

Aging and the cocktail party problem: where is the problem?

Claude Alain (Rotman Research Institute, Baycrest Centre/University of Toronto)

From the cochlea to associative auditory cortex, incoming acoustic data undergo a series of transformations that allow us to build representations of the various sound sources present in the environment. Listeners' ability to perceptually organize sounds that occur sequentially and/or simultaneously is a prerequisite in solving the cocktail party problem. In this presentation, I will review studies that have examined the effects of age on auditory stream segregation as measured by the "gallop" paradigm. Specific findings related to aging and concurrent sounds organization will also be presented. More specifically, I will present evidence for an age-related decline in periodicity coding, which may account for older adults' difficulties in understanding speech in noise. Evidence from our group also show that listeners' ability to segregate concurrent sounds can be modulated by experience, providing a basis for further studies assessing the potential rehabilitative effects of training on solving complex scene analysis problems illustrated by the cocktail party example.

Tonogenesis in two Mayan languages: a contextual acoustic analysis

Francisco Arellanes Arellanes (Universidad Nacional Autónoma de México), Fidencio Briceño Chel (Instituto Nacional de Antropología e Historia), Mario E. Chávez-Peón (University of British Columbia), H. Antonio García Zúñiga (Universidad de Oriente)

This study presents an acoustic analysis of the different contexts within which tone has evolved in two Mayan languages: Yucatec and Motozintlec. On the one hand, it is well known that Yucatec is the only language in the Mayan family with contrastive tone. On the other hand, scholars have proposed that other languages in the family, such as San Bartolo Tsotsil (García de León 1971), Motozintlec (Martin 1984), and Uspantek (Can Pixabaj 2007) have an incipient tonal contrast. In these cases, the origin of the contrast is not aspiration, as in Yucatec, but rearticulated vowels, ([a/a]). It is interesting that, today, rearticulated vowels have also started to indicate tonogenesis in Yucatec. We will present phonetic evidence of the realizations described above and, particularly, compare the differences between Yucatec and Motozintlec. The results will be embedded in a general discussion and typological patterns of tonogenesis (Haudricourt 1954; Hombert, Ohala and Ewan 1979).

Analysis and classification of a vowel database

Peter F. Assmann (University of Texas at Dallas), Terrance M. Nearey (University of Alberta), Sneha Bharadwaj (University of Texas at Dallas)

To study developmental changes in the acoustic properties of vowels, we have collected a database of recordings from adults and children ranging in age from 5 through 18 years from the Dallas, Texas region. The database includes each of the 12 monophthongs and 3 diphthongs of American English, spoken in hVd words, both in isolation and in a brief carrier sentence. Across age/gender groups, we find a systematic relationship (Pearson $r = 0.8$) between the geometric mean of the formant frequencies across all of the vowels for a given speaker (a measure related to vocal tract length) and the geometric mean fundamental frequency. We will report a comparison of several alternative models of the perceptual contribution of this co-variation to vowel identification by human listeners, using statistical pattern recognition techniques to classify the vowels based on the acoustic measurements.

An efficient 3D PE Propagation model with tessellation

Melanie Austin (University of Victoria/JASCO Research Ltd), Ross Chapman (University of Victoria), David Hannay (JASCO Research Ltd)

One class of very accurate computer models that can be used to predict underwater acoustic fields is based on solutions of the parabolic form of the acoustic wave equation. Currently the most accurate parabolic equation (PE) models take into full account the influence of the three-dimensional (3D) variability of the ocean and seafloor properties through partial differential operators in both depth and azimuth. These 3D solutions are very computationally intensive, but strategic definition of the model grid can save computation time. An efficient model accurately samples the acoustic field surrounding a source without wasteful re-computation of redundant information on a grid that is excessively dense. A 3D PE model (MONM3D) has been developed that tessellates the model grid pattern to optimize the radial grid density as a function of range. The model marches the solution out in range along several radial propagation paths emanating from a source position. Since the arc-length separation between radial paths increases as does the range away from the source, a greater number of radial paths are required at longer ranges to maintain a sufficient grid density to accurately sample the field. Using a fixed number of radial propagation paths would either oversample the field close to the source, or under-sample the field at long ranges. The idea of tessellation, as implemented in MONM3D, is to allow the number of radial paths in the model grid to be updated based on range from the source. The efficiency and accuracy of the model will be presented by demonstration of a test case.

The temporal coordination of recited and spontaneous speech

Adriano V Barbosa (University of British Columbia), Barry Ross (University of British Columbia), Johanna Tan (Melbourne University), Eric Vatikiotis-Bateson (University of British Columbia)

Twenty-five years ago, the question of whether or not utterance duration could be predicted from the first pitch prominence of an utterance received a lot of attention. The idea was that, if such correspondences could be measured, they would indicate speech planning - higher initial f_0 prominence corresponding to longer utterances. In our study we have found measured correspondences between fundamental frequency (f_0) and utterance duration, and have shown that perceivers can use f_0 height to predict duration. This is interesting since the bulk of our experience is not in producing recited utterances, but in producing continuous speech for which there is no simple link between f_0 height and duration. This has led us to look for other indicators in the temporal patterning of speech signals that might account for both recited and continuous utterances. Specifically, we have developed an algorithm that computes correlations within and between signals at any temporal offset. Since head motion is known to be correlated to f_0 and to provide discourse level information, we can examine the synchronization between head motion, f_0 , and another important concomitant of prominence, RMS amplitude of the acoustics, as they evolve through time. One reason why f_0 peaks and utterance (or in this case phrase/clause) duration may not correspond in continuous speech is that phrasing is patterned over relatively short spans of speech, too short for f_0 to be physically or perceptually linked to duration. The recursive correlation algorithm treats temporal coordination as a time-varying pattern.

CSA Appendix on measurement of noise exposure from headsets

A. Behar (University of Toronto), C. Giguère (University of Ottawa), T. Kelsall (Hatch Associates Ltd.)

CSA Standard Z107.56 provides procedures for measuring employee noise exposure in the workplace. A new appendix of this standard is being developed which will cover measurement of

noise exposure from employees wearing headsets for communication. This paper reviews different methods of assessing occupational noise exposure for individuals using headsets or headphones in the workplace. Standardized methods to measure sound levels directly under the device include the use of acoustic manikins, artificial ears and real-ear procedures. An alternative calculation method is proposed that includes other determinants of exposure, such as the background noise around the worker and the attenuation of the device, as input parameters in the assessment. This method has the advantage that it facilitates the implementation of solutions or treatments to reduce exposure.

Conflict resolution in SENĆOŦEN: uvulars - /e/ sequences

Sonya Bird (University of Victoria), Janet Leonard (University of Victoria)

This paper explores conflict resolution strategies in SENĆOŦEN, a dialect of North Straits Salish spoken on the Saanich Peninsula of Vancouver Island and in the surrounding Gulf and San Juan Islands. Previous acoustic work on SENĆOŦEN has shown that the strategies used to resolve articulatory conflicts between /i/ and uvular consonants are asymmetrical (Bird & Leonard, to appear). In this paper we consider /e/, to see whether similar effects occur, and whether the /i/ ~ /e/ contrast is lost following uvular consonants. Preliminary data show that /e/ lowers to [æ] both preceding and following uvular consonants, such that the /i/ ~ /e/ contrast is indeed maintained following uvulars. The symmetrical nature of co-articulatory effects is in contrast with that previously found for /i/, suggesting that speakers employ different conflict resolution strategies even within a language (c.f. Gick and Wilson, 2006). Furthermore, the lowered /e/ has formant values in the range of [æ] (Kent & Read, 2002), contrary to Montler's (1986) auditory description of the facts. This paper contributes to the growing body of literature on the phonetics of Pacific Northwest languages, and highlights the importance of considering both auditory and instrumental analysis in phonetic description.

On the phonetics of schwa in Sliammon (M. Comox Salish): implications for the representations of Salish vowels

Susan J. Blake (University of British Columbia), Kimary Shahin (Simon Fraser University)

This paper presents the results of an acoustic study of schwa in Sliammon (M. Comox) Salish. The aim was to determine the distinction, if any, between schwa and the reduced variants of the full vowels in the language. Acoustic data show that schwa and the reduced variants of the full vowels are all distinct from each other, and that Sliammon vowel reduction is of the raising/prominence-reducing type identified by Crosswhite (2001). The acoustic properties of Sliammon crescent schwa are discussed, as are the implications of the findings for the phonological representations of Salish vowels in general.

An overview of standards and guidelines influencing the implementation of Noise Emission Declarations for Machinery

Stephen H.P. Bly (Health Canada), Ifaz Haider (Health Canada)

The promulgation of increasingly stringent occupational noise exposure standards for workers would appear to necessitate the creation of quieter workplaces. Cost effective methods include the purchase of quieter machinery and the planning of noise control at the workplace design stage. This can be facilitated by the implementation of noise emission declarations for machinery. Such declarations can also help reduce environmental noise. In Europe, these considerations have led to the promulgation of two European Union Directives requiring noise emission declarations on machinery being brought to market. European/ISO standards have been

developed to support these Directives. Canada has responded with two key voluntary standards, CSA Z107.58 and CSA Z107.10, to support the implementation of noise emission declarations for machinery. This paper provides an overview of some of the European, International and Canadian standards for machinery noise measurement, noise emission declarations, occupational exposure and environmental noise that influence the implementation of noise emission declarations in Canada. Linkages to the development of a planned Health Canada Guideline for Noise Emission Declarations for Machinery will also be discussed.

Ideal maximum noise levels for elementary school classrooms

John S. Bradley (National Research Council of Canada)

Intelligibility tests on grades 1, 3 and 6 students in their classrooms have better defined children's ability to understand speech in noise. Measurements of teacher voice levels, ambient noise levels and room acoustics conditions in the classrooms have characterized acoustical conditions in the rooms. Combining the data makes it possible to estimate the percentage of the children that experience really good acoustical conditions during typical teaching situations and further to estimate maximum acceptable ambient noise levels so that most students experience good conditions for speech communication.

Speech privacy classes for rating the speech security of meeting rooms

John S. Bradley (National Research Council of Canada), Bradford N. Gover (National Research Council of Canada)

The paper will describe a new procedure for assessing the speech security of meeting rooms. This includes consideration of the statistics of speech in meeting rooms and the measurement of sound propagation from meeting rooms to spot receiver positions outside the room. The combination of sound transmission characteristics and measured ambient noise levels outside the room are used to create Speech Privacy Classes, which can be related to the probability of speech security lapses.

The effects of room acoustics on speech privacy of meeting rooms

John S. Bradley (National Research Council of Canada), Marina Apfel (National Research Council of Canada), Bradford N. Gover (National Research Council of Canada)

Although speech privacy is often only related to signal-to-noise ratios, the temporal and spatial effects in rooms also influence the intelligibility of speech transmitted from meeting rooms. This paper will describe the results of subjective studies that have explored these effects.

Auditory scene analysis as a system

Al Bregman (McGill University)

There are a number of different phenomena, including stream segregation (as caused by a number of variables, including differences in pitch, location, spectral content and dynamics), illusory continuity, fusion and decomposition of complex sounds, and so on. These are best viewed as glimpses of a single, coherent system. The alternative view would see them as distinct, each with its own physiological basis. There are a number of arguments against this latter view. They include (1) the desire for parsimonious explanation, (2) the fact that these phenomena result from processes that serve a common function, (3) the fact that they interact in predictable ways,

(4) that they respond to many of the same variables, and (5) the heuristic value of the assumption that they are parts of a coherent function in finding the factors that may affect all of them. Recent approaches to physiological investigation, and the explanations that have been offered to explain them will be examined in the light of these ideas.

Perturbations of pitch by ejectives in Upriver Halkomelem

Jason Brown (University of British Columbia), James J. Thompson (University of British Columbia)

It is a general observation that prevocalic stop consonants can have an effect on the fundamental frequency of a following vowel. For instance, it has been demonstrated that in English, voiceless stops raise the f_0 of a following vowel relative to voiced stops. Furthermore, it has been argued that these effects can be responsible for tonogenesis in some languages. Studies have also been conducted which show the time-course of these consonantal effects on f_0 . While there are several different consonant types that have been studied in this regard (including voiced, voiceless, and aspirated stops, plain sonorants, etc.), there have been few attempts to determine the effects on f_0 that ejective consonants exhibit. This paper has two objectives: to provide phonetic documentation of an under-described and critically endangered language, and to contribute to the general theoretical discussion surrounding the release properties of ejectives. This paper explores the perturbations of f_0 that ejectives in Upriver Halkomelem (Coast Salish) have.

The acoustic correlates of the unparsed: why we need more than a strong-weak distinction

Marion Caldecott (University of British Columbia)

This paper reports results from 2 experiments on St'át'imcets (Lillooet Salish) that test the prediction that phonologically distinct domains in the Prosodic Hierarchy (Selkirk, 1984) are also acoustically distinct. In particular, when a three-way phonological distinction between syllable-types exists, a three-way acoustic distinction should also be produced by speakers, contra the traditionally assumed strong-weak dichotomy. These acoustic differences should be reflected in traditional prominence cues (F_0 , duration, intensity) (Lieberman, 1959; Fry, 1955; Fry, 1958; Klatt, 1976) as well as boundary effects (Cho, 2005). Preliminary results support this prediction. The Prosodic Hierarchy is an organization of prosodic constituents, which are considered to be the domains of phonological processes. The current model permits a three-way distinction between syllables at the Pword level: 1) stressed head of foot, 2) unstressed non-head of foot; and 3) unstressed and unfooted. The acoustic correlates of this third type of syllable are hitherto unresearched. Given a systematic mapping between prosodic domains and acoustics, it follows that this third syllable should be acoustically distinct from the other two. St'át'imcets, in which such a ternary distinction is common (Roberts and Shaw, 1994), presents an ideal case to test this prediction. The predicted distinctions are 1) a three-way, directional distinction in the magnitude of prominence cues (head>unparsed>non-head) and 2) that segments at both a foot and Pword boundary should be more peripheral on the F_1/F_2 plane than those with no foot boundary. Results support these predictions, indicating that the traditional strong-weak dichotomy must be expanded to include the possibility of a ternary distinction.

Definitely indefinites? Using acoustics as a diagnostic in St'át'imcets

Marion Caldecott (University of British Columbia), Henry Davis (University of British Columbia)

This paper presents results from an experiment which compares the acoustic characteristics of indefinites and in situ WH-words in St'át'imcets (Lillooet Salish), in order to establish if speakers distinguish these otherwise homophonous forms intonationally. Speakers allow 2 readings for the following question (Davis 2005, 2006, 2008):

- (1) swat ku=az'-en-táli ku=stám'
 who DET=buy-DIR-TOP DET=what
 "Who bought what?" and "Who bought something?"

Given that in situ WH-phrases in non-question contexts are freely construed as indefinites, the question arises as to whether the in situ WH-phrase in (1) is ambiguous between a true in situ WH-phrase and an indefinite object, or whether it is *unambiguously indefinite*. One way to distinguish between these hypotheses is via intonation, as in languages like German. Oberg (2007) found significant differences in pitch peak alignment and intonation contour between *initial* WH-phrases and non-WH-phrases in St'át'imcets. If St'át'imcets in situ WH-phrases were ambiguous, we would expect to find similar intonational differences; if not, we would expect them both to show the acoustic characteristics of indefinites. In order to examine whether St'át'imcets speakers produce an acoustic distinction between the WH-phrases for the two different readings, speakers were presented with scenarios on powerpoint slides and asked to produce appropriate WH-questions. The WH-phrases were then analysed in terms of traditional prominence correlates (maximum F0, pitch peak alignment, duration and amplitude). Preliminary results indicate that speakers make a marginally significant distinction in intensity between the two phrases, supporting the hypothesis that they are not unambiguously indefinites.

Laryngealized vowels in Quiaviní Zapotec: from phonetic realization to phonological contrast

Mario E. Chávez-Peón (University of British Columbia)

Previous cross-linguistic studies have contributed significantly with respect to the possible states of the glottis including, particularly, different degrees of constriction: tense voice, *e.g.* Chong (DiCanio 2007), creaky voice, *e.g.* Jalapa Mazatec (Blankenship 2002) and total glottal closure, *e.g.* Mundurukú (Picanco 2006). Despite this variation, according to Ladefoged and Maddieson (1996: 55) "Languages contrast modal voice with no more than one degree of laryngealized voice". Quiaviní Zapotec conflicts with the previous statement, as it distinguishes two degrees of laryngeal constriction: creaky / a̰ / and checked / a̰̰ / vowels, along with breathy / a̰ / and modal / a / phonations (Munro, Lopez et al 1999). This study compares two analyses: 1. Quiaviní Zapotec has in fact two contrastive degrees of laryngealization; 2. Quiaviní Zapotec has a single phonological degree of laryngealization with contextually derived realizations. Considering different linguistic variables, such as tone, length, syllable shape and phrasal level, as well as extra-linguistic variables, gender and age, this study presents preliminary phonetic and phonological evidence that supports the contrast between creaky and checked vowels; hence, two degrees of constricted glottis, manifested along a continuum, are proposed (*cf.* Arellanes in prep.). Strong laryngealized vowels are accompanied by a short or long glottal closure, a noticeable degree of creakiness, or both. For these vowels, some manifestation of the laryngeal feature is always preserved. Weak laryngealized vowels are realized as creaky or tense vowels and, under certain circumstances, the laryngeal feature is neutralized, preserving other cues such as tone and amplitude envelope. Implications for phonological theory and phonetics-phonology interface will be discussed.

Noise control in hydraulic system driven by swash plate pump by optimizing control unit

Molham Hasan Chikhalsouk (Concordia University), Rama B. Bhat (Concordia University)

Variable displacement pumps are often used in hydraulic systems due to their high specific power, improved static and dynamic characteristics, and their ability to produce the required flow rate according to the load demands. The control unit of the pump changes the flow rate by controlling the length of the pistons strokes, and that is adjusted by the swivelling angle of the swash plate. The current models of variable displacement pumps (swash plate pump) are equipped with double negative feedback controllers, where the inner loop controls the position of the spool of the hydraulic proportional valve and the outer loop controls the position of the swash plate. The model experiences high levels of vibration due to the long rise time, which generates power shocks on the pump, and it costs more. Another model was proposed by Khalil and Bhat to reduce the rise time and power shocks and that design implemented a single feedback with PD controller to control the swivelling angle of the swash plate. They were able to reduce the rise time to 50 msec, however, they did not evaluate the vibration and noise levels. In the present paper, a novel control strategy with single PID controller is proposed and the rise time, and noise and vibration levels are obtained experimentally, and compared with the different models. The measured vibration and noise levels using the single feedback PID controller show remarkable improvements on the hydraulic system performance and quietness over the previous two types of controllers.

FE modeling of aircraft fuselage structure treated with viscoelastic material

Esen Cintosun (Université de Sherbrooke, MTI Polyfab Inc.), Nouredine Atalla (Université de Sherbrooke), Tatjana Stecenko (MTI Polyfab Inc.)

FE analysis was used to model the response of a representative aircraft fuselage structure to random excitation. The FE model was developed to replicate a vibration test set-up. As part of the test, a shaker was used to excite the 19" by 20" Al ribbed panel structure with and without viscoelastic material treatment. A laser vibrometer was used to collect velocity measurements at 26 random locations on the surface of the Al ribbed panel. The FE parameters that were modeled included input mobility (IM), damping loss factor (DLF), and average squared velocity. The measured and calculated IM values were compared to gain confidence in the FE model. The DLF values that were obtained using the decay rate, power input and half-power bandwidth methods were presented as averages in 1/3 octave frequency bandwidths (from 100 Hz to 2000 Hz). The space and frequency averaged squared velocity was used to quantitatively categorize the viscoelastic damping materials. The effect of viscoelastic damping material coverage was also evaluated by comparing both calculations (at various coverage) and measurements (at 50% and 80% coverage). The results of this study will be utilized in optimization of viscoelastic material treatment and development of alternatives to viscoelastic material damping.

The perception of auditory continuity with interrupted masking sounds

Valter Ciocca (University of British Columbia), Julie Chang (University of British Columbia)

When gaps of silence replace portions of a continuous soft sound, the sound is perceived as interrupted. However, the perceived continuity of the interrupted sound (for example, a 1 kHz pure tone) is restored when the silent gaps are replaced by louder sounds (such as loud bursts of white noise). This phenomenon (known as auditory continuity or auditory induction) has been considered as a masking analog: the loud sound should stimulate the same peripheral units that are stimulated by the soft (interrupted) sound, as if the loud sound had actually masked a portion of the soft sound (Warren, Obusek, and Ackroff, 1972). Bregman (1990) further proposed that continuity occurs if the interrupting sound effectively masks the offset and onset that delimit the

interruption in the soft sound. Some informal observations in our laboratory showed that the perceived continuity of an interrupted pure tone (presented at 55 dBSPL) is relatively strong even though an 80-dBA interrupting noise is itself interrupted by a 40-ms gap of silence. We also observed that the two 50-ms noise bursts that flank the 40-ms gap are unable to mask an intervening 40-ms, 55-dB pure tone presented at either 0, 12 or 24 dB below the level of the interrupted pure tone. Experimental results obtained with these stimuli will be presented. The implications of the experimental findings for models of auditory event formation (see recent proposals by Nakajima, Sasaki, Kanafuka, Remij, and ten Hoopen, 2000; Remijn, Nakajima, and Tanaka, 2007) will be discussed.

The effect of the degree of acoustical distortion on lexical access by younger adults

Marco Coletta (University of Toronto at Mississauga), Marianne Pelletier (University of Toronto at Mississauga), Renee Giroux (University of Toronto at Mississauga), Huiwen Goy (University of Toronto at Mississauga), Kathy Pichora-Fuller (University of Toronto at Mississauga/Toronto Rehabilitation Institute)

The goal of this study was to investigate how the speed of processing a sentence-final word is influenced by different degrees of acoustic distortion applied to the semantic context provided by the preceding portion of the sentence. Thirty-six undergraduate students participated in each of two experimental groups. Participants listened to one of six lists consisting of 48 sentence contexts which were either congruent or incongruent with the meaning of the sentence-final word, or neutral. Half of the sentence contexts were low-pass filtered and half remained unaltered. The altered contexts for Group 1 were low-pass filtered at 1000 Hz and for Group 2 they were low-pass filtered at 1750 Hz. All target words were unaltered. Participants were instructed to listen to each sentence and decide if the sentence-final word was a real word or a non-word by pressing "yes" or "no", respectively, on a response box. An analysis of the reaction time data for both groups revealed that in the unaltered condition, but not the altered, responses were faster for congruent contexts and slower for incongruent contexts in comparison to neutral contexts. Significant main effects of context and semantic priming (congruent vs. incongruent) for both degrees of distortion emerged from the reaction time data; however, facilitation (congruent vs. neutral) was significant only when there was less distortion. The results of this study support the hypothesis that the degree of distortion influences the extent to which context contributes to lexical access.

Sound transmission loss of green roofs

Maureen Connelly (British Columbia Institute of Technology/University of British Columbia), Murray Hodgson (University of British Columbia)

Green roofs have the potential to provide excellent external/internal sound isolation due to their high mass, low stiffness and damping effect, and through surface absorption, reduce noise pollution in the community from aircraft, elevated transit systems, industrial sites and noise build-up in urban areas. This paper reviews the acoustical characteristics and the potential contributions of green roofs to the acoustical environment, investigates applicable literature and sound transmission theory and reports on new empirical findings on the transmission loss of green roofs. The existing infrastructure at the Centre for the Advancement of Green Roof Technology provided an opportunity to evaluate established green roof systems with known system variables, and reference roof systems. A diffuse to free field intensity level measurement methodology was developed to obtain the presented results and for use in a future field test facility. Green roof technologies may be optimized to increase transmission loss and ameliorate the coincidence effect. Current construction practices, driven in part by sustainable building rating programs, have led to an increased use of lightweight metal roof assemblies and a decreased use of ceilings. Green roofs can provide a higher transmission loss than the additional ceiling

element and improve transmission loss throughout the full architectural frequency range, specifically desirable in residential occupancies developed below aircraft flight paths. The field testing conducted on two 33 m² low profile extensive green roofs indicated an increase of 5 to 13 dB in transmission loss over the low and mid frequency range (50 Hz to 2000 Hz), and 2 dB to 8 dB increase in transmission loss in the higher frequency range relative to the transmission loss of a reference roof.

Is there a 'ramp archetype' of intensity in electroacoustic music?

Roger T. Dean (University of Western Sydney, australYSIS), Freya Bailes (University of Western Sydney, australYSIS)

We seek to identify statistically recurrent structures in the temporal features of a wide range of musics, beyond metrical and tonal music, which may relate to perceived affect. We observe a recurrent 'rise-fall archetype' in the dynamics (intensity changes) of pieces in an anthology of Canadian computer music. To complement David Huron's proposal of the similar 'ramp archetype', based on study of Western classical score notations, we present acoustic analysis of recorded sound files, using Praat, and with a range of time windows from 40msec (close to the minimum period in which intensity changes can be detected) to 10 sec (longer than most musical phrases). We find multiple crescendo-diminuendo pairings in which crescendi are shorter than the following diminuendo. The crescendi show more rapid intensity change than the diminuendi. In contrast to Huron's original observations, we find that crescendi and diminuendi are equally common. The temporal pattern of rise-fall in intensity we describe bears some relationship with that of auditory looming, suggesting a possible shared origin in the evolution of ecological audition. The pattern may serve to recruit and retain listener attention, and thus ambient music may lack such patterns. We suggest that such archetypal patterns are also key to perception of affect, via our FEEL interpretation (produced force-effort \longleftrightarrow perceived energy-loudness, which we investigate in separate cognitive studies). Comparative acoustic analyses of some of the classical works studied by Huron will also be presented.

The noise scales and their units

Himanshu Dehra (American Institute-Industry Forum on Energy)

The standardization of noise scales is performed by presenting their sources, their definitions, their measurement equations and their units. The interference of noise arises due to difference of power of two intensities. The intensity of power for any particle body is a function of development of various stresses. The phenomenon of acoustic resonance occurs when critical stress level, matches with the natural stress level necessary for oscillation of a particle body. The criteria for generation of acoustic resonance include waves propagated with transmission of light, sound, noise, heat, electricity, fluid and fire from a particle body. The psychological feeling of sensation and perception of noise from light, sound, heat, electricity, fluid and fire is a physiological response from the sensory organs of a standard (average) human body. The objective is to standardize the characterization of noise interference due to difference of power of two intensities, which can be due to transmission of light, sound, heat, electricity, fluid and fire into a particle body. The sources of noise, their definitions, their measurement equations and their units are presented.

Acoustic modeling in automatic speech recognition

Li Deng (Microsoft Research)

In this talk, I will address the general problem of acoustic modeling in automatic speech recognition in the context of the prevailing statistical modeling and pattern recognition frameworks. I will discuss the state of the art approaches and systems, their capabilities and limitations, and the future research directions that focus on making effective use of the knowledge gained from scientific studies on how human processes and recognizes speech.

Quantitative analysis of subphonemic flap/tap variation in NAE

Donald Derrick (University of British Columbia), Bryan Gick (University of British Columbia)

Categorical behavior in speech is useful for understanding motor planning. English flaps provide up to four categorically distinct kinematic variations. We demonstrate that there are two phonetically distinct variants of flaps, and two variants of taps in American English depending on the speaker and context. This degree of 'covert' categorical subphonemic variation has been thought to occur only in rare cases such as American English /ɹ/ (Westbury et al. 1999). M-mode ultrasound was used to measure high-speed tongue tip movements. An up-flap is produced when the tongue tip is raised from below, makes contact with the alveolar ridge and keeps moving above the ridge. A down-flap is produced when the tongue tip begins from above the alveolar ridge, makes contact and moves below the ridge. The alveolar tap is similar to the one in Spanish (Ladefoged and Maddieson 1996) where the tongue moves from below to the alveolar ridge, hits the ridge and is retracted back. A post-alveolar tap is produced when the tongue tip begins raised in the post-alveolar region, moves anterior to make alveolar contact and is retracted straight back to the post-alveolar position. Surface distinctions between up/down flaps, and alveolar/post-alveolar taps have not been previously described for any language. Explanations for variation in terms of local phonetic context and speech planning will be presented, along with a quantitative description of the variation across speakers.

Bayesian source tracking and ocean environmental inversion

Stan Dosso (University of Victoria)

This paper describes a Bayesian approach to two related inverse problems in underwater acoustics: localizing/tracking an acoustic source when ocean environmental properties are unknown, and determining environmental properties using acoustic data from an unknown (moving) source. The goal is not simply to estimate values for source and environmental parameters, but to determine parameter uncertainty distributions, quantifying the information content of the inversion. A common formulation is applied for both problems in which source parameters (location and spectrum) and environmental parameters are considered unknown random variables constrained by noisy acoustic data and by prior information on parameter values (e.g., physical limits for environmental properties) and on inter-parameter relationships (limits on horizontal and vertical source speed). Given the strong nonlinearity of the inverse problem, marginal posterior probability densities are computed numerically using efficient Markov-chain Monte Carlo importance sampling methods. Source tracking results are represented by joint marginal probability distributions over range and depth, integrated over unknown environmental parameters. The approach is illustrated with examples representing tracking a quiet submerged source and geoacoustic inversion using noise from an unknown ship-of-opportunity.

Effects of emotional content and emotional voice on speech intelligibility in younger and older adults

Kate Dupuis (University of Toronto), Kathy Pichora-Fuller (University of Toronto)

Standardized speech tests have been used extensively by both clinicians and researchers to measure speech perception and spoken language understanding in different listening conditions and in people of all ages. Stimuli are typically recorded in an artificially neutral way, devoid of any affective cues, whereas most everyday speech is produced with emotional inflection that is interpreted differently depending on age. Nevertheless, the emotional implications of the linguistic content of these stimuli have not previously been controlled for or investigated. The first purpose of the current study was to investigate how the emotional properties of valence and arousal affect how well participants are able to identify 200 words from a standardized test (Northwestern University Auditory Test No. 6; NU6) presented in varying signal-to-noise conditions. The second purpose was to create a new version of emotionally-spoken NU6 words so that the effect of emotional speech on intelligibility could be tested. A younger and an older female actor each re-recorded the 200 words from the standardized test in seven emotional tones of voice (happy, sad, neutral, angry, fearful, pleasantly surprised and disgusted). The stimuli were then presented to both younger and older listeners to determine the accuracy with which each portrayed emotion was identified. Results from both younger and older adults will be discussed and both clinical and experimental implications of this research will be addressed.

Organising by object: how auditory memory can be structured within complex scenes

Ben Dyson (Ryerson University)

Previous work has shown that visual short-term memory can be organised according to the object of origin, with participants better at retrieving multiple pieces of information from the same object relative to different objects. Participants were asked to respond to complex scenes containing multiple auditory objects in order to assess the extent to which similar memorial processes operate in audition. In Experiment 1, a benefit for recalling multiple attributes of the same object was revealed for two multi-dimensional sounds presented simultaneously in time and space. Experiment 2 showed that while preparing participants' memory accrued a processing benefit, the same-object advantage could not be abolished. Experiment 3 revealed that spatial differences between sounds failed to accentuate the same-object advantage. Experiment 4 exposed the limits of auditory object-based memory by introducing temporal delays. Experiment 5 ruled out a simple association account of the data. Identifying modality-independent organisational principles of memory such as object-based coding suggest mechanisms by which the human processing system may organise and remember the multi-modal phenomenology of everyday life.

Array element localization of a bottom-mounted hydrophone array using ship noise

Gordon R. Ebbeson (Defence Research and Development Canada - Atlantic), Laetitia Thierion (École Nationale Supérieure d'Ingénieurs)

The Rapidly Deployable Systems (RDS) Technology Demonstration Project (TDP) was a major research effort that was conducted by Defence Research and Development Canada (DRDC) - Atlantic. The purpose of this project was to develop an acoustic array system that could be deployed in minutes. However, for this system to be functional, the locations of the deployed hydrophones must be known with considerable accuracy. Sensor localization is done using a technique referred to as Array Element Localization (AEL). The AEL process consists of estimating the three-dimensional positions for the hydrophones of the array. It is based on the linearized inversion of the measured arrival time data from a series of controlled impulse sources

activated in a pattern around the array. In addition, it uses the method of regularization to include *a priori* information such as the approximate positions of the sources and hydrophones. Traditionally, imploding light bulbs were used as the sources. Unfortunately, the deployment of a light-bulb field around an array can take in excess of an hour and is not operationally relevant. Recently, researchers at the University of Victoria in BC (Morley *et al*) have been investigating the use of ship noise as a source of broadband energy for AEL. Using their method, they successfully localized the 16 elements of a bottom-moored vertical array. Encouraged by these results, we used ship noise to carry out AEL on two RDS bottom-mounted horizontal arrays that were previously localized using light-bulb noise. In this paper, we present the results of that analysis and describe our analysis technique.

Cell expansion genes expression by therapeutic ultrasound. Pros and cons.

Tarek El-Bialy (University of Alberta), Taghreed Aldosary (University of Alberta), Jie Chen (University of Alberta), Ying Tsui (University of Alberta)

The biological effect of therapeutic ultrasound on cell proliferation is controversial. Nucleostemin is essential for the proliferation and survival of stem and cancer cells. We have studied the effect of therapeutic ultrasound on gene expression by different cell types. Cell tested were bone marrow stem cells, and Human umbilical Perivascular stem cells. Daily use of therapeutic ultrasound upregulated Nucleostemin in all types of stem cells used in this study. Our data suggests that Therapeutic ultrasound might be beneficial in enhancing proliferation of stem cells and maintaining their pluripotent characteristic, however this might be a negative effect on neoplastic cells. It also should be recommended that caution must be taken before application of therapeutic application should cancerous cells are expected to be in the vicinity of the application area until further investigation might shows it is safe in such a condition.

In-vivo ultrasound assisted tissue engineered articular condyle

Tarek El-Bialy (University of Alberta), Hasan Uludag (University of Alberta), Walied Mousa (University of Alberta), Saranjeev Laih (Renew Oral and Maxillofacial Surgery), Nadr Jomha (University of Alberta), Stephen Badylak (McGowan Institute for Regenerative Medicine)

The aim of this pilot study was to evaluate the effect of therapeutic ultrasound on enhancing tissue engineering articular condyle in vivo. Eight skeletally mature rabbits were selected and divided into three groups. Group 1 [3 rabbits] (Ultrasound with tissue engineered mandibular condyle), group 2 [3 rabbits] (Ultrasound with empty scaffold), and group 3 [2 rabbits] (empty scaffold, no ultrasound). All rabbits had bone marrow stem cells (BMSCs) isolated from their femur bone. For the first group, BMSCs were differentiated into either chondrogenic or osteogenic cells. The chondrogenic and osteogenic differentiated cells were seeded into collagen sponges that were housed into a biodegradable scaffolds to form tissue engineered articular condyle. The tissue engineered condyles were implanted into the amputated temporomandibular joint (TMJ) articular condyle. In group 2 and 3, the amputated TMJ articular condyles were replaced by empty scaffolds. Groups 1 and 2 were treated daily for twenty minutes by an ultrasound device that delivers a power of 30 mW/cm² with pulse frequency of 1.5 M Hz, pulse repetition frequency of 1 K Hz. Four weeks after implantation of the tissue engineered articular condyles or empty scaffolds, rabbits were euthanized and evaluated by microCT scanning as well as by histological examination. The results showed that ultrasound treated tissue engineered articular showed enhanced matrix production and integration of the chondrogenic and osteogenic differentiated cells than the untreated condyles. The conclusion is that therapeutic ultrasound enhances tissue engineered condyles in vitro and in vivo.

Finite element model of a SAW sensor for hydrogen detection

Mohamed M. EL Gowini (University of Alberta), Walied A. Moussa (University of Alberta)

Surface Acoustic Wave (SAW) sensors are widely used in sensing applications. Confinement of the wave near the surface makes it vulnerable to changes in the adjacent environment. In this study a three dimensional finite element model of a SAW sensor is developed and the model is used in studying the response due to Hydrogen absorption. A thin Palladium film is placed on the surface of the sensor and different concentrations of Hydrogen in Palladium are tested. The corresponding changes in the wave velocity are recorded. Results indicate that the reduction in wave velocity is proportional to the concentration of Hydrogen in Palladium up to a concentration of 0.3 atomic fraction (a.f). In the concentration range 0.3-0.5a.f the change in wave velocity is almost constant. The results of the wave velocity are justified by the deterioration of the mechanical (modulus of elasticity) and physical (density) properties of the Palladium film.

The effect of informational masking and word position on sentence recall

P. Ezzatian (University of Toronto at Mississauga), L. Li (University of Toronto at Mississauga/Peking University), M. K. Pichora-Fuller (University of Toronto at Mississauga), B. Schneider (University of Toronto at Mississauga)

Speech maskers can reduce the audibility of target speech (energetic masking), as well as create interference due to linguistic and acoustic similarities to target speech (informational masking). Stream segregation may take time to develop in situations where informational masking is prevalent. In addition, age-related changes in the auditory system may interfere with the buildup of auditory streaming in such situations. The present study examined the influence of informational masking on auditory stream segregation in normal-hearing younger and older adults. In Experiment 1, nonsense sentences with 3 keywords were presented against a background of speech-spectrum noise or two-talker nonsense speech. With the speech masker, accuracy increased with word position in a linear fashion. However this was not the case with the noise masker. In Experiment 2, the speech-masker was noise-vocoded to preserve envelope information while disrupting fine structure cues and minimizing semantic content. In this case, performance did not improve systematically with word position. Furthermore, while younger adults outperformed older adults in both experiments, the three-way interaction of age, type of masker, and word position was not significant in either experiment. These results suggest that stream segregation in an informational masking situation improves over time, and that this improvement is equivalent in younger and older adults.

Modelling high frequency acoustic backscatter response from non-nucleated biological specimens

Omar Falou (Ryerson University), J. Carl Kumaradas (Ryerson University), Michael C. Kolios (Ryerson University)

It has been shown that for cells with a nucleus to cell volume ratio of 0.50, the high frequency (10-65 MHz) ultrasonic backscatter response cannot be modelled well neither as a fluid nor an elastic sphere. It was hypothesized that the cytoplasm of such cells is of fluid nature whereas the nucleus possesses elastic properties that gave rise to this discrepancy. This work attempts to confirm the fluid nature of non-nucleated biological specimens by measuring the ultrasonic backscatter response from individual sea urchin oocytes and comparing it to theoretical predictions. A sparse suspension of sea urchin oocytes was prepared in artificial seawater at room temperature. Data acquisition was performed using a VisualSonics VS40B (VisualSonics Inc., Canada) ultrasound imaging device. The oocytes were imaged at 19, 40, and 55 MHz to get information for a wide band of frequencies (10-62 MHz). The ultrasonic backscatter responses vs.

frequency were plotted and compared with theoretical predictions of fluid sphere scattering model. A very good agreement was found between the experimental and theoretical results in both time and frequency domains suggesting that the non-nucleated oocytes are of fluid nature. Finally, the implications of these results on the prediction of ultrasound backscatter from cells and on ultrasound tissue characterization techniques are discussed.

A formant frequency estimator for noisy speech based on correlation and cepstrum

S. A. Fattah (Concordia University), W.-P. Zhu (Concordia University), M. O. Ahmad (Concordia University)

Formant frequency estimation of speech signals plays a fundamental role in speech analysis/synthesis, recognition, coding, and compression. Formant estimation from noise-corrupted speech is difficult but essential as far as practical applications are concerned. The objective of this paper is to estimate formant frequencies accurately in a noisy environment. The estimation performance of correlation based formant estimators deteriorates noticeably in the presence of noise. In order to overcome the detrimental effect of noise, first, instead of conventional autocorrelation function (ACF), we propose a repeated ACF (RACF) of the observed noisy speech with the zero lag compensation. It has been shown that the RACF is pole-preserving and capable of drastically reducing the effect of noise. Next, a ramp cepstrum model of the one-sided RACF of speech signal is developed and represented directly in terms of formant parameters. The proposed ramp cepstrum model is advantageous since it offers more noise-robustness and avoids the rapid decay occurs in the conventional cepstrum. Finally, in order to extract the formants from windowed and pre-emphasized observed speech, a ramp cepstral model-fitting is introduced where a residue based adaptive least-square optimization is employed. The proposed algorithm has been tested on synthetic and natural vowels as well as some naturally spoken sentences in the presence of noise. A better accuracy in formant estimation obtained by the proposed method significantly improves the recognition performance as well as the quality of the synthesized speech. The experimental results demonstrate a superior performance obtained by the proposed scheme in comparison to some of the existing methods at low levels of signal-to-noise ratio.

Dentals are grave

Darin Flynn (University of Calgary), Sean Fulop (California State University, Fresno)

We defend the novel claim that dental consonants are “grave” in the Jakobsonian sense of “having predominantly low frequency energy.” First, we show that the aperiodic energy associated with dental consonants is concentrated in the low region (< 2.5 kHz). Though relatively weak, this noise is nonetheless auditorily salient because it is not overshadowed by higher-frequency energy (cf. sibilants). Second, we show that the distinctive low-frequency energy of dentals is effectively enhanced by lowering second formant transitions. This can be achieved by secondary lip rounding, velarization, or pharyngealization. (F2 can also be lowered via retroflexion, but this is presumably impossible in dental articulations.) We argue that lip rounding and velarization indeed accompanied dentals historically in NT Athabaskan languages, so much so that [t θ , t θ^h , t θ' , θ , δ] evolved into [k w , k wh , k w , m , w] in Dogrib and dialectal Slavey. We also show that F2 was lowered via pharyngealization in BC’s Chilcotin language, so that the same dental series has evolved into Semitic-like ‘emphatics’ [ts $^{\text{f}}$, ts $^{\text{h}}$, ts $^{\text{f}}$, s $^{\text{f}}$, z $^{\text{f}}$]. Third, we present evidence from ‘acoustic assimilations’ that listeners perceive a lowered F2 as a cue to dentalization: coronal consonants become dentalized when adjacent to round and/or back vowels in Warrgamay, to retracted/pharyngealized vowels in Kalenjin, and to retroflex sounds in Northern Irish English. The fact that labialization, velarization, pharyngealization, and retroflexion have

nothing to do with dentals articulatorily strongly supports our claim that the latter are defined crucially by low frequency energy.

The evaluation of noise level in hand-held pneumatic tools (rock drill) by “PENEUROP CAGI TEST CODE” method

Farhad Forouhar Majd (Isfahan University of Medical Sciences), Parvin Nassiri (Tehran University of Medical Sciences)

This study evaluates noise in hand-held rock drills used in lashotor stone mines in Isfahan by the method PENEUROP CAGI TEST CODE, which specifies the procedures which can be applied to certain items of construction equipment. This method is designed to evaluate noise and vibration propagated by the hand-held pneumatic tools. Rock Drill is one of the four product categories specially identified as a major source noise and vibration in the federal noise control Act of 1972. In this study the noise produced by “Rock Drill” used in the above specify test method, its control procedures are also recommended. The method is based on ANSI (1971) to measure hand-held pneumatic totals noise such as rock drill, paving breaker, and is also famous to “PENEUROP CAGI TEST CODE”. All measures are in terms of sound pressure level in a situation of “A” for frequency weighting and “slow” for time weighting by B&K 2230 sound level meter and 1625 analyzer. In order to calculate the power of drill, the following formula is used:

$$LW_A = L_p + 10 \log_{10} \left(\frac{S}{S_0} \right)$$

The background SPL values are defined for two situations while compressor was turned on or off. The total of SPL, based on ANSI S5.1-1971, has been equal to 94.63 dB (A). There is an increase gradually in SPL from frequency 31.5 Hz to 4000 Hz. The critical frequency was 4000 Hz. The noise assessment will be done in a spherical area so that geometric center of drill should be located in a 1 meter distance of land. Thus, the total SPL depending to the different values, among evaluated levels, has been gotten.

The evaluation of whole-body vibration level in hand-held pneumatic tools (rock drill) by “PENEUROP CAGI TEST CODE” method

Farhad Forouhar Majd (Isfahan University of Medical Sciences), Parvin Nassiri (Tehran University of Medical Sciences)

This paper discusses the use of rock drills in lashotor stone mines in Isfahan by the method PENEUROP CAGI TEST CODE, which shows the methods for using construction equipment in stone mines under work circumstances. This method is designed to evaluate noise and vibration propagated by the hand-held pneumatic tools. Rock drills are categorized as the major sources of noise and vibration by the federal noise control Act of 1972. This paper demonstrates vibration values in three directions X, Y, Z and three vital indices for workers while working with equipment, compared to standard graphs, produced by “Rock Drill” used in the above-specified test method. This paper focuses on measuring and predicting of created vibrations as whole-body indices in a rock drill used in stone mines by B & K vibration meter 2512 model. In order to evaluate them we had to locate the whole-body acceleration on the ground where the operator was standing and processing his work. The indices include reduced comport (RC), fatigue decreased proficiency (FDP), and exposure level (EL) for a frequency response of 1-80 Hz. Obtained conclusions have been shown high values of rms acceleration in Z and XY axes for EL and FDP, respectively. All values are based in a 5 hour time of exposure. We were faced with high measures of vibrations in different axes but the most important index in comparison with its recommended limits can be EL, because it is a better index weighted for human vibration in workplaces. RC Index is not also recommended for occupational and industrial jobs because it has more limitations.

“I said made (mate?)”: Catalan/Spanish bilinguals’ production of English word-final obstruents

Natalia Fullana (University of Ottawa), Ian R. A. MacKay (University of Ottawa)

The perception and production of English voicing contrasts often pose difficulties for native speakers of Romance languages (Flege, Munro, & MacKay, 1995; Flege, Munro, & Skelton, 1992; Yavaş, 1994). In particular, Catalan and Spanish learners of English have been reported to devoice voiced obstruents in word-final position (Cebrian, 2000) or to spirantize and delete voiced segments (Flege & Davidian, 1984). In addition to differences in the phonetic inventories between the second language (L2) and the learners’ first language, such findings might be explained as a function of age of onset of L2 learning and experience in the L2, as hypothesized by Flege’s (1995) Speech Learning Model (SLM). The present study aimed to further examine the production of the voicing contrast in English word-final obstruents /p b t d s z/ by Catalan/Spanish bilingual learners of English in a formal learning context, as well as the effects of age of onset (4–14 years) and experience in English (years of formal instruction). Results showed that the 47 Catalan/Spanish bilinguals failed to produce English voiced obstruents accurately. The duration of their preceding vowel was shorter and that of the closure phase or frication longer than those of the 4 English controls (see also Hogan & Rozsypal, 1980). Additionally, learners produced voiceless stops with longer closing phase and preceding vowel durations than native English speakers. Furthermore, results suggested that in formal learning contexts age of onset and experience effects do not conform to SLM’s predictions for immersion settings.

Recent studies of infrasound from industrial sources

William J. Gastmeier (HGC Engineering), Brian Howe (HGC Engineering)

Infrasound from industrial equipment and its effects on residential neighbours has been the subject of much discussion in the literature. This paper summarizes three recent situations which have been investigated by HGC Engineering. These involved the measurement of infrasound sources, the manner in which infrasound was perceived by the neighbours and its physical effects. The sources included a large reciprocating engine used for power generation, an industrial sieve used to sift material and wind turbines used for power generation.

Influence of dilatational propagating motion on diffuse field transmission loss of orthotropic sandwich composite panels

Sebastian Ghinet (Université de Sherbrooke), Nouredine Atalla (Université de Sherbrooke)

The transmission loss modeling of sandwich composite panels with compressible soft core is addressed in the present study. Dilatational and flexural propagating modes of motion are used in a wave approach to express their associated structural impedances and compute the transmission loss of sandwich panels. Each layer of the sandwich is assumed orthotropic with membrane, bending, transverse shearing and rotational inertia behaviours. It is shown that the classical transmission loss modeling accounting for only antisymmetric (flexural) motion is appropriate for panels with relatively stiff and thin core. However, compression deformations over the core’s thickness become important when the core is thick and soft. It is shown that the noise transmission loss is largely affected by dilatational propagating behaviours as well as the orthotropy of the layers. Experimental and literature results are compared with numerical simulations using the present theoretical approach to demonstrate its validity and effectiveness. The influence of the core’s and skins’ stiffness, orthotropy and thickness on the transmission loss is also addressed in a parametric study to highlight the behavioural optimization and design rules.

The temporal window of audio-tactile integration in speech perception

Bryan Gick (University of British Columbia/Haskins Laboratories), Yoko Ikegami (University of British Columbia)

Asynchronously presented audio and visual signals are integrated asymmetrically in speech perception (e.g., Dixon & Spitz 1980, Summerfield 1992, Smeele & al. 1992). Munhall & al. (1996) found that AV integration of speech occurred when the audio signal lagged the video signal by 240ms, but when audio preceded video, integration allowed only 60ms of asynchrony, hypothesizing that this asymmetrical effect window may be attributable to the differing atmospheric speeds of sound and light. However, this explanation has not been substantiated via comparison of other perceptual modalities, nor is it clear whether the perceptual allowance for this physical asynchrony is learned or innate. An experiment was conducted in which tactile stimuli (small bursts of air) were directed at perceivers' necks while they heard productions of *pa* and *ba*. In baseline conditions, a burst occurring immediately prior to vowel onset (simultaneous with aspiration for *pa*) significantly enhanced perception of *pa* and significantly interfered with perception of *ba*. Asynchronous results showed a similar effect window to previous AV studies: For asynchronously presented bursts, the temporal window of the interference effect (but not the enhancement effect) was asymmetrical, with integration occurring when the air burst followed the audio signal by 200ms, but only by 50ms when the air burst followed the audio signal. The direction of asymmetry parallels the temporal difference between the speed of sound vs. air flow (Anderson & al. under review), supporting the physically-based hypothesis. Implications will be discussed for learned vs. innate theories of cross-modal mappings.

Noise exposure from communication headsets: the effects of environmental noise, device attenuation and preferred SNR under the device

Christian Giguère (University of Ottawa), Hilmi R. Dajani (University of Ottawa)

Individuals wearing headsets in the workplace are exposed to both the environmental noise around them and the acoustic signals generated by their device. However, these two sources are not independent. Typically, users will adjust their headset volume so that speech and other acoustic signals are maintained sufficiently above the environmental noise permeating the device to ensure proper communication. This points to a significant dependence of headset sound exposure on the surrounding environment noise, the device attenuation and the user's preferred signal-to-noise (SNR) under the device. This paper will review these factors through a re-analysis of two earlier Canadian studies on communication headset exposure: the surveys of Dajani et al. (1996) at eight industrial sites and Crabtree (2002) in military aircrafts. The purpose of this re-analysis is to gain more insight into the main determinants of headset sound exposure, and to provide an experimental basis for a new calculation method to assess communication headset exposure. This method is under consideration as part of a new appendix to CSA Standard Z107.56, which will cover measurement of noise exposure from employees wearing headsets in the workplace.

Fabrication of acoustic absorbing topologies using rapid prototyping

Oliver Godbold (Loughborough University), Jian Kang (University of Sheffield), Rupert Soar (Loughborough University), Richard Buswell (Loughborough University)

The fabrication of traditional porous absorbing materials utilizes manufacturing processes that impose limitations on pore topology. Pore shape and size are often determined by random processes such as layered fibre deposition, packing of loose granulate or the nucleation and growth of gas bubbles within open celled foams. This results in uncertainty in their absorption

prediction and limits the type of porous structures that can be produced. Advanced, computer driven, additive manufacturing processes often collectively referred to as 'Rapid Prototyping', introduce a method of achieving greater control of absorber topology, and offer more freedom in the design of acoustically absorbing structures. One such process uses the selective deposition of fine thermoplastic filaments to incrementally fabricate a 3D solid object. Modification of the deposition parameters has enabled the production of intricate porous structures, allowing customisable levels of porosity to be incorporated during fabrication. Samples 25mm thick produced using this method have demonstrated significant absorption over 1600Hz. Alternative porous configurations have been used as a resistive covering over the neck of a simple Helmholtz type resonant absorber; their addition damping the resonant effect, resulting in an increased bandwidth of absorption. The ability to incorporate different porous topologies and easily fabricate complex shapes, introduces many previously infeasible design possibilities. This has been exemplified through the fabrication of a single material absorber structure combining porous and resonant absorbing elements.

Structural segmentation of popular music with partially supervised clustering

Daniel Graves (University of Alberta), Witold Pedrycz (University of Alberta)

Structural segmentation of today's popular music is accomplished using MPEG7 spectral descriptors. The purpose is to produce labelled structural segments of music such as chorus, verse, etc that can be used for example to summarize music in order to quickly sample a musical collection. The underlining machine learning framework incorporates hidden Markov models on the MPEG7 spectral descriptors with an excess number of states. The distribution of states across a window of several consecutive audio frames is determined for each frame. The distributions are clustered using a partially supervised hierarchical clustering algorithm. The supervision component of clustering takes into account the temporal proximity of frame samples in order to encourage frames to group together that are close in time. The objective is to summarize music with its structural information with potential applications in creating audio thumbnails or skipping ahead features in music to skip to certain segments in a song. Experimentation will be completed on a set of the popular songs, genres and artists in the music industry.

Durational properties of stressed syllables as a cue for English-accented French?

Christian Guilbault (Simon Fraser University)

This study examines durational patterns of stress groups in an effort to find the cause of greater temporal rhythmic variability in English-accented French. Durational patterns of foreign-accented speech have not been examined closely enough to be fully understood despite the promising results in accounting for dialectal variation in English (Deterding 2001) or in the speech of English-accented French (Guilbault 2002). The current study expands on a previous pilot study (Guilbault 2006) that revealed noticeable differences between stressed syllables of learners and the ones uttered by native speakers of French. Two groups of learners (advanced and intermediate) and one control group (native French speakers) recorded short utterances in a delayed-repetition task. Target words for durational analyses were monosyllabic and trisyllabic words inserted at the end of a stress group which was part of a larger carrier sentence. Results will show noticeably smaller increases in the duration of syllables affected by primary stress when uttered by learners, and a clear segmental effect on the duration of those syllables. Trisyllabic words were produced with greater variability in the duration of the penultimate syllable by second-language learners. These results confirm the role of duration as an important acoustic cue in English-accented French and justify a more in-depth analysis in order to determine the exact causes of divergent temporal structures. Potential consequences for a better account of foreign accent are briefly discussed.

Tsilhqut'in ejectives: a descriptive phonetic study

SooYoun Ham (University of Victoria)

Stops are one of the most common sounds across languages of the world. Among these pervasive sounds, ejectives form a unique group that is distinguishable from other types of stops. Their particular mechanism of articulation, such as larynx raising and unusually high oral pressure, separates them from the others. More interestingly, a listener perceives them differently and makes a distinction from pulmonic stops. What is it that we perceive when hearing ejectives? Do we perceive certain acoustic cues or auditory qualities that are part of their distinctive phonetic nature? Are these phonetic characteristics always distinctive? The present study, phonetically investigating these intriguing sounds in Tsilhqut'in, is composed of two major analyses. One is an acoustic analysis that instrumentally examines a dataset of ejective and non-ejective stops in the language with respect to acoustic dimensions such as Voice Onset Time (VOT) as to compare all the stop classes in terms of their acoustic properties. Next, in order to determine the characteristics of ejectives across languages, Tsilhqut'in ejectives were compared with ejectives in different languages (e.g., Inguish). The other analysis is auditory, whereby I have examined how I perceived the ejectives and compared my auditory judgments with the acoustic measurements to see correlations, if any, between results from the two analyses. The findings of the study indicate that Tsilhqut'in ejectives do not follow a traditional binary typology of ejectives. That is, they are *neither strong nor weak*, as is often claimed in the literature (e.g. Lindau 1984). They are congruent with what recent studies (e.g. Warner 1996) have found of ejectives in other languages – *phonetic variability*. This means that the dichotomy cannot account for the variability in ejectives at the phonetic level and that an optimal way of classifying ejectives across languages still awaits discovery.

Vowel quality and duration in Deg Xinag

Sharon Hargus (University of Washington)

Deg Xinag, an Athabaskan language spoken in western Alaska, has been described as containing the following vowel inventory: /e a o ə ʊ/ (Krauss 1962, Leer 1979). Although Deg Xinag "/ʊ/" is a reflex of Proto-Athabaskan *ʊ, Deg Xinag /ʊ/ gives the auditory impression of being a short version of /o/. If true, then Deg Xinag would be typologically unusual in having no high vowel phonemes. A qualitative, acoustic study of vowel quality with three native speakers revealed some degree of overlap between /o/ and "/ʊ/" when F1xF2 plots are examined. A quantitative study of vowel duration with the same speakers confirmed that /o/ and "/ʊ/" differ in duration. The Deg Xinag vowels can be divided into two sets, a short set (/ə "ʊ"/, mean .097 sec.) and a long set (/e a o/, mean .182 sec.). The vowels within each set are not significantly different from each other but each vowel in the short set is significantly different from each vowel in the long set. The qualitative and quantitative results combined thus suggest that the Deg Xinag vowel inventory can be summarized as /e a o ə ɔ/ (or perhaps /e: a: o: ə o/). Durational differences between the longer and shorter vowels of Deg Xinag are comparable to normative vowel duration data available for other Athabaskan languages (Witsuwit'en (Hargus 2007) and Tsek'ene (Hargus in preparation)), and support the reconstruction of Proto-Athabaskan as having full (*i: *e: *a: *u:) and reduced vowels (*ə *ʊ **"ɑ") (Krauss 1964).

Production of English lexical stress by inexperienced and experienced learners of English

Yunjuan He (University of Florida), Qian Wang (University of Victoria), Caroline Wiltshire (University of Florida)

The study investigated how native speakers of Mandarin, a tonal language, use phonetic cues to differentiate stressed and unstressed syllables in producing English disyllabic words. Sixteen native Mandarin speakers from Northern China participated in a production experiment which involved two tasks: English real word reading and English-like non-word reading. Among the sixteen participants, eight were inexperienced learners of English who were college students in China, and the other eight were experienced learners of English who had studied at a University in the USA for at least two years. It was found that: 1) Mandarin speakers misplace stress more on initial-stressed real words and final-stressed non real words. This indicates that learning experience can change speakers' phonetic preference, since Chinese speakers were taught the tested real words with final stressed pattern during their English education. 2) Compared to inexperienced learners, experienced learners produce more native-like acoustic cues (larger ratios of amplitude and duration in stressed vs. unstressed syllables) to stress English words. However, the ratios are still smaller than English native norms. 3) There is no significant difference between the two tested groups on the ratio of pitch, and the ratio is similar to the English native norm. The above results are discussed in terms of L1 positive transfer (phonological pitch pattern and phonetic pitch value on disyllabic words) and L1 negative transfer (syllable-stressed rhythm and penultimate stress preference).

Measurement and calculation of the Santur parameters

Peyman Heydarian (London Metropolitan University), Lewis Jones (London Metropolitan University), Allan Seago (London Metropolitan University)

The Santur, also known as the Hammered Dulcimer in English, is a flat string instrument, played with a pair of hammer sticks. It is a direct ancestor of Piano. In this paper, we will describe the instrument, its physical characteristics and acoustical properties like pitch deviation and inharmonicity factor. These parameters provide a better understanding of the instrument. The results will be explained and discussed.

Numerical model of a thermoacoustic engine

O. Hireche (Centre national de la recherche scientifique), C. Weisman (Centre national de la recherche scientifique), D. Baltean (Centre national de la recherche scientifique), M. X. Francois (Centre national de la recherche scientifique), P. Le Quere (Centre national de la recherche scientifique), L. Bauwens (University of Calgary)

A Low Mach number model of a complete thermoacoustic engine has been developed. The model is asymptotically consistent. It assumes that in the resonator, the flow is described by isentropic, linear acoustics. The assembly comprised of the stack and heat exchanger is taken to be acoustically compact. The flow in this region is modeled as viscous and conducting; pressure is uniform up to a dynamic pressure correction of the order of the Mach number squared. Thus a closed-form solution of the acoustics is coupled with a transient numerical solution for the stack and heat exchangers. Transient results for the engine start will be presented.

Acoustical evaluation of non-classroom university learning spaces

Murray Hodgson (University of British Columbia), Jorge Moreno Garcia Villareal (University of British Columbia)

This paper reports the results of an acoustical evaluation of non-classroom learning spaces at UBC. In twelve buildings, twenty-five indoor spaces -- a restaurant, a cafeteria, libraries, dedicated study spaces, building atria, etc. -- used for learning activities by at least 50 people were studied. The evaluation involved physical and acoustical (reverberation time, sound propagation, Speech Intelligibility Index) measurements, and occupant activity and satisfaction questionnaires. Questionnaires were administered three times (morning, lunchtime and afternoon) on one day. The questionnaires asked about satisfaction with, and the effects of, the acoustical and non-acoustical environments. The acoustical measurement results were compared with established acceptability criteria. Questionnaires were analyzed for differences between times of day, test space, etc. The questionnaire responses and acoustical-measurement results were correlated. Using both as possible predictors, multivariable-regression models for predicting and explaining occupant satisfaction with, and the effects of, the acoustical environment were developed.

Ray-tracing prediction of sound-pressure fields in empty and fitted rooms

Murray Hodgson (University of British Columbia), Gary Chan (University of British Columbia), Owen Cousins (University of British Columbia)

A Monte-Carlo ray-tracing model has been adapted to the prediction of sound-pressure fields in rooms with surfaces of arbitrary surface impedance, and containing parallelepiped obstacles. Phase changes due to propagation distance and wall reflection were accounted for. Diffraction around obstacle edges was modeled by the Unified Geometric Theory of Diffraction. The new model was validated in comparison with predictions by reference models (e.g. FEM) and with the results of experiments in an anechoic chamber and in real rooms. This paper discusses the development of the new models, the results of the validation tests.

Acoustical evaluation of six 'green' office buildings

Murray Hodgson (University of British Columbia), Rosamund Hyde (Stantec Consulting), Blair Fulton (Stantec Consulting), Catherine Taylor-Hell (Stantec Consulting)

As the acoustical part of a project evaluating the performance of six 'green' office buildings, meetings with the designers, walk-through surveys and detailed acoustical measurements were made to explain the positive and negative comments of building occupants obtained from a questionnaire satisfaction survey. The objective was to determine how to design better green buildings. Design performance criteria were established. Measurements were made of noise level, reverberation time, Speech Intelligibility Index and noise isolation. The results of these measurements and of the occupant satisfaction surveys, and the lessons learned regarding optimal design, are discussed.

Recent developments in environmental assessment methods for wind turbine noise

Brian Howe (HGC Engineering), Nick McCabe (HGC Engineering)

In early 2007 HGC Engineering prepared a document reviewing the assessment of sound from wind turbines and recommending best practices for the Canadian Wind Energy Association. At the time, the assessment guidelines of the Ontario Ministry of the Environment were the most

sophisticated in Canada, relying on internationally recognized standards (ISO 9613 & IEC 61400-11) and allowing for a variation in both wind turbine sound power and background sound as a function of wind speed. The MOE recently completed a review of their guidelines and published a new draft document in May of 2008. This draft did not change the fundamental criteria for the assessment, but it improved the definition of various parameters used as inputs in the analysis (ground attenuation coefficients, air absorption, receptor locations, etc.) and raised, for the first time, the requirement that consideration be given to the site specific wind profile (wind speed as a function of height). This paper reviews the effect of these improvements and looks at the overall degree of precision versus the variability in sound levels as measured during acoustic audits undertaken by HGC Engineering at two wind turbine projects.

Shock wave reflection and focusing phenomena in fluid-interacting shell systems

S. Iakovlev (Dalhousie University), G. Dooley (Dalhousie University), K. Williston (Dalhousie University)

A system consisting of a circular cylindrical shell and two fluids, the internal one and the external one, is addressed. The response of the system to a weak shock wave propagating in the external fluid is analyzed. The focus of the study is on the pressure patterns that develop in the internal fluid, particularly the effects associated with shock wave reflection and focusing. It is demonstrated that when the fluids are identical, the internal wave pattern follows a well-known scenario inherent to the reflection of a shock wave off a concave cylindrical surface, i.e. the primary reflection followed by the focusing. When the fluids are different, however, the situation changes dramatically, and scenarios other than the usual reflection-focusing sequence were shown to exist. In particular, a very interesting focusing-reflection-focusing scenario was observed for certain internal fluids. The practical implications of such changes of the classical reflection and focusing patterns are discussed.

An investigation into wind generated aero-acoustic tones

Peter Alexander Ibbotson (Marshall Day Acoustics), John Pearse (University of Canterbury)

Louvres, grilles and balustrades on the outside of buildings are often located where they can be subjected to high wind velocities. A multi-storey car park building in Christchurch, New Zealand was recently the cause of complaint due to a tonal noise emission audible only during windy conditions. In-situ acoustical and vibration analysis suggested the noise was an aero-acoustic tone generated by wind flowing over a balustrade. The mechanism by which the tone was generated was investigated in a low noise wind tunnel and various treatments were evaluated. This paper presents an overview of the phenomenon and discusses the results of acoustical and vibration testing. The results of noise control treatment testing are summarized and recommendations made to prevent the problem from arising in other developments.

Do Mandarin-English bilinguals have an accent in their L1 vowel production?

Haisheng Jiang (Simon Fraser University)

It is claimed that the L1 phonetic categories established in childhood do not remain static; instead, they may undergo modification when similar L1 and L2 sounds interact in the process of L2 learning (Flege, 1995). If the L1 sounds are influenced by the L2 sounds and deviate from the L1 norm, the L1 monolingual listeners should be able to detect it. In a perception experiment, Mandarin monolingual listeners evaluated the goodness of the Mandarin vowel production by

Mandarin-English bilinguals (n=33). The results show that, when compared with Mandarin monolinguals, Mandarin-English bilinguals received a significantly lower rating for Mandarin vowel /y/, a vowel non-existent in English. Despite the non-significant group difference in the ratings assigned to Mandarin vowels /a/, /aj/, /au/, /ej/, /i/, /o/, and /u/, some individual speakers were judged accented in the production of these vowels. Acoustic dimensions that possibly contributed to Mandarin-English bilinguals' accent in L1 vowel production include lower F1, larger F2 movement and tone deviation. There is no evidence indicating that Mandarin-English bilinguals of low L1 use outnumbered those of high L1 use in being judged as accented. This study provides further evidence for the claim that the L1 phonetic system established in childhood is susceptible to change. It contributes to the less well-studied field of L2 influence on L1. In particular, it suggests the necessity to include dissimilar L1 segments (e.g. Mandarin /y/) in speech learning theories.

ISO/IEC GUM applied to estimation of sound power measurement uncertainties

Stephen E. Keith (Health Canada)

While there have been necessary and significant improvements in ISO machinery sound power measurement standards using sound pressure, over the last 40 years the measurement uncertainty has essentially stayed the same. This is problematic because some users of these standards do not report the uncertainties associated with the methodology, believing that they are too large. An obvious solution is to use an uncertainty budget, so that uncertainty can be better estimated in relation to a given measurement. An additional benefit is the potential for more efficient identification and reduction of the sources of uncertainty. This paper implements the methods of the ISO/IEC Guide 98-3, Guide to the Expression of Uncertainty in Measurement (commonly referred to as the GUM) to illustrate an uncertainty budget for machinery sound power standards. In typical situations the most significant sources of error are due to the background noise, correction for the diffuse field, sound level meter and number and distribution of measurement points. Recommendations to minimize these uncertainty components will be discussed.

Using a change in percent highly annoyed with noise as a potential health effect measure for projects under the *Canadian Environmental Assessment Act*: application to wind turbine noise

Stephen E. Keith (Health Canada), David S. Michaud (Health Canada), Stephen H.P. Bly (Health Canada)

Under the *Canadian Environmental Assessment Act* (CEAA), Health Canada may be requested to provide expert advice on health effects of noise for an environmental assessment. To provide this advice, it is desirable to establish quantitative criteria for adverse health effects as a function of project-related long-term changes in noise. The criteria should be based on scientific research that has demonstrated a reasonable cause-effect relationship between an adverse effect on health and community noise exposure. This paper summarizes a recent review on this topic (Michaud et al. 2008, *Canadian Acoustics*), which provided the rationale for using a change in percentage highly annoyed with noise (%HA_n) as one of the health endpoints for assessing noise impacts. The current paper shows how a change in %HA_n can be used to derive noise mitigation criteria for wind turbine project-related increases in community sound levels based on available dose-response relationships for wind turbines as well as more common sources. Comparison to other accepted criteria suggests that, typically, mitigation is recommended if the magnitude of increase in %HA_n exceeds 6.5%. In a typical quiet rural community, this corresponds to a predicted wind turbine project noise that does not exceed 45 dBA.

Relationship between ventilation, air quality, and acoustics in 'green' and 'brown' buildings

Alireza Khaleghi (University of British Columbia), Karen Bartlett (University of British Columbia), Murray Hodgson (University of British Columbia)

This paper discusses a pilot project involving direct monitoring of ventilation, indoor-air quality and acoustical conditions in 'green' and 'brown' buildings on the UBC campus. The objective was to determine the relationships between various building concepts and environmental factors, and the implications of the results for the ventilation-system concept/design, especially in 'green' buildings. Measurements were made in four categories of rooms in buildings with natural, displacement and forced-air ventilation systems, without and with acoustical treatment. Measurements were made of ventilation rates (air changes per hour), indoor air-quality (fibre concentrations, VOC concentrations, ultrafine-particulate concentrations), and the acoustical conditions (noise levels, reverberation times). Environmental results were correlated with the type of ventilation system and with one another. The buildings studied, the measurements performed and the main results are described. The lessons learned about building ventilation-system design are discussed.

Spectral peaks as acoustic correlates to speech perception

Michael Kieft (Dalhousie University), Tara Collins (Dalhousie University)

How do spectral peaks corresponding to resonances or formants contribute to the identification of vowels by human listeners? Speech perception research in the past has largely focused on the frequencies first two or three resonance peaks, or formants, in the speech spectrum. Although there is a large body of evidence in support of formants as correlates to vowel identity in human speech perception, automatic speech recognition algorithms have largely shunned formant frequencies as they are very difficult to extract from the speech signal and measurement of spectral peak frequencies is considered highly unreliable. Although many believe that a detailed study of formant tracking in human listeners may provide much needed insight into solving this problem, a preliminary question must first be addressed: Are accurately tracked formant frequencies a necessary prerequisite to vowel perception by human listeners? This paper describes results from several studies that examine the relationship between formant frequencies and more global spectral properties like relative formant amplitudes or spectral tilt. In one series of studies, it is shown that formant amplitudes are largely ignored in an /i/-/u/ continuum in which the level of F_2 in /u/ is varied. In another set of studies, it is shown that perceptual extrapolation of a formant sweep is mostly dependent on peak frequency and not amplitude. Both studies carry with them caveats, the most important of which is that, while formant-frequency synthesis parameter is known in these experiments, we cannot know the perceived formant frequency without further psychoacoustic testing.

Case study: development of a high performance acoustical pipe lagging system

Corey D. Kinart (HGC Engineering)

Urban sprawl in recent years has led to encroachment of residential developments on industrial installations unlike ever before. As a result, control of industrial noise emissions to meet government regulations has become increasingly challenging. This paper presents a case study of the development, installation and testing of a high performance acoustical lagging system for above ground high pressure piping at a natural gas compressor station.

Acoustic realization and perception of English lexical stress by Mandarin learners

Yuwen Lai (University of Kansas), Joan Sereno (University of Kansas), Allard Jongman (University of Kansas)

The acquisition of English lexical stress by Mandarin L2 learners was examined through production and perception studies. An acoustic study focusing on the implementation of mean F0, max F0, duration, intensity, and F2 in stressed and unstressed vowels in noun-verb word pairs contrasting in stress location (e.g. *object* and *object*) was conducted. The results from native English speakers (n=10) showed that all correlates were utilized to signal stress in nouns. In verbs, however, mean and max F0 were not utilized and duration cues were amplified. Implementation patterns for Mandarin L2 learners (beginning=9; advanced=9) were similar to native speakers in nouns. However, in verbs learners used mean and max F0 as well. Reduction of unstressed vowels was found to be inconsistent in learners when compared to native speakers. A perceptual study utilizing disyllabic novel word 'dada', with resynthesized max F0, duration, and vowel quality, was conducted in order to evaluate the perceptual relevance of those cues in stress perception. Results from an identification task indicate that full vowels induce significantly stronger stress perception in all listener groups. In terms of max F0 and duration, beginning listeners (n=25) relied mainly on duration, advanced listeners (n=25) focused more on max F0, while native listeners (n=25) made use of both duration and max F0 in perception. These findings are discussed in terms of the similarities and differences in prosodic systems between Mandarin and English, as well as the possible discrepancies in production and perception data from second language learning research.

Differential effect of therapeutic ultrasound on dentoalveolar structures during orthodontic force applications in-vitro (tension vs. compression forces)

B. Lam (University of Alberta), S. Aldagheer (University of Alberta), T. El-Bialy (University of Alberta), A. Sloan (Cardiff University)

The aim was to investigate if there is differential effects of low intensity pulsed ultrasound LIPUS on the dental and periodontal tissue during orthodontic force applied to a rat mandible slice organ culture in-vitro. Mandibles were dissected from eight 28-day-old male Wistar rats; the mandibles were sectioned into 1.5mm with a 0.006" diamond wafer saw and cultured into six well plates and cultured at 37°C in an atmosphere of 5% CO₂ in air, in a humidified incubator. After 24 hours 0.019x0.025" stainless steel wire with a loop was applied to each slice with a calibrated force of 50 grams. The slices were divided into three groups including control, 5 min US application and 10 min US application. The LIPUS were applied using a 2.5 transducer that produces incident intensity of 30 mW/cm² of the transducer's surface area. After five days the slices were fixed and evaluated histologically for the predentin and cementum thicknesses, odontoblast, preodontoblast and periodontal ligament PDL cell count on the sides undergoing compression and tension. The thickness of the cementum and predentine layers were significantly increased in the ultrasound groups, specifically on the tension side. The odontoblast and PDL cell count were increased on the tension side compared to the compression side in all groups. Cell counts were increased in the ultrasound group compared to the non-ultrasound group. Preodontoblastic cell layer were increased in the 10 min LIPUS group. More frontal resorption lacunae were found in the compression side of the ultrasound groups. The conclusion is that more bone remodeling potential results from LIPUS application due to the increased cell number in the PDL, and induced tooth material apposition presented by the increased thickness of the non-mineralized predentine layer. The results were amplified when the samples undergone tension compared to compression.

A semi-analytical approach to the study of the transient acoustic response of cylindrical shells

Cédric Leblond (La Rochelle University), Jean-François Sigrist (DCNS Propulsion), Serguei Iakovlev (Dalhousie University)

A semi-analytical method related to the effects of a transient acoustic pulse on a submerged cylindrical elastic structure is proposed. Contrary to the classical models based on the Kirchhoff-Love hypotheses, this fluid-structure interaction problem is treated here with a fully elastic model for the structural dynamics. The approach is based on the methods of Laplace transform in time, Fourier series expansions and separation of variables in space. This method enables us to eliminate some previously reported drawbacks related to the A0/S0 waves when the Kirchhoff-Love hypotheses are used. It is demonstrated through the comparison with experiments that the new approach results in much more realistic images of the radiated acoustic field.

An acoustic study of the L2 VGN rime production

Ya Li (University of Victoria)

To explain why Mandarin speakers tend to produce English words such as *down* as Mandarin *dàng*-like, this study uses acoustic measurements to investigate Mandarin speakers' production of English VGN (a diphthong vowel followed by a nasal coda) rimes and Mandarin VN (a monophthong vowel followed by a nasal coda) rimes. Test words includes 5 English words with VGN rimes, *pine/coin/gown/pain/cone*, 5 corresponding words with VG rimes, *pie/coy/cow/pay/go*, and 4 Mandarin words with VN/VG rimes, *gàng/kào/gòng/gò*. A total of 1120 tokens (14 words x 4 repetitions x 20 Mandarin speakers) were examined in the phonetics software, Praat. Specifically, the mean F1-F0 and F3-F2 (differences between the first and fundamental formant frequencies and between the third and second formant frequencies) over the first half and the last half of the vowel duration were measured to estimate the vowel height/backness movement over the duration. Also, the first, second, and third nasal formants at the mid point of the nasal duration and the band energy difference between 0~525 Hz and 770~1265Hz bands over the nasal duration were calculated to predict the nasal place (alveolar/velar/uvular). Last, the vowel duration and the nasal duration in each token were used to infer the degree of vowel-nasal coupling. The acoustic results reveal that a strong interaction exists between the nasal place and the vowel height/backness in both L1 and L2 production; that is, the nasal place covaries with the vowel height/backness change over the duration rather than merely backness assimilation as suggested by previous studies (e.g., Chen, 2000).

A perception study of glottalization in Gitksan resonants

John Lyon (University of British Columbia)

There may be generational differences regarding the production and perception of glottalized resonants (GRs) in Gitksan, a Tsimshianic language. Younger speakers are purportedly neutralizing the distinction between glottalized and modal resonants. GRs are interesting in that the relative timing of the glottal and oral gestures which compose these segments may vary across phonological and morphological environments, speakers, and languages. While this makes these segments articulatorily unstable, it may also make them more difficult to consistently perceive. The question this study asks is the following: Does the apparent neutralization of glottal cues by a speaker affect his/her perception of those same glottal cues, and do place, duration, and environment of the glottal cue affect their perception? I first examine productions of the minimal pair /nax/ *snowshoe* and /n'ax/ *bait* by two speakers from different generations. Next, I create a stimulus set for each speaker, consisting of his/her own *snowshoe* utterances re-synthesized with an inserted glottal closure of varying durations and placed in varying positions.

These are then tested on the speaker. Results suggest that a speaker who neutralizes the distinction between /nax/ *snowshoe* and /n'ax/ *bait* in his/her production also requires a longer closure in order to perceive a stimulus as /n'ax/ *bait* than a speaker who does not neutralize the distinction. Pre-glottalization was not found to be significantly more perceptible than mid-glottalization, which is evidence against Silverman (1997). Finally, one speaker showed that less duration of closure was required to shift perception at a word boundary than word internally.

The perceptual basis of velar epenthesis in Costa Rican Spanish

Bethany MacLeod (University of Toronto)

One of the processes that take place when native speakers of Costa Rican Spanish (CRS) produce English words is the insertion of the stop /g/ preceding the glide /w/. This insertion results in the pronunciation of an English word such as *why* as [gwaj]. Velar insertion has received very little attention in the literature. To contribute to filling this gap, the present study collected new data through an experiment and examined the acoustic properties of the CRS /g/ and [w] and the English /w/ to determine if velar insertion is related to a misperception of the English glide caused by acoustic similarities between the languages. An acoustic analysis was conducted on the experimental data measuring the durations, formant values and relative intensities of the velars and glides. In addition, productions of English words containing /w/ by native English speakers were compared to those by CRS speakers. The acoustic and statistical analyses found that while the English and Spanish glides were very different, the Spanish /g/ and English glide were not significantly different. This suggests that the CRS speakers do not perceive the English /w/ as a Spanish [w], but rather as a Spanish /g/ and that this perception results in inserting a velar in the production of English words in place of producing the English glide in the same manner as the Spanish glide. This research shows that velar insertion in the production of English words by CRS speakers derives from an influence of perception.

Listening with ear and hand: cross-modal integration in music perception

Michael Maksimowski (Ryerson University)

In addition to auditory information, the perception of music often involves visual and vibrotactile information, making it an ideal domain through which to study crossmodal integration. Recent research has demonstrated a strong influence of visual information on auditory judgments concerning music. However, we have very little empirical information regarding integration of vibrotactile information in music. In Experiment 1, participants made judgments of interval size for unimodal presentations of melodic intervals in auditory, visual, and vibrotactile conditions. In Experiment 2, participants made judgments of interval size for cross-modal presentations of intervals comprised of stimuli presented in the three unimodal conditions of Experiment 1. Crossmodal presentations were either congruent (derived from the same interval) or incongruent (derived from different intervals). Participants were instructed to base judgments on the auditory information alone. Results will consider differences in the extent of visual and vibrotactile influence on auditory judgments and the role of learning in cross-modal integration in music.

Establishing relationships between acoustic and physical properties of shoddy-based fibre absorbers for acoustic modelling

John Manning (Bauer Industries/Université de Sherbrooke), Raymond Panneton (Université de Sherbrooke)

Post industrial recycled fibres or "shoddies" have found widespread use in the production of fibrous sound absorbers but they are not easily characterized due to the variability in their fibre

composition. This paper presents the results of acoustic and physical property testing on fibrous absorbers composed in whole or in part by shoddies. Testing was completed on various different shoddy constructions processed by three different methods: thermal bonding, chemical bonding, and mechanical bonding. The acoustic parameters measured were porosity, airflow resistivity, and normal incidence sound absorption. The characteristic impedance and complex wavenumber of each material were deduced from the normal incidence sound absorption measurements. Furthermore, empirical relationships between the acoustic properties themselves and measured physical properties such as weight per unit area, thickness and fibre diameter are presented. The more complex acoustical models require several material properties that may be difficult to measure. It is anticipated that the above relationships can be used to reduce the amount and difficulty of testing required when employing an appropriate acoustic model.

Comparison of measured and modelled transmission loss in Emerald Basin

Brian H. Maranda (Defence Research and Development Canada – Atlantic), Nicole E. Collison (Defence Research and Development Canada – Atlantic)

During July 2007, Defence Research and Development Canada – Atlantic conducted sea trials in Emerald Basin, an open-ocean area off the coast of Nova Scotia, Canada. As part of these trials, acoustic data were collected for the purpose of measuring the acoustic transmission loss (TL) at frequencies from 1 to 2 kHz. The acoustic signal that was transmitted during the experiment consisted of multiple continuous-wave (CW) tones, allowing simultaneous measurement of the transmission loss at the tone frequencies. The range was varied to a maximum of about 14 km by towing the acoustic projector from the ship CFAV *Quest* and receiving the transmitted signals on freely drifting sonobuoys. The towed source was at 50-m depth, and the sonobuoy receivers were set for a depth of either 60 m or 120 m (in a water depth of about 265 m). In this paper, the experimental procedures and data collection are first described, and then the measured TL curves are presented. Although the tone frequencies span an octave, the TL curves at the different frequencies cluster together quite closely. However, it was found that the transmission loss to the shallow receivers was much less than for the deep receivers, owing to the presence of a near-surface duct. Finally, the measured TL values are compared with the predictions made with numerical acoustic-propagation models (Bellhop and PECan). There is generally good agreement between the measured and modelled TL values.

An acoustic study of stress in L2 production of German and Spanish

Viola Miglio (University of California, Santa Barbara), Dorothy Chun (University of California, Santa Barbara)

Stress is a multi-faceted construct and its correlates may differ in different languages (Idson & Massaro, 1980, Potisuk, 1996). Among the physical dimensions perceived as stress are higher pitch and longer vowel duration (Lehiste, 1970). The phonetic realization of stress may be subject to interference from the learner's L1, and may be interpreted by a native speaker as a serious phonological problem amounting to misplacement of stress (Low & Grabe, 1999; Mennen, 2007); we therefore set out to compare the correlates of stress in vowels produced by American English speakers learning German or Spanish in a classroom setting. Our preliminary results for German indicate that while the native speaker controls manipulate pitch more than duration to indicate stress, the L2 learners rely more on duration than pitch, and tend to centralize unstressed vowels, just as in Hirschfeld and Trouvain (2007), who suggested that stress and the associated vowel reductions are a major problem of L2 German learners at the word-level. For Spanish, our native speakers' correlates of stress were duration, pitch, and loudness, whereas Spanish L2 learners only manipulated duration consistently to mark stress. We found no reduction to schwa in L2 Spanish (as in Morrison, 2003). Both Spanish and German learners manipulated duration rather than other native correlates of stress. Future research will establish whether this is due to

erroneous perception of prominence or to ease of manipulation of duration compared to other correlates of stress.

Tongue root retraction and tongue tip recoil in Xhosa alveolar click releases

Amanda L. Miller (University of British Columbia/Cornell University)

I present high-speed ultrasound data showing details of the alveolar click release in IsiXhosa. The data are produced in the frame sentence *Ndi qaba isonka*. [ndi laba isɔŋga] 'I am spreading something on the bread.' Data were recorded with both probe anchoring (probe to head stabilization) using an Ultrasound Stabilization Headset developed by Articulate Instruments, and head movement correction using the Palatoglossatron technique (Mielke et al. 2005). Clicks have all been characterized as having a velar posterior place of articulation based on 30 fps X-ray data (Traill 1985), while Miller et al. (2007a, b) have shown that the posterior place of articulation in the Khoisan languages Khoekhoe and N|uu have a uvular posterior place using 30 fps ultrasound data. The 124 fps data presented in this paper allow a much more detailed understanding of the click release. The posterior constriction starts out as velar, similar to the [g] in the frame sentence, but retracts in order to achieve cavity expansion for rarefaction of air necessary in the lingual airstream used in clicks. The release occurs at a lower and retracted uvular place of articulation. Fine detail of the tongue tip release is also seen. After the tongue tip release involved in the alveolar constriction, the tip consistently exhibits a recoil effect, leaving the tip raised higher in the mouth than otherwise seen in the following vowel [a]. Anterior and posterior constriction release dynamics explain coarticulatory effects involving this consonant.

Low intensity pulsed ultrasound stimulates osteogenic differentiation of human gingival fibroblasts

Nesrine Mostafa (University of Alberta), Paul Scott (University of Alberta), Douglas N Dederich (University of Alberta), Michael Doschak (University of Alberta), Tarek El-Bialy (University of Alberta)

Low-intensity pulsed ultrasound (LIPUS) has been reported to enhance cellular differentiation of several cell types including periodontal ligament cells, bone cells, cementoblasts and odontoblasts. However, little is known about its effects on human gingival fibroblasts (HGF). Therefore, our research investigated the in-vitro effects of LIPUS on HGF osteogenic differentiation as a new tool for periodontal therapy. HGF cells were cultured in 48-well plates at an initial density of 2.5×10^3 cells/well. HGF in the experimental group received LIPUS treatment for 5 or 10 min/day for 28 days (frequency 1.5-MHz, 200-s pulse modulated at 1 kHz, with an output intensity of 30 mW/cm^2). They were subsequently analyzed for cell viability, specific alkaline phosphatase (ALP/DNA) activity, and differential gene expression by reverse-transcriptase polymerase chain reaction (RT-PCR) at weeks 1, 2, 3 and 4. Total DNA content showed no significant changes in the LIPUS treated group over 28 d. Moreover, LIPUS treatment did not affect cell viability at any time point. Interestingly, 5 min of LIPUS treatment significantly increased both ALP activity and Osteopontin (OPN) gene-expression by week 3 and 4 compared to other groups ($p < 0.05$). LIPUS treatment for 10 min/day only enhanced the ALP activity after 4 weeks compared to the control ($p < 0.05$). The conclusion is that LIPUS stimulation at intensity of 30 mW/cm^2 (5 min/day) selectively enhanced the differentiation of HGF as evidenced by significantly increased ALP activity and OPN gene-expression. This study should serve as a base for furthering the therapeutic usage and efficacy of LIPUS stimulation for periodontal therapy.

Variability in Cantonese speakers' productions of English vowels

Murray J. Munro (Simon Fraser University)

Although much work on second language (L2) phonetic learning emphasizes the effects of the first language (L1) sound system on that of the second, L2 segmental production is rarely consistent across, or even within, speakers from the same L1 background. Moreover, a close examination of variability in L2 speakers' productions has the potential to yield useful insights into the L2 acquisition process. This investigation focuses on variability in the English high vowel productions of speakers of Hong Kong Cantonese who were relatively homogeneous with respect to linguistic and social background, but who differed in their length of Canadian residence. While [i ɪ] and [u ʊ] occur on the phonetic surface of Cantonese, their distribution in CVCs depends on the final consonant ([t] or [k]). To evaluate Cantonese speakers' success in producing high vowels in the new contexts required by the L2, familiar English CVC words with a wide range of rhymes were elicited under two conditions: a picture naming task and a delayed repetition task. The findings indicate influences of phonetic context with substantial inter- and intra-speaker variability; a positive correlation between L2 experience and vowel production accuracy for some, but not all VC rhymes; and significant effects of elicitation task and word frequency. The results are consistent with a view of phonetic learning that sees L2 acquisition processes as gradual, approximative, and frequency-based.

Prediction of Speech Transmission Index in eating establishments

Musarrat Nahid (University of British Columbia)

The objective of the research is to model existing eating establishments (EEs) and then, taking into account the Lombard effect, predict the acoustical conditions for speech (i.e., speech transmission index (STI) and, therefore, speech intelligibility (SI) and speech privacy (SP)) in EEs without and with sound-control measures, to determine how to design the EE to optimize the acoustical conditions. Using the CATT Acoustics software, STI predictions are initially made for a reference configuration (EE without any control measures). Then predictions are made for new test configurations created by changing one or more design factors (room volume, surface absorption, customer seating position, barriers between tables). The results are compared to those of the reference configuration, in order to evaluate the control measures. CATT does not take the Lombard effect into account automatically. Therefore, using an empirical model of the Lombard effect in EEs, talker voice levels are predicted and input into CATT along with other EE-model parameters. CATT then predicts the total secondary talker's speech level at the primary listener; these levels and the background-noise level (BNL) are added to create the new BNL for the next step. In it, using this new BNL and the primary talker's voice level, STI is predicted for both speech intelligibility and speech privacy. Predictions done so far suggest that putting barriers around tables may be the best way to achieve good speech communication.

A perfectly matched layer technique for Lattice Boltzmann Method

Alireza Najafi-Yazdi (McGill University), Luc Mongeau (McGill University)

In recent years, the Lattice Boltzmann Method (LBM) has emerged as a promising computational technique in fluid dynamics. LBMs have intrinsic advantages over conventional Navier-Stokes schemes. Accurate and robust nonreflecting boundary conditions are needed for their application in aeroacoustics. The focus of the present paper is to develop a non-reflecting boundary condition based on a perfectly matched layer (PML) technique developed for computational electromagnetics. In this technique, an absorbing layer, called a perfectly matched layer, is laid between the interior domain and the computational domain, causing waves to be damped exponentially before

reaching the artificial boundary. Unlike for other absorbing techniques, it can be proven mathematically that there will be no reflection from the interface of the PML and the interior of the domain, for any frequency and direction of the incident wave. A PML for Lattice Boltzmann Equations (LBEs) was developed, which is applicable to viscous flows with arbitrary direction. The performance of the new PML was validated by simulating canonical Gaussian pulse propagation problems.

Pattern recognition in speech perception research

Terrance M. Nearey (University of Alberta)

This paper reviews several relatively successful applications of statistical pattern recognition methods to model human speech perception. For constrained laboratory problems, such as the perception of isolated vowels or CV syllables, very simple static pattern recognition methods often predict listeners' performance quite well. Two recent PhD theses from our laboratories have extended this work fruitfully to crosslanguage and second-language contexts. Pattern recognition tools can make a useful contribution exploring such traditional issues as speaker normalization and the evaluation of competing cue sets. However, modeling the perception of real speech signals beyond syllable-length utterances will likely require dynamic recognition techniques, which can accommodate input of arbitrary length (ranging from monosyllables to whole sentences). Dynamic pattern recognition methods, including hidden semi-Markov models (HSMM), have been studied extensively in automatic speech recognition technology. This paper will briefly sketch an initial application of a an HSMM framework to model the perception of VC(C)V syllables by human listeners. It is argued that speech perception researchers have much to gain from a careful, critical study of the advanced methods available from speech technology.

Occupational noise exposure assessment using perceived and quantitative measures

Richard Neitzel (University of Washington), William Daniell (University of Washington), Lianne Sheppard (University of Washington), Hugh Davies (University of British Columbia), Noah Seixas (University of Washington), Lianne Sheppard (University of Washington)

Characterization of variable noise over long periods of time presents a major exposure assessment challenge. Strategies such as assignment of exposure based on job title may not provide adequate exposure contrast or precision for variable exposures. This study evaluated subjects' perceptions of occupational noise exposure as an alternative or complementary exposure assessment strategy. Twenty subjects were recruited at each of three worksites with different noise environments (continuous, intermittent, and highly variable). Full-shift dosimetry measurements (n=206) were made on each subject during four workshifts over two weeks. Perceived exposure information was collected via surveys on subjects' first (n=58) and last (n=57) monitored shifts, as well as through timeline logs completed during dosimetry. The first survey focused on the first shift only, while the second survey covered the whole two week period. The results were as follows. Timeline log data suggested that subjects could detect changes in noise level and variability within a workshift. Survey items on perceived noise variability and impulsiveness performed well at the continuous and highly variable sites. The contrast between exposure groups created using job title was generally smaller than that provided by subjective survey items. Finally, the precision of exposures predicted from survey items was comparable to, or slightly better than, that of job title for several survey items, and the addition of survey items to prediction models which included job title improved model fit and precision. In conclusion, supplemental perceived noise exposure information appears to offer promise for improving exposure estimates, particularly for individuals with highly variable exposures.

Performance characterization of a small-capacity thermoacoustic cooler for air-conditioning applications

Insu Paek (Kangwon National University), Luc Mongeau (McGill University), James E. Braun (Purdue University)

The performance of a standing-wave thermoacoustic cooler was investigated. Comparisons were made between measured data and predictions from linear acoustic theory for various mean pressures, acoustic powers, and gas mixtures. In general, the experimental data was in good agreement with theoretical predictions. In comparison to the performance without water flow through the cold-side heat exchanger, higher COPs were obtained in presence of water flow because of a reduced stack temperature difference due to higher cooling load and overall heat transfer conductance. Temperature profiles along the stack obtained from DELTAE showed that the temperature profiles with water flow through the cold heat exchanger were non-linear and resulted in a slight decrease in the system performance. DELTAE simulations of the performance of a different, well known quarter wavelength cooler showed that, for a fixed stack length, the temperature span must be large for efficient heat pumping through the stack. If the temperature span is relatively too small compared to stack length, the temperature profile is non-linear leading to a significant reduction in COP. The results described in this paper highlight that standing wave thermoacoustic coolers must be operated at or near their design temperature span for optimal system performance. This work was supported by the Office of Naval Research (ONR), and the Air-Conditioning and Refrigeration Technology Institute (ARTI).

The effect of the type of acoustical distortion on lexical access

Marianne Pelletier (University of Toronto at Mississauga), Marco Coletta (University of Toronto at Mississauga), Renee Giroux (University of Toronto at Mississauga), Huiwen Goy (University of Toronto at Mississauga), Kathy Pichora-Fuller (University of Toronto at Mississauga)

The current study investigates the effects of acoustic distortions on lexical access. Lexical access is broken down according to the Cohort Model (Marslen-Wilson & Welsh, 1978) into three steps: activation, selection, and integration of the candidates. Different acoustic distortions are thought to affect different steps in this process. Our study manipulated the type of distortions to include both low-pass filtering and time compression as a replication of Aydelott and Bates (2004), and it was extended to include an additional distortion: multi-talker noise. Subjects were presented with intact or distorted stimuli, which provided a semantically congruent, incongruent, or neutral context for the following intact target word. The task consisted of determining whether the target word was a real or a non-word, and the length of time needed to make this lexical decision was recorded. This study compared the reaction times between each condition to determine which type of distortion affects which step in the lexical access process. Results from the low-pass filtering distortion replicated Aydelott and Bates. (2004) findings, where there was a decrease in facilitation and inhibition when the stimuli were distorted. However, our results from the time compression distortion failed to replicate the inhibition effect revealed in the previous study. Instead, a facilitation effect was seen. Finally, an analysis of reaction times in the noise distortion condition revealed a facilitation effect, but no inhibition effect. The results of this study support the hypothesis that the type of distortion influences the extent to which context contributes to lexical access.

Measuring the displacement of engines using acoustics

Richard J. Peppin (Scantek, Inc)

Acoustics provides a fast approach to measure engine displacement. Since it does not use a liquid, it is both clean and dry. By using the ideal gas law, and measuring differences in pressure changes, the volume can be determined. As a result, in a matter of seconds the capacity (displacement) of an enclosed volume can be determined within about $\pm 0.1 \text{ cm}^3$. This paper discusses the theory and gives some examples, along with applications for measuring capacity. In addition, the technique to measure the volume of a body, previously done with displacement of water, will be discussed.

A case of superior auditory spatial attention

Robert Quelch (University of Toronto at Mississauga), Gurjit Singh (University of Toronto at Mississauga), M. Kathleen Pichora-Fuller (University of Toronto at Mississauga)

Auditory spatial attention was measured in a 40-year old musician (RQ) with moderate bilateral hearing loss above 4 kHz and chronic tinnitus. In one condition, a target sentence was presented from one loudspeaker, and masker sentences were presented from two different loudspeakers. In a second condition, the precedence effect was used to simulate the location of the target and two masker sentences. Target identity was based on a callsign cue given either before or after stimulus presentation, and *a priori* target location varied from certainty to chance. The data for this exceptional listener are compared to data for a group of younger adults with normal hearing and a group of musicians. Overall, RQ performed similarly to the groups when the callsign was cued after stimulus presentation. When the callsign was cued in advance of stimulus presentation, enabling the allocation of attention, whereas the performance of the young groups and most of the musicians worsened with increasing target location uncertainty, RQ maintained a high level of performance. Possible explanations for RQ's superior performance include changes in auditory processing as a result of prolonged musical expertise or inherited talent that somehow differed from that of most of the other musicians, listening strategy, changes in attention related to tinnitus or hearing loss. Clearly, his auditory spatial attention abilities demonstrate resilience despite high-frequency hearing loss and tinnitus. Implications of these alternatives for rehabilitation will be discussed.

Fire resistance and noise control in multi-family buildings

J.D. Quirt (National Research Council of Canada), T.R.T. Nightingale (National Research Council of Canada)

The walls and floors that separate dwellings in apartment buildings and condominiums must serve many functions other than structural support. To select wall or floor assemblies, designers need ready access to collections of test ratings, so this talk begins with a brief overview of some key factors affecting both the fire resistance and sound transmission ratings of wall assemblies and floor/ceiling assemblies with gypsum board surfaces. For successful buildings, however, designers must also appreciate how the overall fire resistance and sound isolation may be seriously compromised by details at junctions and penetrations, and use a systems approach that respects the requirements for both sound and fire control. This presentation will focus on fire stops and fire blocks in the context of Canadian codes and standards, illustrate design options for fire stops at some common junctions and penetrations (mainly in the context of lightweight framed constructions), and provide guidance on corresponding acoustical issues.

Reverberation rooms and spatial uniformity

Ramani Ramakrishnan (Ryerson University/Aiolos Engineering Corporation), Anant Grewal (National Research Council of Canada)

Reverberation rooms are used to determine sound power levels of sources as well as to expose space articles to intense levels of sound. The volume of the room is usually determined from the frequency of interest and is inversely proportional to the frequency cubed. However, such a large volume requirement places a heavy burden on the low frequency capability of the reverberation room. A series of simulations as well as field measurements were conducted to test the volume requirement when the sources of sound were purely random broadband excitations. Two acoustic chambers located in Montreal and Ottawa were utilized for the field measurements. The volume of the Montreal chamber was less than 200 cu.m. The Ottawa chamber had a volume of 540 cu.m. The results of these investigations will be presented in this paper.

The role of acoustics in sustainable design

Zohreh Razavi (Stantec Consulting)

Good acoustics contribute to a quality work environment but it can pose challenges for sustainable designers. In fact, some of the most common practices associated with green design can actually negatively impact the acoustical performance of our workspaces. The solution, however, could be right above us. Some green buildings have insufficient sound-absorbing materials due to considerable use of radiant chilled and cooling slabs and transparent envelope. This may cause excessive reverberation, resulting in an acoustical environment which "feels" noisy and can result in impaired verbal communication. As architects continually strive to incorporate sustainability into their designs, the need for integrated design of all systems, including acoustics, becomes increasingly important. Since it has been investigated that one of the most important factors in greening a building is to provide an environment in which people can perform at their optimum level, designers must acknowledge that acoustics is a fundamental concern that can greatly contribute to the overall comfort level of a space and employee productivity. In this presentation some acoustical challenges in sustainable designs will be discussed.

The role of source motion on noise generated by wind turbines

Werner Richarz (Aercoustics Engineering Ltd)

From a distance the motion of the large wind-turbine rotor appears rather slow. Nevertheless, the tip speed of such rotors is of the order of 60 to 80 m/s. At such speeds source motion cannot be ignored. In contrast to propellers and helicopter rotors, the audible sound from wind turbines is not directly coupled to the rotating lift and drag forces, but associated with the turbulent boundary layer on and shed from the rotor blades. Well known correlations developed for broadband propeller and rotor noise can be used to predict overall spectrum levels with good accuracy. However, the characteristics time-varying sound of a wind turbine can only be accounted for by the acoustics of moving sources: namely: Doppler shift and convective amplification. These features are readily incorporated in a proper acoustic model. The formalism permits synthesis of wind-turbine noise pressure-time histories from first principles. Results for acoustically compact sources are presented in analytical as well as audible format. The latter is especially useful in assessing the relative importance of the various characteristics of wind turbine noise on the perceived loudness.

Propagation of wind turbine noise in a boundary layer

Werner Richarz (Aercoustics Engineering Ltd)

The doctoral dissertation of van den Berg has sparked a great deal of speculation about the role of wind shear in noise from wind turbines. The focus on a single parameter, the shear exponent α has led to a grotesque oversimplification of the multi-variable system that characterizes the propagation of sound in a real atmosphere. Although non-trivial, there are many suitable propagation models. Under most atmospheric conditions the rotors of a large wind turbine operate in but moderate wind shear. The shear is much less than experienced by wing-mounted propellers that operate in local up-wash without any significant increase in noise radiation. The sound generated by wind turbines is governed by the inflow conditions at the rotor, not those near the ground. A robust ray-tracing model is used to compute the mean propagation path from the elevated sources distributed over the outer region of the rotor disk to observers on the ground. Both wind and temperature gradients are accounted for, resulting in realistic effective sound speed profiles. For nominal source-observer distances greater than 300m, the local emission angle is close to the horizontal and the angle of incidence at ground level is near vertical. Results of a variety of characteristic atmospheric conditions will be presented, illustrating the inherent weakness of the shear exponent formalism.

The acoustics of sustainable buildings

Max Richter (Stantec Consulting), Zohreh Razavi (Stantec Consulting), Murray Hodgson (University of British Columbia), Alireza Khaleghi (University of British Columbia)

Five years ago, the greatest obstacle to green buildings was the perceived cost premium. Today, acoustics is cited as one of the key arguments against building green. This presentation will investigate the issue of the acoustical performance of green buildings. The Center for the Built Environment (CBE) at the University of California, Berkeley has conducted surveys of the occupants of over 300 completed buildings. The survey data reveals that building users have identified acoustical performance as the greatest problem facing green buildings. As the green building community moves quickly to find strategies to achieve carbon neutral buildings, we need to learn from the successes and failures of completed buildings. This feedback will help to make the next generation of green buildings better. The acoustical performance of green buildings cannot be looked at in isolation. Design strategies around ventilation, daylighting, thermal mass, and green material choices all have an effect on the acoustical performance of a building. This presentation will present observations and conclusions and offer guidance for designers and owners who want to improve the acoustical performance of their projects.

Effect of frequency in virtual rumble strips

Frank A. Russo (Ryerson University), Jeffery A. Jones (Wilfrid Laurier University), Mohammad Abdoli-Eramaki (Ryerson University)

The effectiveness of a rumble strip is determined by its ability to convey urgency (Transportation Association of Canada, 2001). When a vehicle makes contact with the rumble strip, a low-tech multisensory alarm is activated that involves auditory and vibrotactile stimulation. Modern vehicles have begun to implement virtual rumble strips as part of driver-assist systems. In principle, these virtual rumble strips should help to minimize the risk of lane departures, a common factor in head-on collisions and other serious accidents. Optimization of virtual rumble strips may be achieved quite simply by altering the frequency (Russo & Jones, 2007). Under three different modality conditions, we tested the effects of altering the frequency of virtual rumble strips on perceived urgency and reaction time: auditory-alone, vibrotactile-alone, and auditory-vibrotactile. Results are discussed with regard to both traffic safety and cross-modal perception.

Pressure/mass method to measure open porosity of porous solids

Yacoubou Salissou (Université de Sherbrooke), Raymond Panneton (Université de Sherbrooke)

This work presents a method to directly determine the open porosity of porous solids, and more particularly those used in sound absorbing liners. The method is based on the measurement of four masses at four static pressures from which the open porosity and true mass density are deduced using the perfect gas law. The precision of the method in relation with the used experimental setup is studied, and a simple expression is derived to predict the experimental error in function of the bulk volume of the sample to test. For a given experimental setup, this simple expression can be used to select the amount of bulk volume to test to reach a given precision. The method and its error prediction are tested experimentally on different samples of theoretically known open porosity.

Adjusting historical noise estimates by accounting for hearing protection use: a probabilistic approach and validation

Hind Sbihi (University of British Columbia), Kay Teschke (University of British Columbia), Ying MacNab (University of British Columbia), Hugh W. Davies (University of British Columbia)

Personal measurement of noise does not take use of hearing protection (HPD) into account, resulting in a potential misclassification of noise exposure. This is especially problematic for retrospective exposure assessments where data on individuals' use of HPD is not available. The objective was to re-assess existing historical noise exposure estimates in a cohort of BC sawmill workers and obtain new exposure values by accounting for HPD use and to validate our method by examining noise/hearing loss exposure response slopes. We previously developed a multilevel model of the likelihood of HPD use. Following the modeling of HPD use, existing unprotected noise exposure estimates were adjusted according to our models predictions. New exposure estimates were cumulated creating a cumulative adjusted exposure metric and tested against noise induced hearing loss using linear mixed effects modeling to account for the correlated nature of the repeated measurement data. The model with adjusted cumulative exposure was compared to an unadjusted cumulative exposure. The results were as follows. Hearing impairment data, adjusted and unadjusted cumulative noise exposure were obtained for 3880 workers in the cohort. The strength of association between exposure and disease was four times stronger and more significant with the adjusted exposure metric. Our study demonstrates that we can and need to account for personal protective equipment when reconstructing historical exposure estimates to remediate to non-differential misclassification. Our findings are important particularly given (a) the growth in large cohort studies to examine "non-auditory" effects of noise, and (b) that exposure misclassification is a major threat to most studies of weak associations, such as noise exposure and cardiovascular diseases.

The McGurk effect affected by the Right Ear Advantage

Mark Scott (University of British Columbia)

The right ear is better than the left at perceiving speech. This dominance is known as the Right Ear Advantage (REA) and is due to the dominance of the brain's left hemisphere in speech-processing (the right ear is primarily connected to the left hemisphere). The REA is typically demonstrated with tests of audio-only perception; this experiment demonstrates that audiovisual perception is also affected by the REA. Subjects were shown audiovisual stimuli in which the audio and visual information disagreed - the audio was /aba/, while the video was "aga". This combination typically produces an illusory perception of /ada/. This illusion is known as the

McGurk effect (McGurk and MacDonald 1976). Since the McGurk effect is more difficult to induce when the audio is perfectly clear, the illusion should be harder to induce when the audio is presented to the right ear compared to the left. That is what this experiment found. With the video component held constant, subjects adjusted the level of white-noise accompanying the audio component until they experienced the illusory /ada/ percept. They required significantly more noise to achieve the McGurk effect when the audio was presented to their right ear compared to when it was presented to their left. This finding extends the demonstration of the REA to audiovisual perception and potentially provides a new and simple clinical tool for the testing of hemispheric language-dominance, as the current tests (e.g. dichotic listening) are both complicated and somewhat unreliable (Hiscock et al. 2000).

Acoustic testing for phonologization

Kimary Shahin (Simon Fraser University)

Flemming (1997) explains how phonetic properties can be understood as the effect of the interaction of weighted constraints: for example, in CV sequences, F_2 of the consonant and vowel are co-determined by a constraint against deviation from consonant and vowel F_2 targets, and a constraint against quick articulator movement between the two targets. These constraints conflict because achieving both targets could mean quick movement, and slow movement could mean achieving neither target, assuming a uniform rate of speech. The actual production of a CV sequence is thus an optimization of this conflict, and both targets are undershot. In Flemming's model, optimization is computed mathematically by a cost function which is the sum of the weighted constraints. For phonetically grounded phonology, phonologization, then, occurs when there is a quantal shift from numerical weighting to strict dominance between constraints. This paper examines how this shift arises from the acoustic signal, focusing on postvelar phenomena in Interior Salish and Arabic.

A cepstral-domain algorithm for pitch estimation from noise-corrupted speech

C. Shahnaz (Concordia University), W.-P. Zhu (Concordia University), M. O. Ahmad (Concordia University)

Pitch is the prime acoustic cue in speaker recognition, speech enhancement, synthesis, coding, and articulation training for the deaf. Since noise obscures the periodic structure of speech, the pitch estimation in noise is an intricate task. This paper presents a new cepstral-domain algorithm for pitch estimation from noise-corrupted speech. We propose to employ a Discrete Cosine Transform (DCT) based power spectral subtraction scheme to enhance noisy speech prior to pitch estimation. In order to remove the deleterious effect of formants, the de-noised speech is passed through an inverse filter whose parameters are derived from the Linear Prediction (LP) analysis, yielding an output referred to as the LP residual. The novelty of the proposed method lies in the formulation of a Hilbert envelope of the LP residual (HELRL) wherein, the instant of significant pitch excitation due to glottal closure is highly emphasized compared to that revealed by the LP residual. With a view to reduce the pitch-errors substantially in a degraded environment, a DCT power cepstrum (DPC) of the mean subtracted HELRL is introduced that exhibits a more prominent peak at the true pitch relative to that demonstrated by the conventional cepstrum. Consequently, global maximization of the DPC results in a momentous improvement in the pitch estimation accuracy. For the performance evaluation, speech from the *Keele* database and additive white, car, tank, f-16 cockpit and babble noises from the *NOISEX'92* database are used. Extensive simulation results confirm that for a wide range of speakers, our algorithm consistently outperforms the state-of-the-art pitch estimation methods in a variety of severe noisy scenario.

FFT Tutor: A MatLab-based instructional tool for FFT parameter exploration

Arvind Singh (University of British Columbia), Tom De Rybel (University of British Columbia), José R. Martí (University of British Columbia)

Although the Fast Fourier Transform (FFT) has been the staple of signal processing for many years, it is still frequently misapplied. In many cases, the confusion stems from misconceptions regarding the relation between time-window size and the corresponding frequency-domain resolution, as well as the relation between time-domain sample rate and the corresponding frequency-domain bandwidth. Also, spectral leakage due to mismatches between the sample rate and the harmonic contents, and the choice of windowing technique, have a significant impact on the quality of the resulting spectral analysis. Spectral averaging techniques and other smoothing operations are then used to improve the appearance of the spectrum. However, windowing, zero-padding, and smoothing techniques change the nature of the signal. In this paper, we will first present a summary of the FFT and how its various parameters can be chosen in a practical way, followed by a discussion on spectral leakage, windowing and zero-padding. Then, a MatLab-based tool is introduced to help visualise the relevant concepts. The tool allows the user to evaluate graphically the influence of the analysis parameters on harmonic signals, as well as on a custom dataset, such as a sound recording. This, then, allows the user to experiment with, and optimize, the FFT analysis parameters easily, to enhance the resulting FFT spectrum, as well as visually compare the inverse of the spectrum produced with the original time-domain signal.

Psychophysical and neuroelectric evidence of context effects in auditory stream segregation

Joel Snyder (University of Nevada Las Vegas)

Past studies indicate that auditory perception can be influenced by previous sensory and perceptual experience. We examined whether such context effects are mediated by distinct brain areas from those representing current stimulus patterns. We recorded perception and event-related brain potentials while presenting an adaptation sequence consisting of repeating low and high tones with a variable frequency separation ($\Delta f = 3, 6, \text{ or } 12$ semitones), followed by a test sequence with a constant Δf (6 semitones). When the adaptation stimulus had a larger Δf , participants were more likely to perceive two segregated objects, resulting in a larger positive wave during adaptation. Conversely, a small- Δf adaptation sequence caused more perception of two objects during the following test sequence, resulting in a larger positive wave during the test. When participants had perceived two objects during the adaptation sequence, participants were also more likely to perceive two objects during the test sequence, resulting in a larger positive wave during the test. Comparing the scalp topographies of these neural modulations suggest the presence of representations of stimulus memory (prior Δf) and perceptual memory (prior perception) that arise from different brain regions than those involved in processing current stimulus characteristics (current Δf). These results suggest a complex set of brain areas involved in auditory perception, contradicting models that rely on peripheral representations only.

Variability of the consonant modulation spectrum across individual talkers

Pamela Souza (University of Washington), Frederick Gallun (National Center for Rehabilitative Auditory Research)

Current auditory models suggest that speech information is conveyed by a composite of modulations superimposed on a carrier signal, and that this modulation spectrum can be used to characterize available acoustic information. Our previous work demonstrated that consonants with similar modulation spectra were likely to be confused with one another, and implied that each consonant has a unique modulation spectrum. This follow-up study investigated variability of

consonant modulation spectra across individual talkers and vowel contexts. Speech recordings were drawn from a database (Markham & Hazan, 2002) of 45 adult and child talkers. Each produced the same set of tokens, including 23 consonants in three vowel contexts. The tokens were recorded on DAT and transferred for further analysis. Spectral Correlation Index (SCI) (Gallun & Souza, in press) values were calculated across subsets of the stimulus set. The SCI¹ is obtained by deriving modulation spectra (six octave-spaced carrier frequencies by six octave-spaced amplitude modulation frequencies) over the duration of an individual phoneme. Similarity across phonemes in a stimulus set is obtained by correlating the six modulation spectra (one for each octave) for each possible pair of phonemes in a stimulus set. Results indicated that the modulation spectrum for a single VCV token was very similar (typical SCIs > 0.9) across speakers and vowel contexts. This supports the idea that consonant identification is based on modulation characteristics which are maintained across different productions of the consonant.

Statistical analysis of classroom questionnaires and acoustical parameters

Gavin Steininger (University of British Columbia), Murray Hodgson (University of British Columbia)

Statistical analysis was done of the relationship between student questionnaire data and the measured physical-acoustical attributes for classrooms at the University of British Columbia. The objective was to determine how a classroom's acoustical characteristics affect student listening experience and, therefore, how to optimize classroom design. A survey was administered to 4882 students in 82 course sections in 10 classrooms. These questionnaires allow for the derivation of a rating of a classroom's perceived listening ease (PLE). Physical-acoustical measures of the 10 classrooms were also taken. The bivariate relations of PLE with each of the physical-acoustical measures are explored: There is a negative linear relation with all frequencies of ventilation noise. The quantity C50 is found to have a significant negative quadratic term when regressed against PLE. Reverberation time and early decay time are found only to affect PLE scores in small classrooms. A multivariate linear model was made to predict PLE score from the physical attributes of the classrooms. The variables used in the model were selected to optimize the AIC score of the model. The appropriateness of using A-weightings for noise levels in classrooms is discussed. A-weightings are found to be reasonable, if not optimal, for predicting PLE values. Recommendations are made on improving the design of future classrooms. These recommendations are: consider all attributes of a classroom together, build smaller classrooms when possible, allow classrooms to have some reverberation, and minimize high-frequency ventilation noise.

The role of attention in the development of auditory scene analysis

Elyse S. Sussman (Albert Einstein College of Medicine)

Auditory scene analysis begins from the moment we hear sounds, making it possible for the infant to distinguish its mother's voice from other noises in the environment. Despite the importance of this process for human behavior, the question of how perceptual sound organization develops during childhood is not well understood. The current study investigated the role of attention for perceiving sound streams in a group of school-aged children and young adults. We behaviorally determined the frequency separation at which a set of sounds was detected as one integrated or two separated streams and compared these measures with passively and actively obtained electrophysiological indices of the same sounds. In adults, there was a high degree of concordance between passive and active electrophysiological indices of stream segregation that matched with perception. In contrast, there was a large disparity between passive and active measures observed in children. Active electrophysiological indices of streaming were concordant with behavioral measures of perception, whereas passive indices were not. In addition, children

required larger frequency separations to perceive two streams of sound compared to adults. Taken together, this suggests that during development, attention refines the pre-attentive sound organization processes seen in adulthood. Thus, the current findings indicate that attention modulates neural activity associated with the discrimination of sound patterns, which is likely to strengthen the neural networks involved in automatically processing complex auditory scenes one encounters in daily life.

Mismatch between the production and perception of F0 in New Zealand English ethnolects

Anita Szakay (University of British Columbia)

Previous production experiments demonstrated that the two main ethnic dialects of New Zealand English significantly differ in prosodic features, such as rhythm and intonation (e.g. Britain 1992, Warren 1998, Szakay 2006). Szakay (2006) also showed that Maori English mean pitch is significantly higher than that of Pakeha English, the dialect spoken by speakers of European descent. The study also indicated that the increasing F0 values of Maori English are a result of a change in progress. The present perception study is part of a larger research project using innovative techniques to isolate the precise prosodic features that listeners might tune into in ethnic dialect identification in the New Zealand context. Seven speech conditions were created, each keeping different suprasegmental information in the speech signal (e.g. rhythm-only condition, intonation-only condition). The results of logistic regression analyses in each condition indicate that listeners do rely on the F0 characteristics to identify speaker ethnicity. However, listeners' expectations are at odds with the production results. Speakers with low pitch are perceived to be Maori, while those with high pitch are perceived to be Pakeha. This suggests that stereotypes might have a stronger effect on speech perception than the actual change in progress in this ethnolect of New Zealand English. Perception studies in the US yielded similar results with regards to the pitch characteristics of a speaker, where lower F0 levels were associated with African Americans and higher F0 with European Americans (Hawkins 1992, Foreman 2000, Thomas & Lass 2005). The paper discusses the results in terms of stereotypes associated with minority groups using an exemplar theoretic framework of speech perception and speech production.

L2 English vowel learning by Mandarin speakers: does perception precede production?

Ron I. Thomson (Brock University)

The hypothesis that L2 speech perception precedes L2 speech production (Best & Tyler, 2007; Flege, 1995) is supported by research showing that naturalistic L2 exposure leads to improvement in L2 pronunciation, and that perceptual training has the same effect. Incommensurate methods used to measure perceptual and production abilities, however, may not accurately reflect the relationship between perception and production. For example, listeners' identification of speech tokens using categorical labels is often contrasted with productions elicited using orthographic prompts. In the first case, native speaker (NS) auditory models are provided immediately before listeners make their choice of category, while in the latter, no auditory model is provided. Instead, listeners must rely on previously established phonological representations. Such a method may bias L2 learners towards better performance on the identification task than on the production task. This study examines the relationship between L2 English vowel identification and L2 English vowel productions using the same auditory stimuli for both perceptual and production tasks. Twenty-two Mandarin L1 speakers were asked to identify ten Canadian English vowel categories produced by two NSs, using visual labels they had become familiar with during a larger training study. The same NS stimuli were used to elicit L2 English vowel productions. Results demonstrate that the learners' performance varied by vowel category. For some vowels, their performance on the identification task was worse than their

performance on the production task. Implications regarding the relationship between L2 speech perception and production will be discussed.

Geoacoustic inversion of noise from ships-of-opportunity with unknown position

Dag Tollefsen (Norwegian Defence Research Establishment), Stan E. Dosso (University of Victoria)

This paper examines a method for geoacoustic inversion using noise from ships-of-opportunity with a priori unknown position. The ship position is estimated by simultaneous optimisation over environment and source track parameters. A Bayesian matched-field inversion method is then employed, with small a priori source position uncertainty centred on the optimal track. The method is applied to ship-noise data recorded on a horizontal array deployed on the seafloor in shallow waters of the Barents Sea (experiment conducted by FFI), including data from a quiet research ship and from a merchant ship. Estimates of the sound-speed profile and density of the seabed are compared with results from inversion of data from a towed controlled-source, and with reference values from other geophysical data collected in the area.

Testing of Health Canada's Acoustic Chamber at the Consumer and Clinical Radiation Protection Bureau, based on ISO Standard 3745:2003

Jason T. Tsang (Health Canada), Stephen E. Keith (Health Canada), William J. Gastmeier (HGC Engineering)

Third party acoustical certification testing was performed to document the baseline acoustical conditions in Health Canada's acoustic chamber at the Consumer and Clinical Radiation Protection Bureau in Ottawa, Ontario. Testing was performed prior to the installation of a new fire suppression system. The anechoic chamber is lined with flattipped fibreglass wedges along the walls, ceiling, and floor, designed for a cut-off frequency of 50 Hz. The interior (wedge tip to wedge tip) is approximately 13 m long, 9 m wide, and 8 m high. The chamber has a removable reflective floor system, constructed of concrete tiles, to convert it to a hemi-anechoic state. Testing was completed in the hemi-anechoic state. Testing used the pure tone method prescribed in Annex A of ISO 3745 (2003). For test purposes, a dynamic loudspeaker and a compression driver were mounted under the floor. The dynamic loudspeaker was used for the frequency range from 50 Hz to 1 kHz, and the compression driver was used for the frequency range from 1.25 kHz to 10 kHz. The results indicate that the acoustic chamber complies with the requirements of ISO 3745 (2003) for pure tone stimuli within its useable volume ($\lambda/4$ from the source and wedge tips). This paper presents the results of the testing and discusses characteristics of the sound sources.

Processing of Japanese pitch accent by native Japanese and English listeners

Jung-yueh Tu (Simon Fraser University), Xianghua Wu (Simon Fraser University), Yue Wang (Simon Fraser University)

It is well-established that language processing is left hemisphere dominant. Previous findings, however, indicate that lateralization of different levels of prosody varies with their functional load as well as listeners' linguistic experience. For example, whereas processing of lexical tone is left hemisphere dominant for native, but not non-native, listeners, that of emotional intonation is right hemisphere dominant for both native and non-native listeners. The processing of pitch accent poses an interesting question, since, in terms of functional load, pitch for pitch accent is less heavy than that for lexical tone but heavier than that for intonation. This study examined hemispheric processing of Japanese pitch accent by 16 native Japanese listeners and 16 English

listeners with no linguistic tone or pitch accent language background. Pitch accent pairs were dichotomically presented and the listeners were asked to identify which pitch accent pattern they heard in each ear. Preliminary results showed that for both the Japanese and English listeners, the percentage of errors for the left ear and that for the right ear were comparable, indicating no ear preference (i.e., no hemispheric dominance). The Japanese group did not reveal left hemisphere dominance for pitch accent, as previously found for linguistic tone processing by native listeners. These findings are discussed in terms of how linguistic function differentially affects the hemispheric specialization of different domains of prosodic processing by native and non-native listeners.

Implementing noise prediction standards in calculation software - the various sources of uncertainty

Erwin Hartog van Banda (DGMR Software bv), Hardy Stapelfeldt (Stapelfeldt Ingenieursgesellschaft mbH)

Uncertainty in noise prediction is usually thought of being related to the accuracy of the input data and the accuracy of the prediction method. However documentation of the prediction standard that contains ambiguous algorithms and unclear definitions of geometrical calculation data will also lead to uncertainty in software implementation. The past has proven that different software packages based on the same prediction standard give different results even in simple cases. This as a result of the unclear documentation of the standard itself. There are 2 important steps needed for a calculation: path detection and path calculation. The first step has to do with the import of digital data. It must be clear how geographical information of buildings, roads, barriers, rivers, parks, bridges, tunnels etc. has to be used in the calculation. The current National and International methods however give only very little information on how to do so. The second step has to do with the calculation itself. There should be no unclear phrases and no ambiguous algorithms in order to avoid different interpretations. This paper addresses the sources of uncertainty that are related to the (un)clearness of the prediction method and the choices software engineers have to make while implementing ambiguous prediction algorithms. This paper also makes recommendations to avoid these uncertainties.

Low noise pavement technologies in the Canadian Context – a review of recent studies conducted in Alberta and Ontario

Peter VanDelden (RWDI AIR Inc.), Scott Penton (RWDI AIR Inc.)

Road traffic noise is a major nuisance issue in Canada. Research conducted originally in the US indicates that porous pavements, often incorporating rubber binder agents or rubber crumbs, can reduce traffic noise. Several Canadian municipalities and provincial governments have investigated the effectiveness of these pavements in the Canadian context. This paper presents a review of the Canadian results in context with other international results.

Typical hourly traffic distribution for noise modelling

Peter VanDelden (RWDI AIR Inc.), Scott Penton (RWDI AIR Inc.)

The modelling of noise from road traffic frequently depends on an understanding of its distribution over the course of a day. Hourly traffic counts are desirable to determine $L_{eq}(1 \text{ hour})$, L_{dn} , and L_{den} values. However, AADT, 8-hour count, or peak am/pm hourly count values are often the only available traffic data. The relationship of these values to hourly distribution is evaluated using a collection of hourly traffic counts from Ontario roads. A typical hour-by-hour traffic distribution is

presented. The variation in noise levels between the actual distributions and the typical distribution is presented.

The Harmonoise/Imagine method

Hans van Leeuwen (1DGMR Industry, Traffic & Environment bv), Erwin Hartog van Banda (DGMR Software bv)

According to the European Noise Directive noise maps and actions plans must be made every 5 years. The first round of noise maps finished in 2007. For countries that do not have a national calculation standard the so called 'interim' standards like ISO 9613 were used. In parallel with the EU noise mapping projects the European Harmonoise and Imagine projects were run by a consortium of European partners. The aim of the projects was to develop a new European noise calculation method that can replace the 'interim' methods for the second round of noise maps in 2012. Although it is not sure that this new Harmonoise/Imagine method will become an European standard in time for the second round of noise maps, it can be considered as a major step forward compared to the current empirical calculation methods. The Harmonoise/Imagine method has been developed for computation of the noise indicators LDEN and Lnight per 1/3-octave. The method uses fresnel zones for more continuous modeling of reflections from ground and objects as well as a physical source model including the latest road and railway noise source separation techniques. Since sound propagation effects strongly depend on meteorological conditions, the method is able to handle different meteorological conditions in different directions of propagation. The method can be used both for detailed computations in case of noise assessment and for noise mapping requiring less detailed data. This paper gives an overview of the method and its applications.

The ANSI STANDARD S12.68-2007 method of estimating effective A-weighted sound pressure levels when hearing protectors are worn: a Canadian perspective

J r mie Voix (Sonomax Hearing Healthcare Inc.), Alberto Behar (University of Toronto), Willy Wong (University of Toronto)

The most serious and difficult issue with hearing protection devices (HPD) is to estimate the protected noise level at the wearer's ear. Such estimation is difficult for two reasons. First, the effective field attenuation of an HPD on a given user is rarely known. Instead, the Noise Reduction Rating (NRR), the 98th percentile of the group attenuation of test-subjects tested in a laboratory, is currently used. Such use of the NRR provides a highly unrealistic assessment, not only because the NRR value dramatically overestimates the attenuation with respect to real-life situations, but also because the NRR does not reflect the inter-subject variability in terms of attenuation for a given HPD. The second problem is that it requires the C-weighted exposure level, which is rarely available in practice, and assumes that the noise spectrum is flat per octave-band (pink noise). The ANSI standard S12.68 has been recently developed to specifically address these two issues. Although the attenuation values to be used in that standard are not specified, three calculations schemes are presented to compute the A-weighted protected level from an A-weighted exposure level. In a quite novel approach, a Noise Reduction Level Statistics (NRSA) is computed on a large dataset of industrial noise spectrum and which also takes into account the inter-subject variability. The NRSA, is expressed at the 20th and 80th percentile values, in order to reflect both this variability's on the noise spectrum content and on the change in the attenuation from individual to individual. Although the NRS is much more easy to use and also more realistic, its direct use in the Canadian workplaces may require some further investigations on three aspects: (i) the representativity for the Canadian workplaces of the NIOSH 100 spectrums used, (ii) the origin of attenuation values to be used for the NRS computation (iii) the percentiles to be used to express the NRS values in order to be representative of the field attenuation achieved by a group of Canadian users.

Addressing the effects of overtopping vegetation on the performance of highway noise barriers

Clair Wakefield (Wakefield Acoustics)

The reflection and scattering of mid-to-high frequency sound by vegetation may be observed in a number of familiar situations. For example, when driving beneath the overhanging branches of a broad-leaved tree with the top down or sunroof open or when shouts or gunshots are echoed back from the edge of a forest. While fascinating, these phenomena have little broader relevance. However, when vegetation is close to and overtops a noise barrier intended to shield residences from highway noise, the resulting scattering of sound will reduce insertion loss, particularly at higher frequencies. Not only is the A-weighted barrier insertion loss reduced, but on the "shielded" side of the barrier, the traffic noise no longer sounds "muffled" and while there may still be a worthwhile noise production provided, the impression may be that the barrier is having little or no effect. This phenomenon may have major implications for the success of a highway noise mitigation program, in particular when the insertion loss of noise barriers must, as a contractual requirement, be confirmed through field measurement. This paper will explore the effects of sound scattering by overtopping vegetation on noise barrier performance and discuss ways in which they might be addressed, either physically or administratively, including the interaction between pavement design (e.g., quiet pavements) and vegetation scattering effects.

Vibrational characteristics of harp soundboards

Chris Waltham (University of British Columbia)

The musical quality of a harp depends on many factors, but key among these are the mechanical properties of the soundboard. First, in order to understand the relationship between the vibrational behaviour of a bare soundboard and that of the completed instrument, a 36-string harp was built from scratch. Measurements made at each stage of construction showed how the bare soundboard properties affect those of the finished harp. Second, the soundboards of several harps of different types and quality were assessed by measuring admittance along the string bar, and these results are presented here. These data suggest that the most crucial relationship is that between the modal shapes and modal frequencies of the soundboard, and the position and fundamental frequencies of the strings attached to it. This allows a general statement to be made about the vibrational qualities of a soundboard, and suggests a recipe for improving poor soundboards.

Sound absorption of a microperforated panel backed by an irregular-shaped cavity

C.Q Wang (Hong Kong Polytechnic University), L. Cheng (Hong Kong Polytechnic University), J. Pan (University of Western Australia)

The sound absorption and the surface impedance of a micro-perforated panel (MPP) backed by a trapezoid acoustic cavity with varying depth are numerically investigated. The coupled acoustic problem between the MPP and the acoustic cavity is formulated using a modal expansion method, which allows the determination of the acoustic pressure and velocity over the MPP surface. The validity of the solution is examined by comparing the predicted sound absorption curves with existing results provided by literature in a special case where a commonly used MPP absorber is involved. Numerical results based on the verified model reveal that, around the first cavity resonance, the irregular-shaped MPP absorber demonstrates similar sound absorption characteristics as that of a regular one having the same depth. However, the irregular-shaped MPP absorber can achieve a good/moderate sound absorption level in between the first and second resonances, as opposed to conventional MPP absorbers with constant cavity depth for which the sound absorption is usually negligible. Further investigation shows that this

phenomenon is caused by the local resonances occurring in the irregular-shaped cavity. This phenomenon can be further explored for improving the acoustic performance of MPP absorbers.

The transfer of L1 acoustic cues in the perception of L2 lexical stress

Qian Wang (University of Victoria), Tae-Jin Yoon (University of Victoria)

In second language (L2) acquisition, first language (L1) transfer is observed at different levels, e.g. syntax, morphology, and also phonology. It is proposed that L1 transfer also operates at the level of acoustics, i.e. the transfer of acoustic cue reliance. F0 is the most reliable cue in Chinese tone perception. We hypothesize that Chinese speakers' heavy reliance on F0 would be transferred to the acquisition of English lexical stress. The hypothesis was tested by conducting two experiments. The first experiment compared Chinese learners of English (CE) with native English speakers (NE) in stress judgment of manipulated nonsense tokens. Duration and intensity cues were kept the same on stressed and unstressed syllables while F0 was manipulated in five steps. Results showed that the F0 change affected NE's stress judgment to a great extent, but its effect is significantly stronger on CE's judgment. The second experiment used accented and unaccented real English words. On accented words, lexical stress is realized by a combination of three cues (i.e. F0, duration and intensity), but on unaccented words F0 is **not** reliably used for signaling stress. The result of the second experiment showed that CE resembled NE in stress perception on accented words where F0 is present. But CE behaved significantly worse than NE for unaccented words in which F0 information is absent. The experiments lead us to conclude that CE relies primarily on F0 cues in English stress perception and to suggest that there is an L1 transfer of cue reliance in L2 acquisition.

Learning Mandarin tones at sentence level through training: a pilot study

Xinchun Wang (California State University, Fresno)

Several previous studies have indicated that beginning level learners of Mandarin showed significant improvement in identification and production of individual Mandarin tones in isolation after taking 2-3 weeks of training. However, perception and production of Mandarin tones on larger linguistic units such as phrases and sentences still poses significant challenges to learners who made progress on isolated words. No training studies on Mandarin tones have explored the effect of training beyond word level. This pilot study investigates the effect of perception with production training for learning Mandarin tones at phrase and sentence levels. Seven beginning level Mandarin learners with different L1 backgrounds completed 6 hours of training on Mandarin phrases and sentences produced by multiple native Mandarin speakers using Kay Elemetric's Sona Speech II software with instant display of pitch contours along with speech output. The trainees had both auditory and visual input when they recorded their own productions of sentences. The trainees' productions of three Mandarin sentences were judged by four native Mandarin speakers in a rating task. In comparison with a control group of five who were taking the same Mandarin course but did not take the training, the trained groups' post test sentences improved significantly from the pretest while the control group did not show such improvement. The findings suggest that perception and production training with visual and auditory input is effective for learning Mandarin tones in a larger linguistic context. The results need to be replicated with larger scale studies in the future.

Effects of training modality on audio-visual perception of place of articulation and voicing in nonnative speech

Yue Wang (Simon Fraser University), Dawn Behne (Norwegian University of Science and Technology), Haisheng Jiang (Simon Fraser University)

Previous research has shown that although visual speech information facilitates native speech perception, the perception of nonnative visual cues remains difficult. This study examines the extent to which nonnative speech perception improves as a result of training with auditory (A) and visual (V) modalities. Native perceivers of Mandarin Chinese were trained to identify English fricatives of three visually distinct places of articulation (interdentals absent in Mandarin, labiodentals and alveolars common in both languages), as well as to identify the voicing distinctions of these fricatives (including the voiced category non-existent in Mandarin). Participants were randomly assigned to a control group or one of three 2-week (six sessions, 40 minutes/session) training groups with a different input modality: A, V, or AV. In pre- and posttraining tests, the participants were asked to identify these fricative categories with all three input modalities (A-only, V-only, and AV). Results show that post-training, the trainees reveal: (1) improvements corresponding to training type (e.g., the V-training group improves most for the V-only stimuli), (2) greater improvements for the familiar, but less visually distinct, alveolars than for the new interdentals, (3) greater improvements in the domain of place of articulation than voicing. Results are discussed in terms of the effects of speech input modality, phonetic features, and L1 experience on non-native AV speech learning.

Source tracking in an unknown ocean environment

Michael J. Wilmut (University of Victoria), Stan E. Dosso (University of Victoria)

This paper develops and compares two approaches, optimization and marginalization, to localizing and tracking a low-level acoustic source when ocean environmental properties are unknown. Both methods are based on a Bayesian formulation in which source and environmental parameters are considered random variables constrained by noisy acoustic data and by prior information on parameter values (e.g., physical limits for water-column and seabed properties) and on inter-parameter relationships (e.g., limits on horizontal and vertical source speed). Optimization is based on determining the source track and environmental parameters that maximize the posterior probability density (PPD). A key to solving this challenging problem efficiently is that the Viterbi algorithm is applied to compute the highest-probability source track for each environmental realization considered in the optimization-this provides the optimal track while requiring the optimization is applied only to environmental parameters. Marginalization involves integrating the PPD over unknown environmental parameters to represent source-track information as a series of joint marginal probability surfaces over range and depth. The Viterbi algorithm is applied to extract the optimal track from these surfaces. For realistic environmental models (e.g., more than a dozen unknown parameters), the integration is carried out using efficient Markov-chain Monte Carlo methods.

Ultrasound transmission through time-varying phononic crystals

Derek Wright (University of Toronto), Richard Cobbold (University of Toronto)

Phononic crystals are periodic media for which the spacing is comparable to the wavelength and in which the structure and material parameters govern the characteristics of wave propagation. Some of the effects already experimentally demonstrated are transmission band gaps, negative wave refraction, ultrasound focusing, and tunnelling. Phononic crystals are static in that their properties are fixed *a priori* by their design parameters, which limits their usefulness to predesigned narrow band operating conditions. We propose that the behaviour of phononic

crystals may be altered by changing the material parameters as a function of time. We have developed a new theory to analyze such a time-varying phononic crystal, which we call the time-varying transmission matrix method (TV-TMM). We demonstrate that the new method matches the predictions of our finite-difference time-domain (FDTD) simulator, and that the periodic solutions can be generated significantly faster than with the FDTD method. Our new method provides a closed-form solution which may provide insight into dynamically controlling the behaviour of a phononic crystal. Such insight may lead to new devices and applications. For example, we have shown that the characteristics of phononic crystal band-gaps, such as the attenuation and gap width, may be altered by varying the material parameters in time. As such, one could envision a dynamically controllable band-pass filter, or a dynamically variable mirror. By extending this theory to two and three dimensions, the effects of parameter variance on wave refraction can be examined, with potential implications for developing new ultrasound imaging methods.

Using acoustics to resolve a place controversy in Deg Xinag fricatives

Richard Wright (University of Washington), Sharon Hargus (University of Washington), Julia Colleen Miller (University of Washington)

Field linguists have disagreed about whether the Deg Xinag reflexes of the Proto-Athabaskan third person plural subject prefix * χ - (Leer 2000) and areal prefix * χ u- (Leer 2005) contain the uvular fricative [χ] or the glottal fricative [h]. In an earlier acoustic study of differences among the eight voiceless fricatives of Deg Xinag that can occur in stems (Wright, Hargus, and Miller 2005, in prep.) we found that [χ] and [h] differ significantly in skew and kurtosis but not center of gravity, lowest spectral peak, or standard deviation. In the present study, word list data comparing the reflex of PA prefixal * χ - (hereafter "x"), stem initial / χ /, and stem-initial /h/ were collected from seven speakers. The following vowel---either /ə/, /a/, or /o/---was controlled for. The same spectral measures were used as in our previous study, with the measurements then subjected to a repeated measures ANOVA. Center of gravity, standard deviation, skewness, and kurtosis all significantly distinguished /h/, on the one hand, from / χ / and "x", on the other. However, / χ / and "x" were not significantly different for these measures. Because "x" patterns with the uvular and not the glottal fricative, we interpret "x" as the uvular fricative. Acoustic measures can therefore be used to resolve some controversies which cannot be resolved on the basis of auditory impressions alone. With respect to prefix fricative place of articulation, Deg Xinag can be seen as conservative.

Mandarin speakers' productions of English vowels in real and pseudo words

Xianghua Wu (Simon Fraser University), Murray J. Munro (Simon Fraser University)

Previous research indicates that L2 vowel production is influenced by experience such that vowel accuracy tends to improve over time, though different learning trajectories can be observed across different vowels. In addition, L2 vowel intelligibility can be influenced by phonetic context, and preliminary findings suggest that word frequency may also play a role. In this investigation we examined a number of contributors to L2 vowel target accuracy by eliciting vowel productions from eight native speakers of Mandarin. The participants, who had lived in Canada for a mean of 2.4 years (range: 4 months to 6 years), produced English real and pseudo words in a delayed repetition task. Paired words were differentiated from each other by onset voicing, rhyme and word frequency, and the vowel targets under investigation were English /I U ʌ/, produced in CVC(C) context. Individual vowel productions were evaluated for intelligibility by a phonetically-trained listener, who identified the English vowel category closest to each production. The results indicate that voicing of the initial consonant had no effect on vowel target accuracy, while coda consonants had a strong effect. There was a weak tendency toward more target-like vowel

productions in real words than in pseudo-words. Length of residence in Canada had a differential impact on productions of real vs pseudo words. These findings will be discussed in terms of contemporary models of L2 acquisition.

Analysing coarticulation in Scottish English children and adults: an ultrasound study

Natalia Zharkova (Queen Margaret University), Nigel Hewlett (Queen Margaret University), William J. Hardcastle (Queen Margaret University)

There have been a number of studies which compared coarticulatory patterns in children and adults, but these studies have produced conflicting results, particularly with respect to anticipatory lingual coarticulation. Some previous research shows that children exhibit less coarticulation than adults; a similar amount; or more. This study uses innovative articulatory measures derived from ultrasound imaging, in order to establish how children's patterns of lingual coarticulation differ from adults', how the observed patterns may be explained and what is the nature and the degree of variability found in children and adults. The participants were five adults and five normally developing children aged 6 to 8 years, all speakers of Standard Scottish English. The data were CV syllables including all combinations of the following segments: /s/, /ʃ/, /i/, /u/ and /a/, in the carrier phrase "It's a ... Pam" (ten repetitions). Synchronised ultrasound and acoustic data were recorded using the Queen Margaret University ultrasound system. A new ultrasound-based measure of coarticulation has been developed; child and adult productions are compared according to this measure. The results will enable us to claim what, if any, differences occur between children and adults in coarticulation extent and within-speaker variability. Examination of individual variation will give us an opportunity to understand whether constraints on coarticulatory processes are different in children and adults, and to establish the nature of any differences between children and adults in the organizational structure of speech in terms of segments, syllables and words.