Musically Representative Test Signals for Interaural Cross-Correlation Coefficient Measurement

Short title: Test signals for cross-correlation measurement

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Summary

Typically, measurements that aim to predict perceived spatial impression of music signals in concert halls are performed by calculating the interaural cross-correlation coefficient (IACC) of a binaurally-recorded impulse response. Previous research, however, has shown that this can lead to results very different from those obtained if a musical input signal is used. The reasons for this discrepancy were investigated, and it was found that the overall duration of the source signal, its onset and offset times, and the magnitude and rate of any spectral fluctuations, have a very strong effect on the IACC. Two test signals, synthesised to be representative of a wide range of musical stimuli, can extend the external validity of traditional IACC-based measurements.

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1. Introduction

Spatial impression has long been considered an important factor in the perceived quality of concert halls [1]. As a result, numerous studies have been conducted to analyze its perceptual characteristics [2-4], and a consensus seems to have been reached that spatial impression can be sub-divided into two perceptual components: apparent source width (ASW) and listener envelopment (LEV) [4]. The interaural cross-correlation coefficient (IACC) has been used by some researchers as the basis for objective measurement of these components [5, 6]. It is traditionally measured using a binaurally-recorded impulse response [7]. Psychoacoustic research has indicated that a stimulus with an IACC close to 1 will sound relatively narrow. As its IACC decreases, however, the sound will be perceived to widen until a point at which it may separate into two spatially distinct components, one positioned at each ear [8, 9].

This paper addresses the problem that arises when impulse-response-based IACC measurements of an acoustic environment are used to predict spatial auditory attributes of musical stimuli in that environment. Its aim is to determine the relationships between various musical source signal parameters and the IACC, and to derive appropriate test signals that can be used to make IACC measurements that are perceptually and musically more relevant than those obtainable from impulse responses. The signal parameters considered are those which would change with variations in instrument timbre, performance style and choice of musical piece. Instrument location and directivity will also affect the IACC but these factors can be taken into account by making IACC measurements at many locations, using a variety of loudspeakers for test signal reproduction. Our focus is on choice of test signal, and the scope of this paper is limited to solo instruments only.

In many situations an impulse is an ideal test signal, being perfectly quantifiable and having a theoretically infinite bandwidth. In [10], however, it was shown that the result of a measurement based on an impulse response can be very different from that of one based on a tonal signal. This was partly traced back to the two signals' different spectra. Since the IACC (and hence perceived auditory width) depends on the interference between a direct sound signal and lateral reflections at a listener's ears [11], the frequency content of a source signal and any reflections will strongly determine the characteristics of such interaction effects [12]. Furthermore, Mason and Rumsey found that because of the short duration of an impulse little, if any, interaction between a direct sound and a set of reflections will occur. Consequently, it will be difficult to predict the effect of the interference that will result from longer musical tones. In this context, Griesinger [13] noticed that when making IACC measurements in concert halls, musical signals generally give lower values than impulses. Hence, the results of measurements performed on impulses may be somewhat misleading if the aim is to predict spatial impression of musical stimuli. As a solution to this problem, Morimoto and Iida [14] argued for the selection of a suitable musical phrase that could act as a standard test signal. Yet, they acknowledged that in view of the virtually infinite number of musical phrases to choose from, ensuring representativeness of a wide range of musical stimuli would be almost impossible to achieve.

The structure of this paper is as follows. In Sect. 2, the findings of published research in musical acoustics and timbre perception are used to derive a set of musical parameters and ranges worthy of subsequent testing. In Sect. 3, the effect of each of these parameters on the IACC is determined. Those parameters having a significant effect are then incorporated into the novel test signals synthesised and evaluated in Sect. 4.

2. Finer spectro-temporal signal characteristics

Previous research, then, has indicated that the differences between IACC measurements performed on impulsive and on musical signals are due to differences in source signal duration and frequency content. In this section, therefore, signal variations occurring in music and affecting these two characteristics are examined in order to determine their nature and magnitude, and to thereby derive a specification for synthetic stimuli covering the appropriate range of variations. These stimuli can then be evaluated with regard to their effect on the IACC.

2.1. Duration and amplitude envelope

No published research into musical note lengths seems to be available, but in the authors' experience a range of 250ms to 2s is typical (these figures correspond, for a tempo of 120bpm and a time signature of 4/4, to the lengths of eighth and whole notes, respectively). In terms of other duration-related parameters, Melka [15] conducted an investigation into the lengths of onsets of musical instrument signals. He found that for non-percussive instruments attack time depended strongly on pitch and performance style (i.e. whether a note was played softly or staccato), whereas for percussive instruments attack time was affected by the dynamics of a signal. On average, onset duration varied between a few milliseconds for percussion (e.g. 1ms for castanets) and a few hundred milliseconds for bowed string instruments (e.g. 255ms for a softly played double bass), with plucked and hit strings, woodwinds and brass falling in-between these two extremes. Looking at orchestral instruments only, Meyer [16] reported the shortest onsets for the double reed instruments (e.g. oboes or bassoons) – their staccato attacks ranging from 15ms to 35ms. Like Melka, he found that a softly played double bass produced the longest onset durations, which according to his data vary between ~150-400ms. Meyer also studied decay times, which were found to range from ~70ms (wind instruments) to several seconds (e.g. timpani or a piano played in the lower registers), but, similar to Melka's finding with respect to onset duration, performance style was also found to influence offset length. Overall, a range of onset lengths from 10ms to 250ms, and offset lengths from 70ms to 1.5s, seems typical.

2.2. Spectral content

Signal spectrum is affected by the harmonicity of musical tones. During the quasi-steady-state segments of notes the spectra of most non-percussive musical signals follow a harmonic series, an exception being the piano whose upper partials are generally tuned high [17]. Also, pizzicato notes produced by other string instruments (e.g. violins and violas) feature inharmonic spectra [18]. Where pitched instruments have inharmonic components in their steady-state spectra, these components are widely spaced. With the exception of the noise-like spectra of unpitched percussion then, the closest and widest spacing between adjacent spectral components in a Western musical signal is therefore likely to be one semitone and one harmonic increment respectively, e.g. ~26Hz and 440Hz for the pitch A4 on the equal-tempered scale [19].

Measurements of overall spectral shape give rise to the concept of spectral centroid or brightness [20]. The trumpet playing fortissimo, and the flute playing pianissimo, are the orchestral instruments richest and sparsest in overtones, respectively [17], and so their spectral slopes represent the extremes of brightness likely to be encountered. Analyses of these shapes from anechoic recordings of the note A4 reveal slopes of –2.6dB/harmonic for the trumpet and –7.8dB/harmonic for the flute.

The phase relationships between individual partials in typical music signal appear to be undocumented, and musical experience gives no clue as to likely values. In order to best reveal any possible relationship between phase and IACC then, the greatest possible extremes (0^{o} and 180^{o}) should be tested.

2.3. Frequency fluctuation

The natural evolution of a single musical note often involves a degree of spectral flux due to differing amplitude and frequency trajectories for each partial, e.g. [21-23]. More marked fluctuations in signal frequency content commonly result from vibratos, glissandi, trills and arpeggios. In terms of rates of change, there is broad agreement that slow vibratos, glissandi and trills occur at rates between 5Hz and 8Hz for both vocal and instrumental performances [24-29], whereas fast ones typically exhibit rates of change of between 11Hz and 14Hz [25, 27, 30]. For vibrato, a range of 6Hz to 12Hz can thus be considered typical; for glissando, a slide would typically be across between six and twelve notes per second; a typical (fast) trill will cycle at a rate of 7Hz between two notes spaced either a semitone [27, 31] or a whole-tone [25] apart (NB 7Hz cycling between 2 notes means 14 note transitions per second). The fastest arpeggio might cover 12 notes per second.

In terms of vibrato depth, Kuttner [31] reported average frequency deviations of 2% and 5.5% for small and large vibratos, respectively. Seashore [29] found deviations of 6% for vocal and 3% for instrumental performances, whilst Prame [28] measured variations between 3% and 7%.

2.4. Synthetic stimuli

In light of the preceding sections, the synthetic stimuli detailed in Table I can be considered to cover the full range of variations of duration and frequency-related parameters typically encountered in musical signals. Stimulus 1 constitutes the baseline case, because it is used (in the following section) to investigate several parameters, and because all the other stimuli are derived from it.

3. Effect of finer physical parameters on the IACC

The stimuli defined at the end of the preceding section can now be used to determine the effect of each parameter on the IACC measured in a live acoustic environment. Variations having no significant effect need not be included in the final test signals. In this way, the minimal representative set of signal features can be determined.

3.1. Synthesis and auralisation of stimuli

Each stimulus is generated as a variation on an eight-harmonic complex tone with a fundamental frequency of 440Hz, the variations being dictated by the detail given in Table I. Since the IACC is strongly affected by the interaction between a direct sound and lateral reflections (see Sect. 1), a change in IACC due to one of the investigated parameters could, in principle, be masked by a particular interference effect. To avoid such uncertainty each stimulus is evaluated in two different acoustic environments, both created as *CATT-Acoustic* models. Model parameters are detailed in Tables II and III. Each stimulus is auralised in each of these models. For both models, an omnidirectional source is placed at a representative stage position that is 4m from the front wall, 1m to the right of the centerline and at a height of 1.7m (see Figs. 1 and 2). The binaural receiver is positioned 3m to the left of the centerline and 4m from the back wall in the case of Model I. In the case of Model II, the receiver is placed 1m to the left of the centerline and 12m from the back wall. In both cases, the receiver has a height of 1.3m and is facing towards the source.

3.2. Measurement of auralised stimuli and evaluation of their effect on the IACC

The IACC of a sound field can vary significantly over time, which can lead to changes in perceived auditory width [32, 33]. Although averaged measures such as $IACC_{E3}$ are useful and can sometimes correlate well with listener preference [5, 34], it can often be valuable to consider the time-varying IACC in order to reveal potential problems in an auditorium such as inappropriate differences between perceived source width and perceived environment width, or radically different perceived widths of different musical notes. To enable examination of the effects of the chosen parameters on the IACC as a function of time, the auralised stimuli are analyzed using a 'running' IACC measurement based on the normalised cross-correlation (NCC) function. The NCC is given by Eq. 1, where x and y are the two signals whose correlation is to be calculated, t_1 and t_2 define the window over which the correlation is to be measured (and are set here to give a window length of 50ms) and τ is a varying offset between the two signals.

$$NCC(\tau) = \left\{ \int_{t_1}^{t_2} x(t)y(t+\tau)dt \middle/ \left(\int_{t_1}^{t_2} x^2(t)dt \int_{t_1}^{t_2} y^2(t)dt \right)^{1/2} \right\} \tag{1}$$

The IACC is taken as the maximum absolute value across the range of τ, as shown in Eq. 2.

$$IACC = \left|NCC(\tau)\right|_{\max}, \text{ for } -1\text{ms} < \tau < +1\text{ms} \tag{2}$$

3.2.1. Effect of signal duration and on- and offset length

As would be expected for different acoustic environments, the IACC measured in Model I is markedly different from that measured in Model II. This is illustrated, for Stimulus 1, in Figs. 3a and 3b. The general form of the running IACC generated from a long spectrally-complex and spectrally-invariant signal is that it begins with a value of 1, when the direct sound only is present, falls, as soon as the first lateral reflection arrives, to a smaller value on which it settles during the steady-state portion of the sound, and then fluctuates greatly during the reverberation-only portion of the received signal. For Stimulus 2 (the shorter duration signal), however, this pattern is not evident; in particular, there is no settled region (Fig. 4). With increased onset length (Fig. 5) the transition, for a long signal, to the settled region is much smoother than with a shorter onset. When offset length increases (Fig. 6), the IACC trace corresponding to the region of the reverberant sound becomes much smoother.

The effects of signal duration are explained by the interference patterns of differing lengths between a direct sound and its reflections, as discussed earlier. The effects of on- and offset time are more likely to be due to the additionally created sidebands resulting from amplitude modulation occurring at different rates [35], which, by altering the stimuli's spectra, give rise to different interference patterns.

3.2.2. *Effect of spectral density, brightness and relative phase*

Increasing signal spectral density decreases the IACC during the settled region in Model I, but not in Model II. It is possible that this effect is due not to spectral density itself, but to the presence or absence of one or more specific frequency components whose wavelengths correspond to the lengths of specular reflection paths in a way which creates significant constructive or deconstructive interference at either ear of the binaural receiver. Such components could lead to drastic changes in IACC in either direction. A further test using variation of the two room models where all surface diffusion factors (see Table III) are set to 100% supports this hypothesis, since there is now no significant effect on the average IACC during the settled region. For diffuse rooms then, spectral density appears to have no significant effect on the IACC. For rooms with strong specular reflections, however, a dramatic but chaotic effect can result from particular relationships between specific frequency components, the dimensions of the room, and the source and receiver positions chosen. Informal experimentation shows that a harmonic series whose fundamental frequency (starting at ~50Hz) sweeps upward at 15Hz/s allows for the detection of this effect in the resulting IACC.

The results for brightness show no significant effect and for phase relationships no effect at all.

3.2.3. *Effect of vibrato, arpeggio, glissando and trill*

The addition of vibrato to a signal has a very dramatic effect, creating large fluctuations in IACC throughout the measurement period (see Fig. 7). Generally speaking, greater magnitudes and rates of frequency change cause greater and faster fluctuations in IACC, for vibrato, arpeggio, trill and glissando. For all of these parameters, however, the effect on the average IACC seems unpredictable.

3.3. Summary

From the above, it appears that there are four signal parameters, which have a significant effect on the IACC and which must therefore be included as variables in a minimal musically representative test signal set. These are signal duration (longer signals leading to a settled IACC region), onset time (longer onsets causing a smoother transition to the settled region), offset time (longer offsets causing a smoother IACC trace during the reverberant sound), and frequency modulation (causing corresponding fluctuations in the IACC). A slowly-sweeping frequency component is also required for detection of chaotic effects in a room with strong specular reflections.

4. Specification, creation and evaluation of pseudo-music test signals

The findings from Sect. 3 are now used to provide the basis for the specification and creation of a minimal representative set of music-like test signals, which are then evaluated in modelled acoustic environments.

4.1. Specification and creation of test signals

To be able to excite reverberant decays by both abrupt and gradual signal offsets, two offsets, and therefore two signals, are required. The important variations identified in Sect. 3 are covered by:

- Signal I : a signal consisting of a relatively long time segment with constant frequency content and a long onset, a comparatively slow/small frequency sweep, and a second relatively long time segment with constant frequency content and a long offset; and
- Signal II : a signal consisting of a relatively short time segment with constant frequency content and a short onset, a comparatively fast/large frequency sweep, and a second relatively short time segment with constant frequency content and a short offset.

As found in Sect. 2, offset lengths of 1.5s and 70ms can be considered representative of slowly and rapidly decaying musical signals, and so these are employed in Signals I and II, respectively. Likewise, onset times of 10ms and 250ms are used, since these are typical extremes. Durations of 2s and 0.25s are typical of long and short notes, and so these values are used for those segments of the test signals featuring constant frequency contents.

In order to be useful in multi-band IACC-based measurements [36], the test signals should cover most of the audible range. In addition, it is necessary to have at least two partials in each critical band within that range, so that interaural decorrelation of the direct signal can occur; as discussed in detail in [10], a single sine tone will give an IACC of 1 once the sound field has reached a steady state, whereas a spectrally complex musical tone will not. To meet these requirements whilst maintaining musical relevancy, the two notes of A1 and A3 (having fundamental frequencies of 55Hz and 220Hz on the equal-tempered scale) are used as the starting point for the two signals. Each signal is based on the sum of harmonics 1 to 34 of A1 (55Hz), plus harmonics 9 to 34 of A3 (220Hz). In order to ensure sufficient energy in each analysis band, each frequency component is included at the same level.

In terms of frequency modulation, the two requirements are for a 15Hz/s sweep to reveal the effect of problematic specular reflections, plus a more realistic musical pitch variation. An appropriate musical sweep rate is one octave per second, representative of a fast glissando (see Sect. 2). Since the signal is a rich harmonic complex, a one-octave sweep is sufficient to ensure coverage of all frequencies.

To allow reproduction at a sufficiently high level without causing significant clipping distortion in the transducer the signals are low-pass filtered at -3dB/octave, resulting in a similar spectral slope to that of pink noise. The final test signal specifications are therefore as defined in Table IV and illustrated spectrographically in Figs. 8a and 8b.

4.2. Evaluation of test signals

Measurement of the IACC in both modelled acoustic environments using the following sound sources reveals the degree to which the new signals provide a more representative result than the traditional impulse:

a) anechoic recording of several notes played on a trumpet;

b) Signal I;

c) Signal II;

d) impulse.

Results are similar across the two acoustic models and, in terms of size and reverberation time, Model II is more representative than Model I of a typical concert auditorium, so only the Model II results are presented here.

The trumpet IACC (Fig. 9) has an average of ~0.4 with fluctuation between ~0.2 and ~0.7. Signal I (Fig. 10) shows a similar result, but with additional settled portions. Signal II (Fig. 11) gives a result even more similar to that provided by the trumpet. The impulse (Fig. 12), however, bears no resemblance at all, showing an average IACC of ~0.7 with relatively little fluctuation but a gradual drift to an IACC of 1 toward the end of the measurement period. IACCs measured using other musical signals provide similar results to that of the trumpet recording. IACCs measured using the same stimuli convolved with binaural room impulse responses recorded in two real acoustical environments (with floor areas of ~240m^2 and ~193m^2, and RT60s of ~1.1s and ~1.7s, respectively) confirm the external validity of these findings.

4.3. Discussion

The synthesised test signals appear to fulfill their intended role of providing a more musically-representative means to perform IACC measurements. In view of the huge number of physical characteristics in terms of which musical instrument signals can differ and change, however, it is possible that there will be situations when Signals I and II will not perform so well. One particular type of music signal that is not well catered for is a very short, transient one. As a direct consequence of their relatively long durations and harmonic spectra, the two synthesised signals are representative of non-percussive, tonal music. Conversely, short percussive signals such as drum hits have properties much more similar to those of impulses. However, by performing IACC measurements using an impulse *as well as* using the two new music-like test signals, a very broad range of musical stimuli can be covered. The (additional) use of the two synthetic signals will therefore greatly extend the external validity or musical representativeness of traditional IACC measurements.

5. Conclusions

Citing existing research, Sect. 1 established that an IACC measured using an impulse will give different results from one measured using musical source material, due to differing frequency contents and durations. In Sect. 2, a review of musical and timbral literature determined the nature and magnitude of variations in typical musical signals affecting these factors and led to specific requirements (Table I) for a set of synthetic stimuli that would allow evaluation of the effect of each of these variations on the IACC. A series of tests with synthesised stimuli meeting these requirements revealed, in Sect. 3, that the IACC is affected significantly by on- and offset time, duration and frequency modulation (such as that introduced by vibratos, glissandi, trills or arpeggios). Additionally, it was established that in a room producing strong specular reflections the presence of particular frequency components can have a dramatic but essentially chaotic effect on the IACC. In Sect. 4, two musically-representative test signals (Table IV) were synthesised to cover all pertinent variations. Evaluation of these signals showed them to provide IACC measurements much more similar to those produced using musical source material than impulse-based measurements, both for modelled and natural acoustic environments.

The limitation of these new test signals lies in the fact that they are representative of non-percussive, finite bandwidth, tonal music only. Short percussive hits (e.g. drums) are not well catered for. However, using the new signals together with an impulse-based measurement will provide a set of IACC readings that are broadly musically representative and likely to predict well the perception of ASW and LEV which would result from a musical performance in a given environment. One further implication of the results of this work is that IACC measurements in halls producing strong specular reflections should be treated with caution, since they can be very sensitive to particular combinations of individual frequency components, room dimensions, and source and receiver locations.

Likely avenues for further research include evaluation of the new test signals with the full range of IACC-based measurements currently in use, and investigation into possible methods of modeling the chaotic effects noted above.

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References

[1] L. Beranek: Concert and Opera Halls – How They Sound. Acoustical Society of America, Woodbury, 1996.

[2] Y. Ando: Architectural Acoustics: Blending Sound Sources, Sound Fields, and Listeners. Springer-Verlag, New York, 1998.

[3] W.d.V. Keet: The influence of early lateral reflections on the spatial impression. Proceedings of the 6th International Conference on Acoustics **(**1968) E53-E56.

[4] M. Morimoto & Z. Maekawa: Auditory spaciousness and envelopment. Proceedings of the 13th International Conference on Acoustics **(**1989) 215-218.

[5] T. Hidaka, L. Beranek & T. Okano: Interaural crosscorrelation, lateral fraction, and low- and high-frequency sound levels as measures of acoustical quality in concert halls. Journal of the Acoustical Society of America **98 (**1995) 988-1007.

[6] R. Mason, T. Brookes & F. Rumsey: Development of the interaural cross-correlation coefficient into a more complete auditory width prediction model. Proceedings of the 18th International Congress on Acoustics (ICA) **IV (**2004) 2453-2456.

[7] ISO: Acoustics – Measurement of the reverberation time of rooms with reference to other acoustical parameters. ISO 3382 **(**1997) .

[8] J. Blauert & W. Lindemann: Spatial mapping of intracranial auditory events for various degrees of interaural coherence. Journal of the Acoustical Society of America **79 (**1986) 806-813.

[9] R.I. Chernyak & N.A. Dubrovsky: Pattern of the noise images and the binaural summation of loudness for the different interaural correlation of noise. Proceedings of the 6th International Conference Acoustics **(**1968) A53-A56.

[10] R. Mason, T. Brookes & F. Rumsey: The effect of various source signal properties on measurements of the interaural cross-correlation coefficient. Acoustical Science and Technology **26(2) (**2005) 102-113.

[11] R. Mason: Elicitation and measurement of auditory spatial attributes in reproduced sound. Doctoral Thesis, Guildford, University of Surrey (2002).

[12] R. Mason & F. Rumsey: Interaural time difference fluctuations: Their measurement, subjective perceptual effect, and application in sound reproduction. Proceedings of the Audio Engineering Society 19th International Conference on Surround Sound **(**2001) 252-271.

[13] D. Griesinger: The psychoacoustics of apparent source width, spaciousness and envelopment in performance spaces. Acustica **83 (**1997) 721-731.

[14] M. Morimoto & K. Iida: A practical evaluation method of auditory source width in concert halls. Journal of the Acoustical Society of Japan **(E)16 (**1995) 59-69.

[15] A. Melka: Messungen der Klangeinsatzdauer bei Musikinstrumenten (Measurements of tone-rise time of musical instruments). Acustica **23 (**1970) 108-117.

[16] J. Meyer: The sound of the orchestra. Journal of the Audio Engineering Society **41 (**1993) 203-213.

[17] J. Meyer: Acoustics and the Performance of Music. Verlag Das Musikinstrument, Frankfurt/Main, 1978.

[18] J.C. Brown: Frequency ratios of spectral components of musical sounds. Journal of the Acoustical Society of America **99 (**1996) 1210-1218.

[19] T.D. Rossing, F.R. Moore & P.A. Wheeler: The Science of Sound. Addison Wesley, San Francisco, 2002.

[20] J.W. Beauchamp: Synthesis by spectral amplitude and 'brightness' matching of analyzed musical instrument tones. Journal of the Audio Engineering Society **30 (**1982) 396-406.

[21] J.M. Grey: Multidimensional perceptual scaling of musical timbres. Journal of the Acoustical Society of America **61 (**1977) 1270-1277.

[22] A. Horner, J.W. Beauchamp & R. So: Detection of random alterations to time-varying musical instrument spectra. Journal of the Acoustical Society of America **116 (**2004) 1800-1810.

[23] S. McAdams, J.W. Beauchamp & S. Meneguzzi: Discrimination of musical instrument sounds resynthesized with simplified spectrotemporal parameters. Journal of the Acoustical Society of America **105 (**1999) 882-897.

[24] S. Ando & K. Yamaguchi: Statistical study of spectral parameters in musical instrument tones. Journal of the Acoustical Society of America **94 (**1993) 37-45.

[25] J.C. Brown & P. Smaragdis: Independent component analysis for automatic note extraction from musical trills. Journal of the Acoustical Society of America **115 (**2004) 2295-2306.

[26] D.E. Hall: Musikalische Akustik – Ein Handbuch (Musical Acoustics – A Handbook). Schott Musik International, Mainz, 1997.

[27] D. Moelants: Temporal aspects of instrumentalists' performance of tremolo, trills and vibrato. Proceedings of the International Symposium on Musical Acoustics (2004) 281-284.

[28] E. Prame: Vibrato extent and intonation in professional Western lyric singing. Journal of the Acoustical Society of America **102 (**1997) 616-621.

[29] C.E. Seashore: Psychology of Music. Dover Publications, New York, 1967.

[30] J. Meyer: Zur klanglichen Wirkung des Streicher-Vibratos (On the tonal effect of string vibrato). Acustica **76 (**1992) 283-291.

[31] F.A. Kuttner: Vibrato, tremolo and beats. Journal of the Audio Engineering Society **11 (**1963) 372-374.

[32] S.E. Boehnke, S.E. Hall & T. Marquardt: Detection of static and dynamic changes in interaural correlation. Journal of the Acoustical Society of America **112 (**2002) 1617-1626.

[33] K. Iida & M. Morimoto: Basic study on sound field simulation based on running interaural cross-correlation. Applied Acoustics **38 (**1993) 303-317.

[34] L. Beranek: Subjective rank-orderings and acoustical measurements for fifty-eight concert halls. Acta Acustica united with Acustica **89(3) (**2003) 494-508.

[35] H.S. Black: Modulation Theory. D. Van Nostrand Company, Princeton, New Jersey, 1953.

[36] R. Mason, T. Brookes & F. Rumsey: Frequency dependency of the relationship between perceived auditory source width and the interaural cross-correlation coefficient for time-invariant stimuli. Journal of the Acoustical Society of America **117(3) (**2005) 1337-1350.

Figures

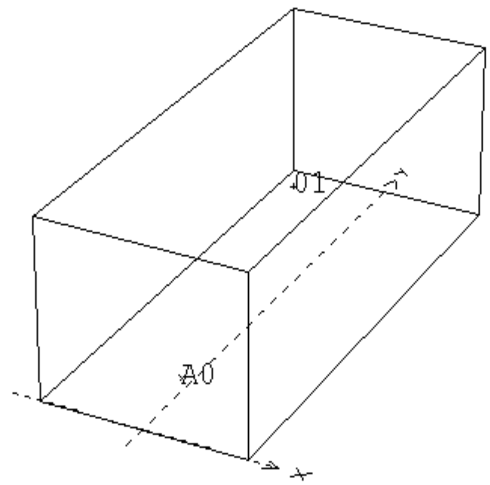

Figure. 1. Diagram of Model I with the source position being indicated by ‘A0’ and the receiver by ‘01’

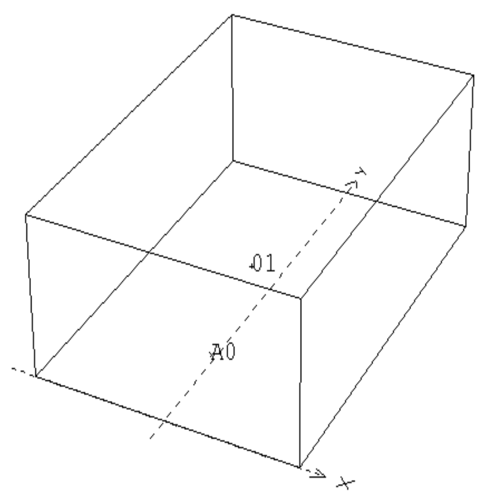

Figure. 2. Diagram of Model II with the source position being indicated by ‘A0’ and the receiver by ‘01’

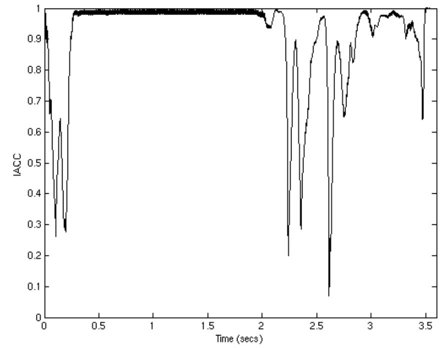

Figure. 3a. IACC measurement of Stimulus 1 (signal duration – long; onset length – short; offset length – short; spectral density – low; brightness – bright; relative phase – in-phase) convolved with the binaural room impulse response from Model I

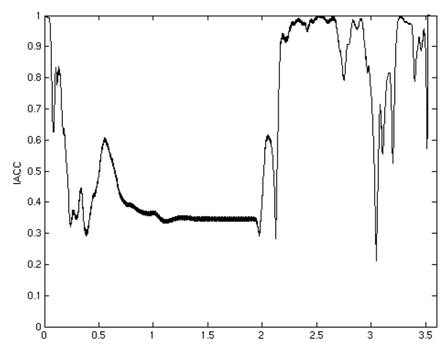

Figure. 3b. IACC measurement of Stimulus 1 (signal duration – long; onset length – short; offset length – short; spectral density – low; brightness – bright; relative phase – in-phase) convolved with the binaural room impulse response from Model II

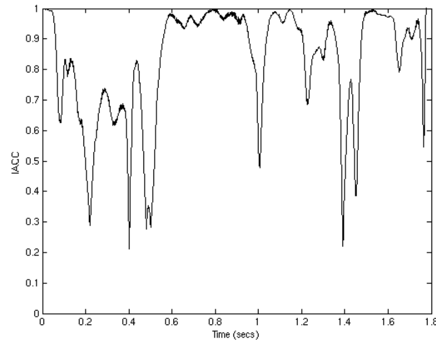

Figure. 4. IACC measurement of Stimulus 2 (signal duration – short) convolved with the binaural room impulse response from Model II

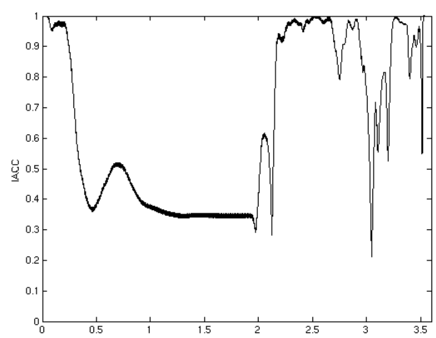

Figure. 5. IACC measurement of Stimulus 3 (onset length – long) convolved with the binaural room impulse response from Model II

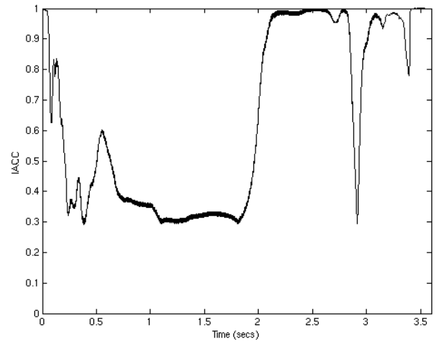

Figure. 6. IACC measurement of Stimulus 4 (offset length – long) convolved with the binaural room impulse response from Model II

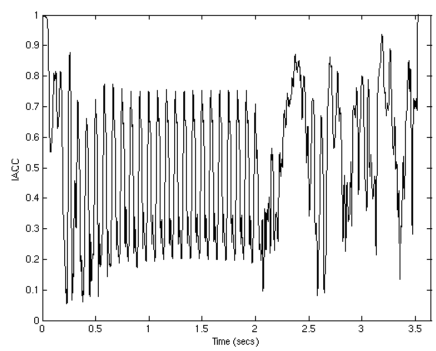

Figure. 7. IACC measurement of Stimulus 9 (vibrato – fast/large) convolved with the binaural room impulse response from Model II

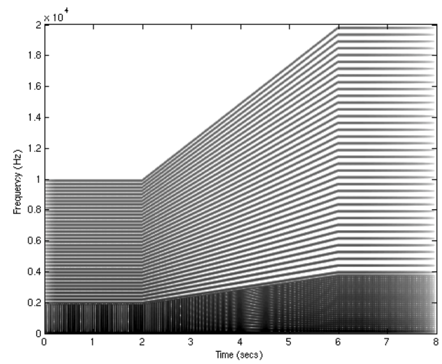

Figure. 8a. Spectrogram of Signal I

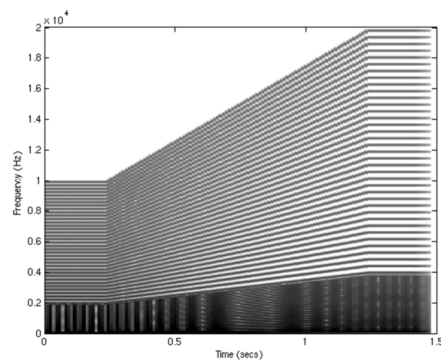

Figure. 8b. Spectrogram of Signal II

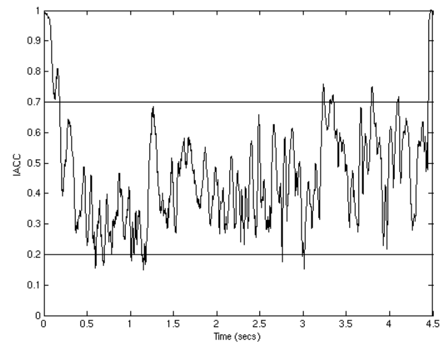

Figure. 9. IACC measurement of trumpet recording convolved with the binaural room impulse response from Model II

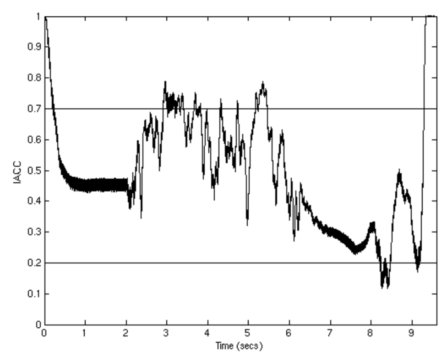

Figure. 10. IACC measurement of Signal I convolved with the binaural room impulse response from Model II

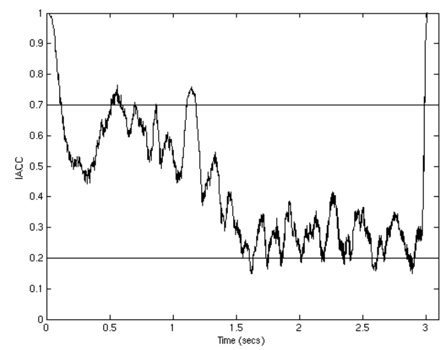

Figure. 11. IACC measurement of Signal II convolved with the binaural room impulse response from Model II

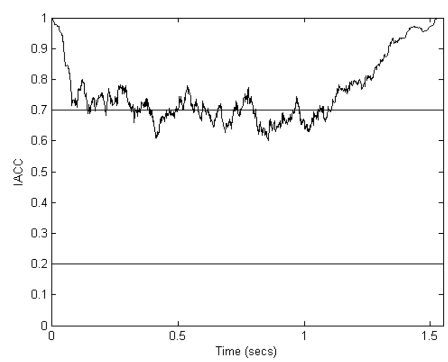

Figure. 12. IACC measurement of the binaural room impulse response from Model II

Tables

#	Investigated Parameter(s)	Parameter Value(s)
	Signal duration – long	**duration = 2s**
	Onset length – short	**onset length = 10ms**
1	**Offset length – short**	**offset length = 70ms**
	Spectral density – low	**partial frequencies =** $f_n(t) = f_n$ **[**n **= 1 to 8]**
	Brightness – bright	f_n **level =** f_{n-1} **level - 2.6dB**
	Relative phase – in-phase	**starting phase = 0º for all partials**
2	**Signal duration – short**	**duration = 0.25s**
3	**Onset length – long**	**onset length = 250ms**
4	**Offset length – long**	**offset length = 1.5s**
5	**Spectral density – high**	**partial frequencies =** $f_1 + n \times 25\text{Hz}$ **[**n **= 0 to 7]**
6	**Brightness – dull**	f_n **level =** f_{n-1} **level – 7.8dB**
7	**Relative phase – out-of-phase**	**starting phase of** $f_n = n \times 180^{\circ}$
8	**Vibrato – slow(6Hz)/small(3%)**	$f_n(t) = f_n \times (1 + 0.03\sin 12\pi t)$
9	**Vibrato – fast(12Hz)/large(6%)**	$f_n(t) = f_n \times (1 + 0.06\sin 24\pi t)$
10	**Glissando – slow/small**	$f_n(t) = f_n \times (t/2 + 1)$ $[t = 0 \text{ to } 1\text{s}]$
11	**Glissando – fast/large**	$f_n(t) = f_n \times (t + 1)$ $[t = 0 \text{ to } 1\text{s}]$
12	**Arpeggio – slow/small**	$f_n(t) = f_n \times 2^{i/12}, i/7 \le t < (i+1)/7, i \in \Im$
13	**Arpeggio – fast/large**	$f_n(t) = f_n \times 2^{i/12}, i/13 \le t < (i+1)/13, i \in \Im$
14	**Trill (wholetone at 7Hz)**	$f_n(t) = \left\{ \begin{array}{ll} f_n & ,\sin 14\pi t \ge 0 \\ f_n \times 2^{1/6} & ,\sin 14\pi t < 0 \end{array} \right\}$

Table I: Created stimuli, corresponding parameters and parameter values, where f_n = start frequency of n^{th} harmonic and $f_n(t)$ = frequency at time t. f_1 = 440Hz for all stimuli. Unspecified parameters for stimuli 2 to 14 are as specified for stimulus 1.

	Dimensions	Volume	RT60
Model I	**10m (W) by 24m (D) by 8m (H)**	**1920m^3**	**~1.3s**
Model II	**17m (W) by 25m (D) by 9m (H)**	**3825m^3**	**~1.6s**

Table II: Dimensions, volumes and RT60s of Models I and II

Surface	Absorption Factors (%) for Octave Bands from 125Hz to 4kHz	Diffusion Factors (%) for Octave Bands from 125Hz to 4kHz
Floor (audience)	**40, 50, 60, 70, 80, 80**	**30, 40, 50, 60, 70, 80**
All other (wood)	**15, 13, 10, 9, 8, 7**	**30, 30, 30, 30, 30, 30**

Table III: Surface absorption and diffusion characteristics employed for Models I and II

	Section 1: Steady I		Section 2: Sweep	Section 3: Steady II	
	Attack	Duration	Duration	Duration	Decay
Signal I	250ms	2s	4s	2s	1.5s
Signal II	10ms	250ms	1s	250ms	70ms
Spectral content	Harmonics 1-34 of 55Hz and harmonics 9-45 of 110Hz		Linear upward octave sweep	Harmonics 1-34 of 110Hz and harmonics 9-45 of 220Hz	
Spectral slope	-3dB per octave				

Table IV: Specification of the two final test signals (Signals I and II)