital Audio Workstations especially when using small displays.

HMI 10

Discrete isomorphic completeness and a unified isomorphic layout format

Brett Park and David Gerhard

A Unified Isomorphic Layout (UIL) format is presented in order to create a common specification for describing hexagonal isomorphic layouts. The UIL format provides an unambiguous description of relative pitch orientations and is easily visualized. The notion of complete and degenerate isomorphic layouts (along with a proof) is introduced to narrow down the number of valid isomorphic layouts used for exhaustive evaluations.

HMI 1P

Amarok Pikap: Interactive percussion playing automobile

Selcuk Artut

Alternative interfaces that imitate the audio-structure of authentic musical instruments are often equipped with sound generation techniques that feature physical attributes similar to those of the instruments they imitate. Surfaces that will be struck to produce sounds in percussive instrument modeling are commonly selected as distinctive surfaces such as pads or keys. In this article we will carry out a status analysis to examine to what extent a percussion-playing interface using FSR and piezo sensors can represent an authentic musical instrument, and how a new interactive musical interface may draw the interests of the public to a promotional event of an automobile campaign: Amarok Pikap. The structure that forms the design will also be subjected to a technical analysis.

HMI 2P

Full automation of real-time processes in interactive compositions: Two related examples

Javier Alejandro Garavaglia

This article analyses two interactive compositions of my own authorship: both include live instruments and a fully automated programming of the live electronics part using MAX. On the one hand, the paper analyses Intersections (memories) for clarinet in Bb, (2007/8); on the other hand, a comparison is offered, about how Confluences (Rainbows II) for flute, clarinet, violin, cello and piano (2010/12), is an amplification of the former piece with regard to not only its compositional further development, but also as a much more complex case of full automated live-electronics. The subject of full automation, including a historical perspective is explained in an article I have written in 2010 [1]. From a purely compositional perspective, both works share also a similar type of music dramaturgy due to their common something to hold on to factors (STHotF). Hence, the poietics and aesthesis of both compositions are also hereby introduced, in order to shed more light about the reasons for the full automation of their electronic parts, as these two aspects are solidly united to the electronics used and their relationship to the intended dramaturgy embedded in the two works.

HMI 3P

Mocap Toolbox - A MATLAB toolbox for computational analysis of movement data

Birgitta Burger and Petri Toiviainen

The MoCap Toolbox is a set of functions written in Matlab for analyzing and visualizing motion capture data. It is aimed at investigating music-related movement, but can be beneficial for other research areas as well. Since the toolbox code is available as open source, users can freely adapt the functions according to their needs. Users can also make use of the additional functionality that Matlab offers, such as other toolboxes, to further analyze the features extracted with the MoCap Toolbox within the same environment. This paper describes the structure of the toolbox and its data representations, and gives an introduction to the use of the toolbox for research and analysis purposes. The examples cover basic visualization and analysis approaches, such as general data handling, creating stick-figure images and animations, kinematic and kinetic analysis, and performing Principal Component Analysis (PCA) on movement data, from which a complexity-related movement feature is derived.

HMI 4P

Relationships between spectral flux, perceived rhythmic strength and the propensity to move

Birgitta Burger, Riikka Ahokas, Aaro Keipi and Petri Toiviainen

The tendency to move to music seems to be built into human nature. Previous studies have shown a relationship between movement and the degree of spectral flux in music, particularly in the lower sub-bands. In this study, listeners' perceptions of a range of frequency-restricted musical stimuli were investigated in order to find relationships between perceived musical aspects (rhythm, melody, and fluctuation) and the spectral flux in three different frequency bands. Additionally, the relationship between the perception of features in specific frequency bands and participants' desire to move was studied. Participants were presented with clips of frequency-restricted musical stimuli and answered four questions related to musical features. Both perceived strength of the rhythm and the propensity to move were found to correlate highly with low-frequency spectral flux. Additionally, a lower but still significant correlation was found between these perceived musical features and high-frequency spectral flux. This suggests that the spectral flux of both low and high frequency ranges can be utilized as a measure of perceived rhythm in music, and that the degree of spectral flux and the perceived rhythmic strength in high and low frequency bands are at least partly responsible for the extent to which listeners consciously desire to move when listening to music.

HMI 5P

Programming interactive music scores with INScore

Dominique Fober, Stéphane Letz, Yann Orlarey and Frederic Bevilacqua

INScore is an environment for the design of interactive music scores that includes an original event-based interaction system and a scripting language for associating arbitrary messages to these events. We extended the previous version by supporting scripting languages offering a great flexibility in the description of scores and in the interactions with scores. The textual format is directly derived from the OSC message format that was defined in the original INScore version. This article presents the scripting language and illustrates its ability to describe interactions based on events, while remaining in the temporal space. It also introduces the IRCAM gesture follower and how it is embedded into INScore to provide gestural interaction capabilities.

HMI 6P

Real-time event sequencing without a visual interface

Tiago F. Tavares, Adriano Monteiro, Jayme G. A. Barbedo, Romis Attux and Jônatas Manzolli

In electronic music, it is often useful to build loops from discrete events, such as playing notes or triggering digital effects. This process generally requires using a visual interface, as well as pre-defining tempo and time quantization. We present a novel digital musical instrument capable of looping events without using visual interfaces or explicit knowledge about tempo or time quantization. The instrument is built based on a prediction algorithm that detects repetitive patterns over time, allowing the construction of rhythmic layers in real-time performances. It has been used in musical performances where it showed to be adequate in contexts that allow improvisation.

HMI 7P

Melody Bounce: Mobile rhythmic interaction for children

Stefano Baldan, Stefania Serafin and Amalia De Götzen

This paper presents an audio-based game for mobile devices, designed to develop rhythmical and timing abilities in elementary-school-aged children. Developing such skills is believed to be very important for social interaction and interpersonal coordination. Moreover, increasing evidence suggests that rhythmicity has a direct influence on other cognitive abilities such as motor coordination and sustaining attention. The game makes exclusive use of motion-based input and non-verbal audio feedack, being therefore equally enjoyable by children which might speak different languages and might or might not have visual impairments. The game logic is inherently collaborative and multiplayer, in order to promote a sense of inclusion of the child among the group of players. The game design is heavily inspired by observations of children's activities in schools, which are usually characterized by strong rhythmical patterns.

HMI 8P

Plucking Buttons: An alternate soft button input method on touch screens for musical interaction

Edward Jangwon Lee and Woon Seung Yeo

This article introduces plucking buttons, an alternate method of interacting with soft buttons on touch screens that can provide more sound parameters that are expected to enhance expressiveness in digital music. Rather than pushing buttons, users are required to start and end touches inside and outside of the button, respectively, in order to activate the button. This gesture is similar to flicking (swiping) gestures on touch screens and plucking strings on musical instruments. Advantages of this button and gesture include providing extra sound parameters, preventing accidental input, and not requiring additional screen space. The largest challenge of this gesture to be used in music is the possible delay and inaccuracy of input due to relatively complex interaction, and this is tested by comparing two input types: plucking vs. pushing buttons. Test results suggest that plucking can be used, but can be efficiently used after training. Melodic musical tasks are also executed, and users were able to successfully play a simple song.

HMI 9P

Robin: An algorithmic composer for interactive scenarios

Fabio Morreale, Raul Masu and Antonella De Angeli

This paper presents Robin, an algorithmic composer specifically designed for interactive situations. Users can interface in real-time with the algorithmic compostion by means of control strategies based on emotions. This study is driven by the necessity of providing a system for automatic music generation to be used in interactive systems for music creation that target non-musicians. Robin composes original tonal music in classical piano style by following a rule-based approach. The first practical application of Robin is the Music Room: an interactive installation where acouple of people composes music by moving in a space following the analogy with emotions.

HMI 10P

x-OSC: A versatile wireless I/O device for creative/music applications

Sebastian Madgwick and Thomas Mitchell

This paper introduces x-OSC: a WiFi-based I/O board intended to provide developers of digital musical instruments with a versatile tool for interfacing software to the physical world via OSC messages. x-OSC features 32 I/O channels supporting multiple modes including: 13-bit analogue inputs, 16-bit PWM outputs and serial communication. The optimised design enables a sustained throughput of up to 370 messages per second and latency of less than 3 ms. Access to settings via a web browser prevents the need for specific drivers or software for greater cross-platform compatibility. This paper describes key aspects x-OSC's design, an evaluation of performance and three example applications.

HMI 11P

The Airsticks: A new interface for electronic percussionists

Alon Ilsar, Mark Havryliv and Andrew Johnston

This paper documents the early developments of a new interface for electronic percussionists. The interface is designed to allow the composition, improvisation and performance of live percussive electronic music using hand, finger, foot and head movements captured by various controllers. This paper provides a background to the field of electronic percussion, outlines the artistic motivations behind the project, and describes the technical nature of the work completed so far. This includes the development of software, the combination of existing controllers and senses, and various ideas of mapping movement to sounds through MIDI note and control changes.

HMI 12P

A computational method for exploring musical creativity development

Antonis Alexakis, Armen Khatchatourov, Angeliki Triantafyllaki and Christina Anagnostopoulou

The development of musical creativity using non-standard methods and techniques has been given considerable attention in the last years. However, the use of new technologies in teaching improvisation and thus development of creativity has received relatively little attention to date. The aim of this paper is two-fold: firstly to propose a way of formalising the measurement of creativity, and secondly to test whether the use of a particular interactive system built to support musical improvisational dialogues between the user and the computer (MIROR IMPRO), can develop creativity. First, based on previous research, we define a set of variables aiming at evaluating creativity, and we create a computational model to automatically calculate these variables in order to assess the development of creative abilities. Second, we assess the advancement of creativity in 8-10 year-old children, who spent six weeks interacting with MIROR-IMPRO. We used two groups of children in assessing this advancement: a group of children with no musical background (n=20) and a group of young pianists (n=10). We carried out a free improvisation test before the start and after the end of six sessions with the system. The results suggest a potential progress related to a number of these variables, which could be indicative of creativity advancement. The issue of measuring creativity is discussed in the light of these findings.

HMI 13P

The actuated guitar: A platform enabling alternative interaction methods

Jeppe Larsen, Dan Overholt and Thomas Moeslund

Playing a guitar is normally only for people with fully functional hands. In this work we investigate alternative interaction concepts to enable or re-enable people with non-functional right hands or arms to play a guitar via actuated strumming. The functionality and complexity of right hand interaction with the guitar is immense. We therefore divided the right hand techniques into three main areas: Strumming, string picking / skipping, and string muting. This paper explores the first stage, strumming. We have developed an exploratory platform called the Actuated Guitar that utilizes a normal electrical guitar, sensors to capture the rhythmic motion of alternative fully functioning limbs, such as a foot, knee or the head, and a motorized fader moving a pick back and forth across the strings. A microcontroller is utilized for processing sensor data, which allows flexible mapping of user input to the actuation of the motorized fader. Our approach employs the flexibility of a programmable digital system, allowing us to scale and map different ranges of data from various sensors to the motion of the actuator – thereby making it easier adapt to individual users.

HMI 14P

Komeda: Framework for interactive algorithmic music on embedded systems

Dariusz Jackowski and Krystian Baclawski

Application of embedded systems to music installations is limited due to absence of convenient software development tools. This is a very unfortunate situation as these systems offer a set of advantages in comparison to desktop or laptop computers. Embedded devices are small in size therefore easier to incorporate into the form of the work. These devices are effortlessly expandable with various sensors and controllers. Moreover they are affordable, which creates possibility to build networks of cooperating devices. In this paper we describe a design of Komeda - the platform for interactive algorithmic music on embedded systems. The framework consists of language based on the score-with-blanks approach, the intermediate binary representation, portable virtual machine and module system.

HMI 15P

Sound Hunter – Developing a navigational HRTF-based audio game for people with visual impairments

Sebastian W. Brieger

In this article, a framework is proposed for designing 3D-based audio-only games in which all navigation is based on perceiving the 3D-audio, as opposed to relying on other navigational aids or imagining the audio as being spatial, where additional sounds may be added later on in the development process. To test the framework, a game named Sound Hunter was developed in an iterative process together with both sighted and visually impaired participants. The results indicate that the suggested framework might be a successful guidance tool when wanting to develop faster perception-based 3D-audio games, and the learning curve for the navigation was approximately 15 minutes, after which the participants navigated with high precision. Furthermore, with only small alterations to game menus and the iPhone's accelerometer function, both older and younger visually impaired people can navigate through 3D-audio environments by using simple hand movements. Finally, the results indicate that Sound Hunter may be used to train people's spatial hearing in an entertaining way with full experimental control. Two main factors seem to affect the learning curve for adapting to a foreign HRTF during virtual interactive gaming experiences; the adaptation to the navigational controls, and the experience of front/back confusion, where control adaptation is promoted by having a strong default setting with customizable sensitivity, and the experience of front/back confusion can be greatly reduced by introducing complex distance-dependent meta-level communication in synthesized sounds.

HMI 16P

Energy harvesting power flower bell – A cybernetic sound installation driven by a dirt-battery

Josef Schauer, Winfried Ritsch and Lothar Fickert

This work describes the art-driven development of an energy harvesting system in sound installations. The used energy source is a dirt-battery. It is built by digging a piece of copper and a piece of zinc in a soil. Sound is generated when there is sufficient energy to trigger a bell. In the described sound installation, such a system looks like a flower and the bell represents its bloom. With its roots (electrodes) dug into the soil, it generates electrical energy to make sound. It is shown that this concept works. It is possible to make sound by dirt-energy. In a further step, many of such devices which are called Power Flower Bells (PFBs) should be spread in a meadow, communicating with low-power Radio Frequency (RF) technology realizing musical compositions.

HMI 17P

Artificial affective listening towards a machine learning tool for sound-based emotion therapy and control

Alexis Kirke, Eduardo Miranda and Slawomir Nasuto

We are extending our work in EEG-based emotion detection for automated expressive performances of algorithmically composed music for affective communication and induction. This new system will involve music composed and expressively performed in real time to induce specific affective states, based on the detection of affective state in a human listener. Machine learning algorithms will learn: (1) how to use EEG and other biosensors to detect the user's current emotional state; and (2) how to use algorithmic performance and composition to induce certain affective trajectories. In other words the system will attempt to adapt so that it can – in real time - turn a certain user from depressed to happy, or from stressed to relaxed, or (if they like horror movies!) from relaxed to fearful. As part of this we have developed a test-bed involving an artificial listening affective agent to examine key issues and test potential solutions. As well as giving a project overview, the prototype design and first experiments with this artificial agent are presented here.

INVITED PERF 1

Modelling emotional effects of music: Key areas of improvement

Tuomas Eerola

Modelling emotions perceived in music and induced by music has garnered increased attention during the last five years. The present paper attempts to put together observations of the areas that need attention in order to make progress in the modelling emotional effects of music. These broad areas are divided into theory, data and context, which are reviewed separately. Each area is given an overview in terms of the present state of the art and promising further avenues, and the main limitations are presented. In theory, there are discrepancies in the terminology and justifications for particular emotion models and focus. In data, reliable estimation of high-level musical concepts and data collection and evaluation routines require systematic attention. In context, which is the least developed area of modelling, the primary area of improvement is incorporating musical context (music genres) into the modelling emotions. In a broad sense, better acknowledgement of music consumption and everyday life context, such as the data provided by social media, may offer novel insights into the modelling emotional effects of music.

PERF 2

Human-computer music performance: From synchronized accompaniment to musical partner

Roger Dannenberg, Nicolas Gold, Andrew Robertson, Zeyu Jin, Octav-Emilian Sandu, Praneeth Palliyaguru, Adam Stark and Rebecca Kleinberger

Live music performance with computers has motivated many research projects in science, engineering, and the arts. In spite of decades of work, it is surprising that there is not more technology for, and a better understanding of the computer as music performer. We review the development of techniques for live music performance and outline our efforts to establish a new direction, Human-Computer Music Performance (HCMP), as a framework for a variety of coordinated studies. Our work in this area spans performance analysis, synchronization techniques, and interactive performance systems. Our goal is to enable musicians to incorporate computers into performances easily and effectively through a better understanding of requirements, new techniques, and practical, performance-worthy implementations. We conclude with directions for future work.

PERF 3

Conducting a virtual ensemble with a Kinect device

Alejandro Rosa-Pujazón, Isabel Barbancho, Lorenzo Tardón and Ana M. Barbancho

This paper presents a gesture-based interaction technique for the implementation of an orchestra conductor and a virtual ensemble, using a 3D camera-based sensor to capture user’s gestures. In particular, a human-computer interface has been developed to recognize conducting gestures using a Microsoft Kinect device. The system allows the conductor to control both the tempo in the piece played as well as the dynamics of each instrument set independently. In order to modify the tempo in the playback, a time-frequency processing-based algorithmis used. Finally, an experiment was conducted to assess user’s opinion of the system as well as experimentally confirm if the features in the system were effectively improving user experience or not.

PERF 4

Virtual conductor for string quartet practice

Raquel Baez, Ana M. Barbancho, Alejandro Rosa-Pujazon, Isabel Barbancho and Lorenzo J. Tardon

This paper presents a system that emulates an ensemble conductor for string quartets. This application has been developed as a support tool for individual and group practice, so that users of any age range can use it to further hone their skills, both for regular musicians and students alike. The virtual conductor designed can offer similar indications to those given by a real ensemble conductor to potential users regarding beat times, dynamics, etc. The application developed allows the user to rehearse his/her performance without the need of having an actual conductor present, and also gives access to additional tools to further support the learning/practice process, such as a tuner or a melody evaluator. The system developed also allows for both solo practice and group practice. A set of tests were conducted to check the usefulness of the application as a practice support tool. A group of musicians from the Chamber Orchestra of Malaga including an ensemble conductor tested the system, and reported to have found it a very useful tool within an educational environment and that it helps to address the lack of this kind of educational tools in a self-learning environment.

PERF 5

Acoustic score following to musical performance with errors and arbitrary repeats and skips for automatic accompaniment

Tomohiko Nakamura, Eita Nakamura and Shigeki Sagayama

We discuss acoustic score-following algorithms for monophonic musical performances with arbitrary repeats and skips as well as performance errors, particularly focusing on reducing the computational complexity. Repeats/skips are often made arbitrarily during musical practice, and it is desirable to deal with arbitrary repeats/skips for wide application of score following. Allowing arbitrary repeats/skips in performance models demands reducing the computational complexity for score following. We show that for certain hidden Markov models, which assume independence of transition probabilities from and to where repeats/skips are made, the computational complexity can be reduced from square order down to linear order for the number of notes. We experimentally show that the proposed algorithms work in real time with practical scores (up to about 10000 notes) and can catch up with the performances in around 3.8 s after repeats/skips.

PERF 6

A contour-based jazz walking bass generator

Rui Dias and Carlos Guedes

This paper describes a contour-based algorithm for the real-time automatic generation of jazz walking bass lines, following a given harmonic progression. A brief description of the walking bass procedure will be presented, and also a brief survey on some common implementations and techniques. This algorithm was implemented in the Max/MSP graphical programming environment.

PERF 7

LANdini: A networking utility for wireless LAN-based laptop ensembles

Jascha Narveson and Dan Trueman

Problems with OSC communication over wireless routers are summarized and the idea of a separate networking utility named LANdini is introduced. LANdini's models and current structure are explained, and data from tests is presented. Future improvements are listed.

PERF 8

Motion recurrence analysis in music performances

Euler Teixeira, Hani Yehia, Mauricio Loureiro and Marcelo Wanderley

This work presents a method to represent, segment and analyze the recurrence patterns on motion data during musical performances. Physical gestures were extracted during clarinet performances and analyzed according to gestural features, comparing different musicians, musical passages and performance styles. The gestural aspects of the performances were related to the musical structure and its expressive content, and an acoustical analysis validated the results. Results show a recurrent sequence of clarinet gestures inside a defined region of interest, shown to be a key moment in the music.

PERF 9

Observed differences in rhythm between performances of classical and jazz violin students

Enric Guaus, Oriol Saña and Quim Llimona

The aim of this paper is to present a case study that highlights some differences between violin students from the classical and jazz traditions. This work is part of a broader interdisciplinary research that studies whether classical violin students with jazz music background have more control on the tempo in their performances. Because of the artistic nature of music, it is difficult to establish a unique criteria about what this control on the tempo means. The case study here presented quantifies this by analyzing which student performances are closer to some given references (i.e. professional violinists). We focus on the rhythmic relationships of multimodal data recorded in different sessions by different students, analyzed using traditional statistical and MIR techniques. In this paper, we show the criteria for collecting data, the low level descriptors computed for different streams, and the statistical techniques used to determine the performance comparisons. Finally, we provide some tendencies showing that, for this case study, the differences between performances from students from different traditions really exist.

PERF 10

Study of the tremolo technique on the acoustic guitar: Experimental setup and preliminary results on regularity

Sérgio Freire and Lucas Nézio

This paper presents an experimental setup for the study of right hand techniques on the acoustic guitar, and describes the main features of our apparatus regarding the extraction of audio descriptors. A preliminary case study on the tremolo technique is also discussed, where four different musicians played five versions of the same musical excerpt. These versions are compared on the basis of the regularity of the rhythmic pattern, the note durations, and the uniformity of the amplitudes. The comparison results suggest a direct relationship between rhythmic regularity and the player's level of expertise. Nevertheless, this relationship does not apply to the note durations or the dynamic regularity. Finally, some concerns regarding the difficulties in listening to the discovered (ir)regularities are addressed, and some steps for further research are pointed out.

PERF 11

A preliminary computational model of immanent accent salience in tonal music

Richard Parncutt, Erica Bisesi and Anders Friberg

We describe the first stage of a two-stage semi-algorithmic approach to music performance rendering. In the first stage, we estimate the perceptual salience of immanent accents (phrasing, metrical, melodic, harmonic) in the musical score. In the second, we manipulate timing, dynamics and other performance parameters in the vicinity of immanent accents (e. g., getting slower and/or louder near an accent). Phrasing and metrical accents emerge from the hierarchical structure of phrasing and meter; their salience depends on the hierarchical levels that they demarcate, and their salience. Melodic accents follow melodic leaps; they are strongest at con-tour peaks and (to a lesser extent) valleys; and their salience depends on the leap interval and the distance of the target tone from the local mean pitch. Harmonic accents depend on local dissonance (roughness, non-harmonicity, non-diatonicity) and chord/key changes. The algorithm is under development and is being tested by comparing its predictions with music analyses, recorded performances and listener evaluations.

PERF 12

Expressive production of piano timbre: Touch and playing techniques for timbre control in piano performance

Michel Bernays and Caroline Traube

Timbre is an essential expressive parameter in piano performance. Advanced-level pianists have integrated the palette of timbres at their artistic disposal as abstract concepts and multimodal images. A correspondingly imaged vocabulary composed of various adjectival descriptors is used in discussing and designating precise timbral nuances. However, the actual means of production and control of timbral nuances at the piano are not always explicitly expressed. This study explores the precise performance parameters used in producing different timbral nuances. For this aim, four short pieces were composed. Each was performed by four pianists, who highlighted five timbral nuances most representative of the piano timbre-describing vocabulary: dry, bright, round, velvety and dark. The performances were recorded with the Bösendorfer CEUS system, a high-quality piano equipped with high-accuracy sensors and an embedded computer. Fine-grained performance features were extracted from the data collected. The features that significantly differed between different-timbre performances were identified. The performance space resulting from a principal component analysis revealed an average organization of timbral nuances along a circular arc. Thirteen essential, timbre-discriminating performance features were selected. Detailed descriptions were thus obtained for each timbral nuance, according to the fine characteristics of their production and control in piano performance.

PERF 13

Composing social interactions for an interactive-spatial performance system

Adam Parkinson and Koray Tahiroğlu

This paper describes a recent composition, No More Together, in which performers' interactions directly influence the sound of the piece. The composition provides a structure for group interactions, and is performed with the on-body and in-space components of 'PESI', an interactive spatial performance system. Our composition attempts to compose social interactions, drawing upon notions of participatory sense-making, and the idea that these interactions are best construed as emergent systems, possessing their own internal dynamics. The composition is contextualised as part of the repertoire for the PESI system, exploring embodied, social and spatial interactions in sound and music computing.

PERF 14

How do people assess computer generated expressive music performances?

Sergio Canazza, Giovanni De Poli and Antonio Rodà

Music performance has being studied since long time and several computational systems were developed for generating expressive music performances. These models are generally evaluated by comparing their predictions with actual performances, both from a quantitative and a subjective point of view, often focusing on very specific aspects of the model. However little is known about how listeners evaluate the generated performances and which are the factors influencing their judgement and appreciation. In this paper we present two experiments, conducted during two dedicated workshops, to start understanding how the audience judges the entire performances. In particular we analyzed possible different preferences and expectations of the listeners and influencing factors, such as cognitive styles.

PERF 1P

Towards computable procedures for deriving tree structures in music: Context dependency in GTTM and Schenkerian theory

Alan Marsden, Keiji Hirata and Satoshi Tojo

This paper addresses some issues arising from theories which represent musical structure in trees. The leaves of a tree represent the notes found in the score of a piece of music, while the branches represent the manner in which these notes are an elaboration of simpler underlying structures. The idea of multi-levelled elaboration is a central feature of the Generative Theory of Tonal Music (GTTM) of Lerdahl and Jackendoff, and found also in Schenkerian theory and some other theoretical accounts of musical structure. In previous work we have developed computable procedures for deriving these tree structures from scores, with limited success. In this paper we examine issues arising from these theories, and some of the reasons limiting our previous success. We concentrate in particular on the issue of context dependency, and consider strategies for dealing with this. We stress the need to be explicit about data structures and algorithms to derive those structures. We conjecture that an expectation-based parser with look-ahead is likely to be most successful.

PERF 2P

Situating the performer and the instrument in a rich social context with PESI extended system

Callum Goddard and Koray Tahiroğlu

In this paper we present our solutions to the design challenges of facilitating awareness of actions and development of self-identities within the notion of Participatory Enacting Sonic Interaction (PESI) project. The PESI system is a modular framework for participatory music making with three performers. We present a brief technical overview, design considerations and revisions resulting from a user study conducted during the system's development. Through the development process of the PESI project a design approach we term: Non-Behaviourally Restrictive Digital Technology became apparent. In this approach, the shifting focus that embodied agents have in relation to the environment is accounted for and the development of sound-action relationships is encouraged. This is achieved through providing mappings relating to individual sensor values and movement information from motion tracking data. Our approach to the implementation of the PESI system can shift the collaborative music activity to a more engaging and active experience.

PERF 3P

Refined spectral template models for score following

Filip Korzeniowski and Gerhard Widmer

Score followers often use spectral templates for notes and chords to estimate the similarity between positions in the score and the incoming audio stream. Here, we propose two methods on different modelling levels to improve the quality of these templates, and subsequently the quality of the alignment. The first method focuses on creating more informed templates for individual notes. This is achieved by estimating the template based on synthesised sounds rather than generic Gaussian mixtures, as used in current state-of-the-art systems. The second method introduces an advanced approach to aggregate individual note templates into spectral templates representing a specific score position. In contrast to score chordification, the common procedure used by score followers to deal with polyphonic scores, we use weighting functions to weight notes, observing their temporal relationships. We evaluate both methods against a dataset of classical piano music to show their positive impact on the alignment quality.

PERF 4P

A history of sequencers: Interfaces for organizing pattern-based music

Raphael Arar and Ajay Kapur

This paper presents a history of sequencers for musical performance and creation. A sequencer is a musical interface designed to record, edit and playback audio samples in pattern format for both music composition and performance. Sequencers have evolved over the years to take many forms including mechanical and analog sequencers, drum machines, software sequencers, robotic sequencers, grid-based sequencers and tangible sequencers. This vast array of sequencer types brings forth a number of technological approaches including hardware fabrication, software development, robotic design, embedded electronics and tangible interaction design.

PERF 5P

Tale following: Real-time speech recognition applied to live performance

Jean-Luc Rouas, Boris Mansencal and Joseph Larralde

This paper describes a system for tale following, that is to say speaker-independent but text-dependent speech recognition followed by automatic alignment. The aim of this system is to follow in real-time the progress of actors reading a text in order to automatically trigger audio events. The speech recognition engine used is the well known Sphinx from CMU. We used the real-time implementation pocketsphinx, based on sphinx II, with the French acoustic models developed at LIUM. Extensive testing using 21 speakers from the PFC corpus (excerpts in ``standard french'') shows that decent performances are obtained by the system - around 30% Word Error Rate (WER). However, testing using a recording during the reharsals shows that in real conditions, the performance are a bit worse : the WER is 40%. Thus, the strategy we devised for our final application includes the use of a constrained automatic alignment algorithm. The aligner is derived from a biological DNA sequences analysis algorithm. Using the whole system, the experiments report that events are triggered with an average delay of 9 s (+-8 s). The system is integrated into a widely used real-time sound processing software, Max/MSP, which is here used to triggered audio events, but could also be used to trigger other kinds of events such as lights, videos, etc.

PERF 6P

Technical report on a short live-action film whose story with soundtrack is selected in real-time based on audience arousal during performance

Alexis Kirke, Duncan Williams, Eduardo Miranda, Amanda Bluglass, Craig Whyte, Rishi Pruthi and Andrew Eccleston

‘many worlds’ is a short narrative live-action film written and directed so as to provide four optional linear routes through the plot and four endings. At two points during the fifteen minute film, decisions are made based on audience biosignals as to which plot route to take. The use of biosignals is to allow the audience to remain immersed in the film, rather than explicitly selecting plot direction, as done in most interactive films. Four audience members have a bio-signal measured, one sensor for each person: ECG (heart rate), EMG (muscle tension), EEG (“brain waves”) and Galvanic Skin Response (perspiration). The four are interpreted into a single average of emotional arousal. This is used to decide which route to select at each of the two plot selection points. The film starts with a binaural soundscape composed to relax the audience, and depending on which clip is selected at the decision points, a different soundtrack is played under the visual action as well. *‘many worlds’* is the first live action linear plotted film to be screened in a cinema in front of the general public which utilizes the above reactive approach.

PERF 7P

Improving the real-time performance of a causal audio drum transcription system

Marius Miron, Matthew E.P. Davies and Fabien Gouyon

In this paper we analyze an audio drum transcription system with respect to real-time constraints. We present an architecture which gives a better real-time response. Furthermore, we propose a novel evaluation method, which allows us to systematically explore situations which are likely to occur in real-life drum performances. Then, we evaluate the architecture on a drum loops database, and discuss the influence of the size of the evaluation window, and of the classification method. Finally, we present the implementation in Pure Data and Max MSP, and propose a "do-it-yourself" technique which allows anyone to modify, and build a drum transcription system.

PERF 8P

Creating expressive piano performance using a low-dimensional performance model

Yupeng Gu and Christopher Raphael

A model is presented for representing and generating \newline piano performance. The model has far fewer parameters than the number of notes. This model explicitly addresses one of the fundamental characteristic of music performance that different areas in a performance have very different kinds of objectives or strategies that are employed. A graphical model is introduced to represent the evolution of the discrete strategies and tempo and dynamic progression. We design interactive procedures that allow users to modify the model intuitively. An algorithm is described to estimate parameters from partial performances that represent the skeleton of the music. Experiments are presented on the two-piano version of Rhapsody in Blue by George Gershwin.

PERF 9P

Skalldans, an audiovisual improvisation framework

PerMagnus Lindborg

Skalldans is an audiovisual improvisation piece for a solo laptop performer. Sound and video syntheses are piloted with a MIDI interface, a camera, and a Wiimote; also, audiovisual streams influence each other. The present text discusses some of the hardware and software points of interest, for example, how audio and video syntheses are piloted, how the streams interact, and the camera tracking method with a linear regression stabiliser. It also touches upon the sources of inspiration for the piece

PERF 10P

Network music with Medusa: A comparison of tempo alignment in existing MIDI APIs

Flávio Schiavoni, Marcelo Queiroz and Marcelo Wanderley

In network music, latency is a common issue and can be caused by several factors. In this paper we present the MIDI network streaming with Medusa, a distributed music environment. To ease the network connection to the end user, Medusa is implemented using different MIDI APIs: Portmidi, ALSA MIDI and JACK MIDI. We present the influence of the MIDI API choice in the system latency and jitter using the Medusa implementation.

PERF 11P

mono2eN: A multi-channel autospatialisation performance system

Callum Goddard

This paper presents the mono2eN system, a multi-channel autospatialisation performance system. Developed through a practice-led research approach, the system was originally developed for a multi-channel solo acoustic bass performance. Central to the system is an autospatilisation algorithm that controls the multi-channel spatialisation parameters of a spatialised mono sound source as well as applying a magnitude freeze audio effect. The behaviour of both the spatialisation and freeze effect is dependent upon the audio content of the signal. The motivation behind the system and a technical overview of the autospatialisation algorithm is provided. Two studies are detailed, a performance case study and a user study. These were conducted to gain insight into and to convey the impressions and experience of practitioners and users of the system. Although some concerns over the audio effect triggering were raised, overall the results indicated a positive response to the system. This suggests that the mono2eN system has potential as an easy to understand multi-channel performance system that is able to spatialise any mono audio source, allowing for its use within a large number of contexts.

PERF 12P

Modeling and simulation: The *Spectral CANON for CONLON Nancarrow* by James Tenney

Charles de Paiva Santana, Jean Bresson and Moreno Andreatta

This paper presents an approach for the analysis of musical pieces, based on the notion of computer modeling. The thorough analysis of musical works allows to reproduce compositional processes and implement them in computer models, opening new perspectives for their exploration. Computer models enable the simulations and generation of variations derived from the original model. We focus on the piece Spectral CANON For CONLON Nancarrow by James Tenney in order to emphasize the specificity of this approach, and present general perspectives for computational musicology.

PERF 13P

Capacitive left hand finger and bow sensors for synchronization and rhythmical regularity analysis in string ensembles

Tobias Großhauser, Sebastian Feese and Gerhard Tröster

In this paper bow and fingerboard sensors for measurements of synchronization between musicians in group music making are introduced. They are evaluated in several performing situations from advanced musicians in a new founded string trio up to a professional, long time experienced string quartet. The small form factor of the sensors allowed to measure synchronization in musicians' daily life situations. These are a rehearsal, tuition in chamber music class, and a concert situation. Additionally, the musicians filled out a questionnaire rating their grade of preparation, the influence of the sensor while playing, and some more data in each recording session. With the sensors, different rhythmic inaccuracies in seemingly simultaneous bow and note changes between the musicians while making music together are measured and quantified. Further a possibility for sensor based rhythmical regularity measurement while playing equal notes is presented. The results of the questionnaire confirm the unobtrusiveness of the setup and the possible use of it in daily performing situations and even on stage. At the end of this paper an outlook for synchronization skills is introduced and possible impacts into the field of new music is shown.

PERF 14P

Acoustic retroreflectors for music performance monitoring

Heikki Tuominen, Jussi Rämö and Vesa Välimäki

This paper is concerned with acoustic retroreflectors, which reflect sound back towards any sound source. They are here constructed of two reflecting panels connected with hinges and placed on a hard reflecting floor. Acoustic retroreflectors can replace electroacoustic monitoring in music performance, when sufficiently large panels are placed at an appropriate distance from performers. A good distance is between about 3 and 8 m from a player, corresponding to propagation delays between approx. 20 ms and 50 ms from a player to the retroreflector and back. We have conducted acoustic measurements in an anechoic chamber using various retroreflector structures, including symmetric V-shaped and asymmetric L-shaped reflectors of two different heights with various opening angles and incident angles. Our results show that the 90° opening angle produces the strongest reflection. Surprisingly, increasing the opening angle to 100° or larger decreases the magnitude of reflection by more than 10 dB, while a smaller angle, such as 80°, mainly weakens the reflection at high frequencies. User tests with musicians indicate that acoustic retroreflectors can provide desired feedback in performance spaces in which natural reflections to the stage are missing, such as in large halls far away fro m the walls or outdoors.

PERF 15P

Mixing symbolic and audio data in computer assisted music analysis: A case study from J. Harvey's *Speakings* (2008) for orchestra and live electronics

Stéphan Schaub, Ivan Simurra and Tiago F. Tavares

Starting from a (music) analytical question arising from the study of Jonathan Harvey's *Speakings* for orchestra and electronics (2008) we propose a computer based approach in which score (symbolic) and recorded (audio) sources are considered in tandem. After extracting a set of relevant features we used machine-learning algorithms to explore how compositional and auditory dimensions articulate in defining the identity of certain sounds event appearing in the first movement of the composition and how they contribute to their similarity with events occurring in the second movement. The computer-assisted approach was used as basis for discussion on the metaphors that inspired this particular piece, but has the potential to be extended to consider other compositions in the repertoire.

PERF 16P

Brazilian challenges on network music

Julian Jaramillo Arango, Marcio Tomiyoshi, Fernando Iazzetta and Marcelo Queiroz

This paper presents an overview of research and development of Network Music in Brazil, and particularly the production of two concerts at the University of São Paulo in partnership with the Sonic Arts Research Centre in Belfast, Northern Ireland. We present technical issues encountered that were of substantial impact on the realization of rehearsals and concerts, and also discuss aesthetic issues related to composition, performance and perception on distributed environments. From these concerts we emphasize the lessons we learned and also the perspectives for future research, always from both technical and artistic points-of-view.

Sonic interaction design Oral

INVITED SID 1

Sonification and auditory displays in electronic devices

Bruce N. Walker

Sonification is the intentional use of sound to represent data. As visual displays both shrink and grow, as datasets grow in size and complexity, and as mobile data access increases, sophisticated auditory displays become crucial. Computers and devices that support audio are wide-spread, but there remains relatively little knowledge and experience among user interface designers in how to use auditory displays effectively. This paper presents a taxonomy of auditory display methods, and discusses implementation and design issues in multimodal interaction. Some examples of auditory displays developed by the author's research group, the Georgia Tech Sonification Lab, are presented.

SID 2

Controlling a sound synthesizer using timbral attributes

Antonio Pošćić and Gordan Kreković

In this paper we present the first step towards a novel approach to visual programming for sound and music applications. To make the creative process more intuitive, our concept enables musicians to use timbral attributes for controlling sound synthesis and processing. This way, musicians do not need to think in terms of signal processing, but can rely on natural descriptions instead. A special point of interest was mapping timbral attributes into synthesis parameters. We proposed a solution based on fuzzy logic which can be applied to different synthesizers. For a particular synthesizer, an audio expert can conveniently define mappings in form of IF-THEN rules. A prototype of the system was implemented in Pure Data and demonstrated with a subtractive synthesizer. A survey conducted among amateur musicians has shown that the system works according to their expectations, but descriptions based on timbral attributes are imprecise and dependent on subjective interpretation.

SID 3

A quantitative review of mappings in musical iOS applications

Thor Kell and Marcelo Wanderley

We present a quantitative review of the mappings and metaphors used across the most popular music creation iOS applications. 337 applications were examined in terms of both the metaphor they present to the user (piano, guitar, etc), and the exact nature of their mappings (pitch mapped horizontally, time mapped vertically, etc). A special focus is given to applications that do not present a well-known interaction metaphor to the user. Potential reasons for the popularity of certain metaphors are given. We suggest that this data could be used to help explore the iOS design space, and offer some examples.

SID 4

Acoustics-like dynamics in signal-based synthesis through parameter mapping

Brendan B. Gaffney and Tamara Smyth

To ideally expand a sound synthesis parameter mapping strategy is to introduce complexity and capability without sacrificing its ease of use. Following work done with dynamical systems and catastrophe theory by René Thom, Sir E.C. Zeeman and others, we are able to create a general purpose model for introducing extended behaviors, akin to the dynamics of acoustic instruments, in low complexity interfaces without adding control parameters or losing the possibility of reverting to a simple, near-linear mapping. Herein, we explore the principles of catastrophe theory, paying particular attention to the cusp model in which two input parameters yield a third and fourth describing the "catastrophic" events after which the theory is named. As acoustic systems possess several attributes of the catastrophic models, we experiment using the cusp model to enhance mapping of control parameters to FM synthesis parameters, in an attempt to give these signal-based virtual instruments the nuance and capability of their acoustic counterparts.

SID 5

Real, foley or synthetic? An evaluation of everyday walking sounds

Amalia de Götzen, Erik Sikstöm, Francesco Grani and Stefania Serafin

The aim of this paper is to evaluate whether foley sounds, real recordings or synthetic sounds can be distinguished while used to sonify a video. In particular this work focuses on walking sounds: five different scenes of a walking person were video recorded and each video was then mixed with the three different kind of sounds mentioned above. Subjects were asked to recognise and describe the action performed, to evaluate their confidence, the realism of the action and its expressiveness. Early results show that foley sounds and real sounds cannot be distinguished by the subjects. A preliminary audio-only test was performed with the sounds used in the audio-video test in order to assess the recognition rate without the visual help.

SID 1P

Urb: Urban sound analysis and storage project

José Alberto Gomes and Diogo Tudela

This paper introduces Urb, a system for automated analysis and storing of an urban soundscape. Urb complements traditional sound maps, allowing the direct access of its features at any arbitrary moment since the system's boot, thus facilitating the study of the soundscape's evolution and the differences between specific timeframes, and facilitating artistic approaches to such data. In this paper, we will describe the creative and technical aspects considered during its early development, whilst addressing its three fundamental parts: the hardware and software for capturing and transmitting audio recordings, the software for analyzing the soundscape and the management of the database.

SID 2P

Non-realtime sonification of motiongrams

Alexander Refsum Jensenius

The paper presents a non-realtime implementation of the sonomotiongram technique, a technique for the sonification of motiongrams. Motiongrams are spatiotemporal displays of motion from video recordings, based on frame-differencing and reduction of the original video recording. The sonomotiongram technique is based on turning these visual displays of motion into sound using an “inverse FFT” process. The paper presents the non-realtime application ImageSonifyer, accompanied by video examples showing the possibilities of the sonomotiongram technique for both analytic and creative applications.

SID 3P

Impulse response estimation for the auralisation of vehicle engine sounds using dual channel FFT analysis

Simon Shelley, Damian Murphy and Simon Goodwin

A method is presented to estimate the impulse response of a filter that describes the transformation in sound that takes place between a close-mic recording of a vehicle engine and the sound of the same engine at another point in or near to the vehicle. The proposed method makes use of the Dual Channel FFT Analysis technique and does not require the use of loudspeakers, computer modelling or mechanical devices. Instead, a minimum of two microphones is required and the engine itself is used as the source of sound. This is potentially useful for virtual reality applications or in sound design for computer games, where users select their virtual position at points inside or outside the vehicle. A case study is described to examine the method in practice and the results are discussed. The described method can be readily extended for surround sound applications using spatial microphone array recording techniques.

SID 4P

Image sonification using local keypoint features

Keunhyoung Luke Kim and Woon Seung Yeo

We introduce a new paradigm for image sonification based on extraction of abstract features. Unlike most image sonification examples that convert low-level raw data into sound, this method utilizes scale invariant feature transform (SIFT) for image abstraction to obtain higher-level information, thereby producing more robust results with a variety of images and visual transformations. To separate visual components from an image and enhance hierarchical information to SIFT features, the sonification also utilizes an image structure analysis algorithm. Being invariant to object-level changes such as rotating, moving, or scaling, sonified sound describe the characteristics of different polygons well. We first describe our sonification model with SIFT features, and discuss its performance.

SID 5P

Real-time hallucination sonification and simulation through user-led development of an iPad augmented reality system

Alexis Kirke, Joel Eaton and Eduardo Miranda

The simulation of visual hallucinations has multiple applications. For example in helping diagnosis, in helping patients to express themselves and reduce their sense of isolation, for medical education, and in legal proceedings for damages due to eye / brain injuries. We present a new approach to hallucination simulation, which allows real-time audio and visual expression, using an iPad. An individual can overlay their hallucinations in real-time on the iPad screen over the iPad's video camera image. The system has been developed focusing on the visual symptoms of Palinopsia, experienced by the first author, and hence has initially been user-led research. However such an approach can be utilized for other conditions and visual hallucination types. The system also allows the hallucinations to be converted into sound through visual sonification, thus providing another avenue for expression for the hallucinating individual. Furthermore, a musical performance is described which uses the system, and which has helped to raise awareness and comfort some people who have Palinopsia symptoms.

Sound processing Oral

SP 1

Spectral distortion using second-order allpass filters

Greg Surges and Tamara Smyth

This work presents a technique for detuning or applying phase distortion to specific spectral components of an arbitrary signal using a cascade of parametric second-order allpass filters. The parametric second-order allpass provides control over the position and slope of the transition region of the phase response, and this control can be used to tune a phase distortion effect to a specific frequency range. We begin by deriving the phase response of a cascade of first-order filters, which we relate to that of the parametric second-order allpass. Time-varying parameters and the time-varying phase response are derived for the second-order case, and we provide examples demonstrating the frequency-selective phase distortion effect in the context of processing of instrumental sounds.

SP 2

Multichannel control of spatial extent through sinusoidal partial modulation (SPM)

Andres Cabrera and Gary Kendall

This paper describes a new sound processing technique to control perceived spatial extent in multichannel reproduction through artificial decorrelation. The technique produces multiple decorrelated copies of a sound signal, which when played back over a multichannel system, produce a sound image that is spatially enlarged. Decorrelation is achieved through random modulation of the time-varying sinusoidal components of the original signal's spectrum extracted using a modified version of the Loris sinusoidal modeling technique. Sinusoidal partial modulation (SPM) can be applied in varying measure to both frequency and amplitude. The amount of decorrelation between channels can be controlled through adjusting the inter-channel coherency of the modulators, thus enabling control of spatial extent. The SPM algorithm has lent itself to the creation of an application simple enough for general users, which also provides complete control of all processing parameters when needed. SPM provides a new method for control of spatial extent in multichannel sound design and electroacoustic composition.

SP 3

Real time digital audio processing using Arduino

André J. Bianchi and Marcelo Queiroz

In the search for low-cost, highly available devices for real time audio processing for scientific or artistic purposes, the Arduino platform comes in as a handy alternative for a chordless, versatile audio processor. Despite the fact that Arduinos are generally used for controlling and interfacing with other devices, its built-in ADC/DAC allows for capturing and emitting raw audio signals with very specific constraints. In this work we dive into the microcontroller's structure to understand what can be done and what are the limits of the platform when working with real time digital signal processing. We evaluate the behaviour of some common DSP algorithms and expose limitations and possibilities of using the platform in this context.

SP 4

Audio interpolation and morphing via structured-sparse linear regression

Corey Kereliuk, Philippe Depalle and Philippe Pasquier

We present a method of audio interpolation suitable for the restoration of missing and/or corrupted audio samples. Our method assumes that the missing/corrupted samples can be easily identified and are subsequently treated as missing data. We then model the audio signal as a linear combination of elementary waveforms (referred to as atoms) and estimate the values of the missing samples by solving a penalized linear regression problem. A first work in this direction was recently presented using the moniker `audio inpainting' (in deference to similar work in the image processing community). We extend this avenue of research by incorporating additional continuity constraints into the problem, which leads to improved estimates of the missing data. Furthermore, we show how our method leads to a natural framework for morphing/transitioning between two sounds. Finally, we present several examples that illustrate the effectiveness of our interpolation strategy and the quality of morphing that can be attained.

SP 5

Warped Frames: Dispersive vs. non-dispersive sampling

Gianpaolo Evangelista

Conventional Time-Frequency and Time-Scale Representations are often too rigid to capture fine details of sound or musical signals. Adaptation of ideal time-frequency tilings is often desirable in order to represent the signal in terms of components that are meaningful from a physical or perceptual point of view. Remapping of the time and frequency axes by means of time and frequency warping can help achieve the desired flexibility of the representation. However, in the general case, the conjugate variable is affected as well, so that the resulting representation plane is distorted. In this paper we show methods to redress the conjugate distortion introduced by warping, both in the unsampled case of the integral Short-Time Fourier Transform and in the sampled case of generalized Gabor frames. Ultimately, the methods illustrated in this paper allow for the construction and computation of Gabor-like non-uniform time frequency representations in which the new frames are obtained from uniform Gabor frames by frequency warping both the time variable and the time index. This provides a very general design procedure based on a prescribed warping map that can be derived, e.g., from a tonal scale.

SP 6

Improved polynomial transition regions algorithm for alias-suppressed signal synthesis

Dániel Ambrits and Balázs Bank

One of the building blocks of virtual analog synthesizers is the oscillator algorithm producing simple geometric waveforms, such as saw or triangle. An important requirement for such a digital oscillator is that its spectrum is similar to that of the analog waveform, that is, the heavy aliasing that would result from a trivial modulo-counter based implementation is reduced. Until now, the computationally most efficient oscillator algorithm with reduced aliasing was the Polynomial Transition Regions (PTR) method. This paper shows that the efficiency can be increased even further by eliminating the phase offset of the PTR method. The new Efficient PTR (EPTR) algorithm produces the same output as the PTR method, while requiring roughly 30% fewer operations, making it the most efficient alias-reduced oscillator algorithm to date. In addition to presenting an EPTR sawtooth algorithm, the paper extends the differentiated parabolic wave (DPW) triangle algorithm to the case of asymmetric triangle waves, followed by an EPTR implementation. The new algorithm provides continuous transition between triangle and sawtooth signals, while still remaining computationally efficient.

SP 7

Towards a discrete electronic transmission line as a musical harmonic oscillator

Kurijn Buys and Roman Auvray

In analogy with strings and acoustic pipes as musical harmonic oscillators, a novice electronic oscillator is considered. The equivalent circuit of a discrete representation of strings and pipes, which takes the form of a discrete transmission line, is constructed with real electronic components. The proposed model includes the "equivalent series resistances", which seems to be the only relevant default for both capacitors and inductors for this application. In an analytical approach, the complex wave number is derived, allowing the study of both the wave's dispersion and attenuation in function of frequency and resulting in recommended and critical component values. Next, components are selected for a first eight-node prototype, which is numerically evaluated and then practically constructed and measured. The results prove a good match between theory and practice, with five distinguishable modes in the entrance impedance. A new prototype design is planned, which is expected to have much improved quality factors.

SP 8

Solving interactions between nonlinear resonators

Joël Bensoam and David Roze

In the context of musical acoustics, physical models of musical instruments have to be more and more sophisticated. For string models, realism is obtained by taking into account tension, flexion, shear, rotation and coupling phenomena but also nonlinear effects due to large displacements. The sound synthesis modal method is extended to the nonlinear case using Volterra series. The inverse problem of interaction between two acoustical objects is solved by finding the roots of a polynomial at each time step.

SP 9

An energy conserving finite difference scheme for simulation of collisions

Vasileios Chatziioannou and Maarten van Walstijn

Nonlinear phenomena play an essential role in the sound production process of many musical instruments. A common source of these effects is object collision, the numerical simulation of which is known to give rise to stability issues. This paper presents a method to construct numerical schemes that conserve the total energy in simulations of one-mass systems involving collisions, with no conditions imposed on any of the physical or numerical parameters. This facilitates the adaptation of numerical models to experimental data, and allows a more free parameter adjustment in sound synthesis explorations. The energy preservedness of the proposed method is tested and demonstrated though several examples, including a bouncing ball and a non-linear oscillator, and implications regarding the wider applicability are discussed.

SP 10

On finite difference schemes for the 3-D wave equation using non-cartesian grids

Brian Hamilton and Stefan Bilbao

In this paper, we investigate finite difference schemes for the 3-D wave equation using 27-point stencils on the cubic lattice, a 13-point stencil on the face-centered cubic (FCC) lattice, and a 9-point stencil on the body-centered cubic (BCC) lattice. The tiling of the wavenumber space for non-Cartesian grids is considered in order to analyse numerical dispersion. Schemes are compared for computational efficiency in terms of minimising numerical wave speed error. It is shown that the 13-point scheme on the FCC lattice is more computationally efficient than 27-point schemes on the cubic lattice when less than 8% error in the wave speed is desired.

SP 1P

Automatic tuning of the OP-1 synthesizer using a multi-objective genetic algorithm

Matthieu Macret and Philippe Pasquier

Calibrating a sound synthesizer to replicate or approximate a given target sound is a complex and time consuming task for musicians and sound designers. In the case of the OP1, a commercial synthesizer developed by Teenage Engineering, the difficulty is multiple. The OP-1 contains several synthesis engines, effects and low frequency oscillators, which make the parameters search space very large and discontinuous. Furthermore, interactions between parameters are common and the OP-1 is not fully deterministic. We address the problem of automatically calibrating the parameters of the OP-1 to approximate a given target sound. We propose and evaluate a solution to this problem using a multi-objective Non-dominated-Sorting-Genetic-Algorithm-II. We show that our approach makes it possible to handle the problem complexity, and returns a small set of presets that best approximate the target sound while covering the Pareto front of this multi-objective optimization problem.

SP 2P

An open-source framework for time-domain simulations

Clemens Geyer and Wilfried Kausel

In scientific research simulation of new or existing acoustical models is typically implemented using commercial numerical programming environments like Simulink/Matlab or expensive simulation packages like COMSOL or FLUENT. In this paper, a new version of the open-source simulation library ART (Acoustic Research Tool) is presented where time-domain simulation capabilities have now been added to existing frequency domain models. The concept allows mixing of modeling elements belonging to different levels of abstraction and it relieves the user from tricky implementation details like scheduling, data dependencies and memory allocation. Starting with an equation in the z-Domain, signals can be described recursively as a function of other current or previous signal samples and local or global simulation parameters. Alternatively signals can also be generated by specifying a certain topology of predefined elements with certain input and output ports. The library can be called from any programming environment running on Microsoft Windows or on Linux which allows it to be integrated in any application software project. The examples shown here have been written in the open-source high-level programming language Python. They can be downloaded together with the library and documentation from the project site http://artool.sourceforge.net.

SP 3P

Auralization of coupled spaces based on a diffusion equation model

Paul Luizard, Jean-Dominique Polack and Brian F.G. Katz

Auralization of room acoustics consists in audio rendering based on the sound characteristics of a virtual space. It is defined by Vorländer as “the creation of audible acoustic sceneries from computer-generated data”, as the auditory equivalent of visualization techniques. Auralization is obtained by convolving a room impulse response with an anechoic recording, adding room presence to the reverberation-free excitation signal, providing subjective immersion in the considered space. Since acoustically coupled spaces are encountered in various venues such as large stairways distributing corridors or rooms, naves and side galleries in churches, even crossing streets in dense cities, it becomes interesting to produce accurate auralization in these types of venues. Such coupled room impulse responses can be synthesized using a recently proposed sound energy decay model based on a diffusion equation and adapted to coupled spaces. This paper presents the parametric model of sound energy decay and describes the impulse response synthesis process leading to auralization of coupled spaces.

SP 4P

Warped low-order modeling of musical tones

Rémi Mignot, Heidi-Maria Lehtonen and Vesa Välimäki

Source-filter modeling of musical tones requires a filter model for the spectral envelope of the signal. Since the perceptual frequency resolution is best at low frequencies, frequency warping has been previously shown to improve spectral envelope estimation of audio signals. In this paper, considering low-order modeling for harmonic tones, we investigate the perceptual performance of three warped models which extend the filter models: Linear Prediction Coding (LPC), True-Envelope based Linear Prediction (TE-LPC), and Discrete All-Pole method (DAP). The modified TE-LPC method, called the Warped True-envelope Linear Prediction (WTLP), allows continuous control of the warping factor. Furthermore, a warped version of the DAP method, called WDAP, is introduced. The respective warped methods allow a continuous control of the warping factor, and here we are interested in the perceptual quality of the envelope estimation according to the warping factor for all methods. Results of our listening tests show that the frequency warping which best approximates the Bark scale, does not always give the best results.

SP 5P

Four-part harmonization using probabilistic models: comparison of models with and without chord nodes

Syunpei Suzuki and Tetsuro Kitahara

In this paper, we explore machine learning models for generating four-part harmonies given a soprano voice. Although there have been attempts of four-part harmonization based on machine learning, the computational models proposed in most studies ontained nodes or states representing chords or harmonic functions. Explicitly introducing such nodes or states is suitable from the viewpoint of practically achieving musically acceptable harmonization, but it would be unsuitable from the scientific viewpoint of acquiring the fundamental concept of harmonies purely by learning from actual music data. In this paper, therefore, we develop two kinds of computational models, one of which contains chord nodes and the other of which does not, and investigate to what extent the model without chord nodes acquires the fundamental concept of harmonies compared to the model with chord nodes. In our models, musical simultaneity (i.e., the appropriateness of combinations of simultaneously played notes) and musical sequentiality (i.e., the smoothness of the melodic line within each voice) are described as dependencies between random variables in Bayesian networks. Experimental results on learning 254 pieces taken from a Hymn corpus show that the Bayesian network without chord nodes acquired some basic rules in harmony.

SP 6P

A versatile toolkit for controlling dynamic stochastic synthesis

Gordan Kreković and Davor Petrinović

Dynamic stochastic synthesis is one of the non-standard sound synthesis techniques used mostly in experimental computer music. It is capable of producing various rich and organic sonorities, but its drawback is the lack of a convenient approach to controlling the synthesis parameters. Several authors previously addressed this problem and suggested direct parameter control facilitated with additional features such as parameter automation. In this paper we present a comprehensive toolkit which, besides direct control, offers several new approaches. First, it enables controlling the synthesizer with an audio signal. Relevant audio features of an input signal are mapped to the synthesis parameters making the control immediate and intuitive. Second, the toolkit supports MIDI control so that musicians can use standard MIDI interfaces to play the synthesizer. Based on this approach we implemented a polyphonic MIDI-controlled synthesizer and included it in the toolkit along with other examples of controlling the dynamic stochastic synthesizer. The toolkit was developed in the widely used visual programming environment Pure Data.

SP 7P

Visions of sound: The Centro di Sonologia Computazionale, from computer music to sound and music computing

Sergio Canazza, Giovanni De Poli and Alvise Vidolin

Centro di Sonologia Computazionale (CSC) scientific research was the premise for subsequent activities of musical informatics, and is still one of the main activities of the Centre. Today CSC activities rely on a composite group of people, which include the Center board of directors and personnel, guest researchers and musicians, and particularly on master students attending the course "Sound and Music Computing" at Dept. of Information Engineering (DEI), which is historically tightly linked to the CSC. The dissemination of scientific results as well as the relationship between art and science is hard and surely not trivial. With this aim, this paper describes an exhibition that illustrated the history of CSC, from the scientific, technological and artistic points of view. This exhibition is one of the first examples of "a museum" of Computer Music and SMC researches.

SP 8P

Smoothness under parameter changes: Derivatives and total variation

Risto Holopainen

Apart from the sounds they make, synthesis models are distinguished by how the sound is controlled by synthesis parameters. Smoothness under parameter changes is often a desirable aspect of a synthesis model. The concept of smoothness can be made more accurate by regarding the synthesis model as a function that maps points in parameter space to points in a perceptual feature space. We introduce new conceptual tools for analyzing the smoothness related to the derivative and total variation of a function and apply them to FM synthesis and an ordinary differential equation. The proposed methods can be used to find well behaved regions in parameter space.

SP 9P

Audio restoration of solo guitar excerpts using a excitation-filter instrument model

Juan Parras-Moral, Francisco Canadas Quesada, Pedro Vera Candeas and Nicolas Ruiz Reyes

This work proposes a denoising algorithm for musical instruments based on the use of an excitation-filter instrument model. Firstly, frequency patterns for the musical instrument are learned. These patterns are trained in advance from the RWC database and classified into harmonic and transient components. The harmonic patterns of the target instrument are modelled with an excitation-filter approach. Frequency patterns from the beginning of different notes (onsets) are also learned. Secondly, frequency patterns from noise are trained. Two different types of global degradations from vinyl audio (hum and hiss), apart from localized degradations from crackle noise, are used in this work. Two different types of global degradations from vinyl audio (hum and hiss), apart from localized degradations from click, crackle and scratch noise, are used in this work. Two databases (click+crackle+scratch+hiss and click+crackle+scratch+hiss+hum) are collected in order to obtain different subsets for training and testing. Finally, an NMF approach is applied to separate instrument signal and noise from noisy performances. The proposed approach is compared with some commercial algorithms when denoising a vynil degraded guitar database. The separation measures indicate that the proposed approach obtains competitive results.

SP 10P

Spatium, tools for sound spatialization

Rui Penha and João Pedro Oliveira

In this paper we present spatium, a set of free, open source and modular software tools for sound spatialization, describing the creative and technical aspects considered during its development. The system is comprised of spatialization renderers, spatialization interfaces, DAW plugins and Max objects that communicate via OSC (Open Sound Control). They aim to: facilitate the exploration of different approaches to sound spatialization, ease the integration of sound spatialization into diverse compositional workflows, smooth the transition from the studio to different performance environments and be easily expandable to cater for growing needs

SP 11P

Dynamic FM synthesis using a network of complex resonator filters

Julian Parker and Till Bovermann

There is a strong analogy between the sinusoidal operator used in FM synthesis, and the resonator filter. When implemented in a direct-form structure, a resonator filter is not suitable for use as a substitute for an FM operator, as it is not stable under centre frequency modulation. Recent, more robust resonator filter structures have made this use a possibility. In this paper we examine the properties of this structure that makes it appropriate for this application, and describe how a network of these filters can be combined to form a dynamic FM synthesis network. We discuss the possible range of sounds that can be produced by this structure, and describe its application to a performance system for improvised electroacoustic music.

SP 12P

Reconfigurable autonomous novel guitar effects (RANGE)

Duncan MacConnell, Shawn Trail, George Tzanetakis, Peter Driessen and Wyatt Page

The RANGE guitar is a minimally-invasive hyperinstrument incorporating electronic sensors and integrated digital signal processing (DSP). It introduces an open framework for autonomous music computing eschewing the use of the laptop on stage. The framework uses an embedded Linux microcomputer to provide sensor acquisition, analog-to-digital conversion (ADC) for audio input, DSP, and digital-to-analog conversion (DAC) for audio output. The DSP environment is built in Puredata (Pd). We chose Pd because it is free, widely supported, flexible, and robust. The sensors we selected can be mounted in a variety of ways without compromising traditional playing technique. Integration with a conventional guitar leverages established techniques and preserves the natural gestures of each player’s idiosyncratic performing style. The result is an easy to replicate, reconfigurable, idiomatic sensing and signal processing system for the electric guitar requiring little modification of the original instrument

INVITED MIR 1

PHENICX: Performances as highly enriched and interactive concert experiences

Emilia Gomez, Maarten Grachten, Alan Hanjalic, Jordi Janer, Sergi Jordà, Carles F. Julià, Cynthia Liem, Agustin Martorell, Markus Schedl and Gerhard Widmer

Modern digital multimedia and internet technology have radically changed the ways people find entertainment and discover new interests online, seemingly without any physical or social barriers. Such new access paradigms are in sharp contrast with the traditional means of entertainment. An illustrative example of this is live music concert performances that are largely being attended by dedicated audiences only. This paper introduces the PHENICX project, which aims at enriching traditional concert experiences by using state-of-the-art multimedia and internet technologies. The project focuses on classical music and its main goal is twofold: (a) to make live concerts appealing to potential new audience and (b) to maximize the quality of concert experience for everyone. Concerts will then become multimodal, multi-perspective and multilayer digital artifacts that can be easily explored, customized, personalized, (re)enjoyed and shared among the users. The paper presents the main scientific objectives on the project, provides a state of the art review on related research and presents the main challenges to be addressed.

MIR 2

Semi-automatic melody extraction using note position and pitch information from users

Antti Laaksonen

Automatic melody extraction from music audio has proven to be challenging. In this paper we focus on semi-automatic melody extraction, where prior information produced by the user is used in the algorithm. Our experiment shows that users - even without a musical background - are able to produce useful approximations of both the note onset times and the pitches in the melody that is being extracted. We present a dynamic programming algorithm that takes this user-generated information and uses it for melody extraction. The algorithm is based on audio samples that are built around approximate note onset times. In addition to this, approximate note pitches can be used to constrain the set of possible melodies. We compare our algorithm with a state-of-the-art melody extraction algorithm using orchestral music material. In the evaluation we use simulated note approximations that could have been produced by a user without a musical background. In this setting, the accuracy of our algorithm is remarkably better than that of the automatic algorithm.

MIR 3

Joint f0 and inharmoncity estimation using second order optimization

Henrik Hahn and Axel Röbel

A new method is presented for the joint estimation of the inharmonicity coefficient and the fundamental frequency of inharmonic instrument sounds. The proposed method iteratively uses a peak selection algorithm and a joint parameters estimation method based on nonlinear optimization. We further introduce an adapted tessitura model to evaluate our proposed method and to compare it with state-of-art techniques.

MIR 4

Large data sets & recommender systems: Feasible approach to learning music

Jamie Gabriel

One of the critical challenges in music teaching is provid ing ways for students to search easily across very large amounts of music, in order that they can build intuition and gain experience around the ways in which different music styles are comprised. This paper demonstrates how MusicXML can be used to create large music data sets that can be utilized for searching and recommendation.

MIR 5

Comparing timbre-based features for musical genre classification

Martin Hartmann, Pasi Saari, Petri Toiviainen and Olivier Lartillot

People can accurately classify music based on its style by listening to less than half a second of audio. This has motivated efforts to build accurate predictive models of musical genre based upon short-time musical descriptions. In this context, perceptually relevant features have been considered crucial but only little research has been conducted in this direction. This study compared two timbral features for supervised classification of musical genres: 1) the Mel-Frequency Cepstral Coefficients (MFCC), coming from the speech domain and widely used for music modeling purposes; and 2) the more recent Sub-band Flux (SBF) set of features which has been designed specifically for modeling human perception of polyphonic musical timbre. Differences in performance between models were found, suggesting that the SBF feature set is more appropriate for musical genre classification than the MFCC set. In addition, fluctuations at both ends of the frequency spectrum were found to be relevant for discrimination between musical genres. The results of this study give support to the use of perceptually motivated features for musical genre classification.

MIR 6

Similarity search of freesound environmental sound based on their enhanced multiscale fractal dimension

Motohiro Sunouchi and Yuzuru Tanaka

In this paper, we propose a new acoustic feature signature based on the multiscale fractal dimension extracted from sound signals for the content-based retrieval of environmental sounds such as field-recording sounds shared through Freesound. The multiscale fractal dimension derived from the fractal theory is known as a descriptor representing several features of the sound waveform. We report the basic characteristics of the enhanced multiscale fractal dimension (EMFD) extracted from each sound signal. Furthermore, we developed a similarity search system for environmental sounds using EMFD and Mel frequency cepstral coefficients 39 (MFCC39). We have compared the descriptiveness of EMFD signature and MFCC39 for the search purpose and found some competitive aspects of EMFD signature against MFCC39. These results show that EMFD signature is useful for describing the features of environmental sound and applicable to the search of large-scale sound databases.

MIR 7

Using semantic layer projection for enhancing music mood prediction with audio features

Pasi Saari, Tuomas Eerola, György Fazekas and Mark Sandler

We propose a novel technique called Semantic Layer Projection (SLP) for predicting moods expressed by music based on audio features. In SLP, the predictive models are formed by a two-stage mapping from audio features to listener ratings of mood via a semantic mood layer. SLP differs from conventional techniques that produce a direct mapping from audio features to mood ratings. In this work, large social tag data from Last.fm music service was analysed to produce a semantic layer that represents mood-related information in low number of dimensions. The method is compared to baseline techniques at predicting the expressed Valence and Arousal in 600 popular music tracks. SLP clearly outperformed the baseline techniques at predicting Valence ($R^2=0.334$ vs. 0.245), and produced roughly equivalent performance in predicting Arousal ($R^2=0.782$ vs. 0.770). The difficulty of modelling Valence was highlighted by generally lower performance compared to Arousal. The improved prediction of Valence, and the increasingly abundant sources of social tags related to digital music make SLP a highly promising technique for future developments in modelling mood in music.

MIR 8

Beat-station: A real-time rhythm annotation software

Marius Miron, Fabien Gouyon, Matthew E.P. Davies and Andre Holzapfel

This paper describes an open-source software for real-time rhythm annotation. The software integrates several modules for graphical user interface, user management across network, tap recording, audio playing, midi interfacing and threading. It is a powerful tool for conducting listening tests, but can also be used for beat annotation of music or in a game setup. The parameters of this software, including the real-time constraints, are not pre-defined in the code but can be easily changed in a settings file. Finally, the framework used allows for scalability, as it was developed in openFrameworks. The software was applied in a cross-cultural beat tapping experiment during the ISMIR 2012 conference. We discuss the experimental setup, and an analysis of the collected real-time annotations indicates some interesting differences related to the musical style of the stimuli.

MIR 9

Modelling perception of speed in music audio

Anders Elowsson and Anders Friberg

One of the major parameters in music is the overall speed of a musical performance. Speed is often associated with tempo, but other factors such as note density (onsets per second) seem to be important as well. In this study, a computational model of speed in music audio has been developed using a custom set of rhythmic features. The original audio was first separated into a harmonic part and a percussive part and onsets were extracted separately from the different layers. The characteristics of each onset were determined based on frequency content as well as salience, and these characteristics were exploited to better model the notion of speed. In a previous study 20 listeners rated the speed of 100 ringtones consisting mainly of popular songs, which had been converted from MIDI to audio. The ratings were used in a regression to evaluate the validity of the model as well as to find appropriate features. The computed audio features were able to explain about 90 % of the variability in listener ratings.

MIR 1P

Global key extraction from classical music audio recordings based on the final chord

Christof Weiss

This paper presents a novel approach to global key extraction from audio recordings, restricted to the genre Classical only. Especially in this field of music, musical key is a significant information since many works include the key in their title. Our rule-based method relies on pre-extracted chroma features and puts special emphasis on the final chord of the piece to estimate the tonic note. To determine the mode, we analyze the chroma histogram over the complete piece and estimate the underlying diatonic scale. In both steps, we apply a multiplicative procedure to obtain high error robustness. This approach helps to minimize the amount of false tonic notes which is important for further key-related tonality analyses. The algorithm is evaluated on three different datasets containing mainly 18th and 19th century music for orchestra, piano, and mixed instruments. We reach accuracies up to 97% for correct full key (correct tonic note and mode) classification and up to 100% for correct tonic note classification.

MIR 2P

PEVI: Interface for retrieving and analyzing expressive musical performances with scape plots

Shota Miki, Takashi Baba and Haruhiro Katayose

Although a variety of interfaces for music retrieval have been proposed so far, they are not always valid for retrieving classical music, a piece of which is recorded by many players. The lineup that current music retrieval systems suggest for a given musical piece is likely to be in order of sales. This is not always desired by classical music lovers, who are interested in various interpretations of a piece. In this paper, PEVI, a novel interface based on a scape plot for finding interpretations of classical music, is presented. The scape plot window, which visualizes the most similar performances of a specified scope (multiple layers) in a specified piece by using color tags, is used as the key to assigning a range of musical pieces to be referred to. Similar performances are displayed, on a different window, as their coordinates represent the similarity of two selected musical features in regard to tempo, dynamics, and delicate control within a beat. Users of PEVI are able to observe the transition of the indices of similar performances by changing the scope on the scape plot and each weight of the musical features. In this paper, the effectiveness of PEVI is discussed with an analysis of difference performances of “Romance de Amor.”

MIR 3P

Segmentation and timbre similarity in electronic dance music

Bruno Rocha, Niels Bogaards and Aline Honingh

In this paper we argue that the notion of music similarity should be expanded into sub-similarities, meaning that similarity of music has to be judged with respect to a certain context, such as melody, harmony, rhythm or timbre. We start by focusing on timbre similarity, restricted to the domain of Electronic Dance Music (EDM). We will assess the similarity of segments of music, thus we start by studying segmentation before we come to the topic of similarity. The segmentation algorithm performs well on an EDM dataset as well as on a standard MIREX dataset. Initial listening tests of the similarity model give promising results but will have to be further evaluated in future research.

MIR 4P

Melodic outline extraction method for non-note-level melody editing

Yuichi Tsuchiya and Tetsuro Kitahara

In this paper, we propose a method for extracting a melodic outline from a note sequence and a method for re-transforming the outline to a note sequence for non-note-level melody editing. There have been many systems that automatically create a melody. When the melody output by an automatic music composition system is not satisfactory, the user has to modify the melody by either re-executing the composition system or editing the melody on a MIDI sequencer. The former option, however, has the disadvantage that it is impossible to edit only part of the melody, and the latter option is difficult for non-experts, musically untrained people. To solve this problem, we propose a melody editing procedure based on a continuous curve of the melody called a melodic outline. The melodic outline is obtained by applying the Fourier transform to the pitch trajectory of the melody and extracting low-order Fourier coefficients. Once the user redraws the outline, it is transformed into a note sequence by the inverse procedure of the extraction and a hidden Markov model. Experimental results show that non-experts can edit the melody to some extent easily and satisfactorily.

MIR 5P

SoundAnchoring: Content-based exploration of music collections with anchored self-organized maps

Leandro Collares, Tiago Tavares, Joseph Feliciano, Shelley Gao, George Tzanetakis and Amy Gooch

We present a content-based music collection exploration tool based on a variation of the Self-Organizing Map (SOM) algorithm. The tool, named SoundAnchoring, displays the music collection on a 2D frame and allows users to explicitly choose the locations of some data points known as anchors. By establishing the anchors' locations, users determine where clusters containing perceptually similar pieces of music will be placed on the 2D frame. User evaluation showed that the cluster location control provided by the anchoring process improved the experience of building playlists and exploring the music collection.

MIR 6P

SmartDJ, An interactive music player for music discovery by similarity comparison

Maureen Aw, Chung Sion Lim and PerMagnus Lindborg

In this digital music era, sorting and discovery of songs is getting harder and more time consuming than before, due to the large pool of songs out there. Many music recommendation system and other similar applications in the market make use of collaborative filtering and social recommendation to suggest music to listeners. However, the problem arises when there is not enough information collected for the song, which happens mostly to new and less popular music. Other issues include missing or inaccurate metadata, the need for Internet connection, etc. We present research on acoustic features to automatically classify songs according to user-friendly and high-level concepts that indicate social contexts for music listening, and a prototype application called "SmartDJ". We aim to provide novel ways that the user can browse her/his music collection, with a player that enhances interaction via a visual feedback, personalised DJ trajectories, smooth mix transitions and so forth. SmartDJ sorts the songs based on similarity by extracting low level features, then reducing feature space dimensionality with principle component analysis (PCA) and multidimensional scaling (MDS) methods, and plotting songs in a GUI for manual or automatic browsing, where song similarity is given by Euclidian distance in a lower-dimension song space. Users are able to visualise their music library and select songs based on their similarity, or allow the system to perform automation, by selecting a list of songs based on the selection of the seed song. Users can maneuver with the high-level descriptor on the interactive interface to attain the different song space desired.

Sound analysis based on phase information that connects time and frequency

Peter Pabon and Jordy van Velthoven

This paper intends to reveal some of the properties and possibilities for sound analysis combining the Fourier and Mellin transform. First, the general transforms are defined and it is introduced how these signal and spectrum representations relate to each other. Second, a central property of Mellin-based form of the Fourier transform; its affine scaling, which leads to the concept of a joined, logarithmic time/frequency-axis, is introduced. Third, the concept of a time-frequency continuum that is perpendicular to the logarithmic time-frequency axis is introduced. Next is discussed how information guides itself through the time-frequency continuum and how components link and move together depending on their spectrum and signal characteristics. Finally, an attempt is made to connect the special features that characterize this analysis method to other signal analysis methods.

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