"Listening Tests Performed Inside a Virtual Room Acoustic Simulator"

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GOALS

This paper describes the processing required for obtaining a realistic audible sound reproduction and how to perform good listening tests.

The basic tool is a convolution procedure between “dry” music and the impulse responses measured (or numerically simulated) of a theatre.

- The overall process is usually known as “Auralization”, and traditionally it was performed through the binaural technology (headphones reproduction)

- Here the process is generalized to many more loudspeaker-based reproduction systems: Stereo, Binaural, Stereo Dipole, Dual Stereo Dipole, Ambisonics, Ambiophonics.
Theatre la Fenice, Venice

- The first theatre was realised in 1792 by Gian Antonio Selva, after the burning of Teatro San Benedetto.
- In December 1836 the theatre burned down again and was rebuilt by G. and T. Meduna the year after.
- The theatre was closed in 1995 for maintenance; it had to open again in February 1, 1996, but it burned two days before (January 29, 1996).
- A few weeks before the fire, L. Tronchin measured binaural impulse responses.
Acoustical measurements

- Measurements were performed by L. Tronchin in November-December 1995 with a modified gun (with omnidirectional diffuser) and binaural microphones.
- The goal was to analyze some acoustical problems about intercommunication between orchestra pit and stage and to gather information for designing the orchestra shell.
- The data were processed with the software Aurora, which had been developed just 2 months before.
Acoustical measurements

- In 27 positions a series of binaural impulse responses (with gun shots) was recorded
- Each recording is consequently a stereo file at 16 bits, 48 kHz
- During the measurements the room was perfectly fitted, whilst the stage was empty (no scenery)
Impulse Responses of La Fenice

Point n. 12
Auralization Examples – La Fenice

Overture alle Nozze di Figaro di Mozart
  • Dry music
  • Convolution with experimental I.R. (pt. 12)
  • Convolution with computer-simulated IR

Preludio al primo atto della Traviata di G. Verdi
  • Dry music
  • Convolution with experimental I.R. (pt. 12)
  • Convolution with computer-simulated IR

Stop
The Past
A PRELIMINARY SUBJECTIVE TEST

Test: questionnaires compiled during headphones listening of several tracks played in 5 different theatres

Italian theatres chosen:

- Opera theatres
  - Teatro Regio (Parma)
  - Teatro Valli (Reggio E.)

- Auditoria
  - Paganini (Parma)
  - Sala 700 (Roma)

- Historical theatres
  - Teatro Olimpico (Vicenza)
Track used for the test: anechoic tracks auralized with binaural Impulse Response of 5 theatres. Acquisition of Impulse Response made with source on the stage and receiver placed between 5th and 6th of every room.

Teatro Regio (Parma)  Auditorium Paganini (Parma)
Teatro Regio in Parma (Italy)

$T_{20} = 1.11 \text{ s}$
Teatro Valli, Reggio Emilia, Italy

![Acoustical Parameters according to ISO3382](image)

\[T_{20} = 1.55 \text{ s}\]
Paganini Auditorium, Parma, Italy

\[ T_{20} = 2.08 \text{ s} \]
S. Cecilia Auditorium 700, Rome, Italy

**Acoustical Parameters according to ISO 3382**

![Graph showing acoustical parameters with T20 = 2.04 s]

<table>
<thead>
<tr>
<th>Frequency (Hz)</th>
<th>31.5</th>
<th>63</th>
<th>125</th>
<th>250</th>
<th>500</th>
<th>1k</th>
<th>2k</th>
<th>4k</th>
<th>8k</th>
<th>16k</th>
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<td>StrnGrn (dB)</td>
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<td>D50 (dB)</td>
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<td>0.6</td>
<td>2.0</td>
<td>0.7</td>
<td>0.8</td>
<td>0.7</td>
<td>2.3</td>
<td>5.2</td>
<td>5.9</td>
<td>4.5</td>
<td>4.3</td>
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<td>D50 (%)</td>
<td>51.2</td>
<td>67.9</td>
<td>53.4</td>
<td>61.7</td>
<td>51.1</td>
<td>55.6</td>
<td>65.2</td>
<td>77.7</td>
<td>77.1</td>
<td>95.0</td>
<td>73.5</td>
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<td>0.96</td>
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<td>1.00</td>
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<tr>
<td>rT20</td>
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<td>0.98</td>
<td>0.99</td>
<td>1.00</td>
<td>1.00</td>
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<td>1.00</td>
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<td>1.00</td>
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<td>1.00</td>
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<tr>
<td>T30 (s)</td>
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<td>1.75</td>
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<td>2.04</td>
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<td>1.45</td>
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<td>0.44</td>
<td>1.72</td>
<td>1.60</td>
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<td>0.98</td>
<td>0.99</td>
<td>1.00</td>
<td>1.00</td>
<td>1.00</td>
<td>1.00</td>
<td>1.00</td>
<td>1.00</td>
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<td>1.00</td>
<td>1.00</td>
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<tr>
<td>Tuser (s)</td>
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<td>2.77</td>
<td>2.00</td>
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<td>2.05</td>
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<td>1.82</td>
<td>1.48</td>
<td>0.93</td>
<td>0.23</td>
<td>1.64</td>
<td>1.60</td>
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<tr>
<td>rTuser</td>
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<td>0.99</td>
<td>0.97</td>
<td>0.99</td>
<td>1.00</td>
<td>1.00</td>
<td>1.00</td>
<td>1.00</td>
<td>0.97</td>
<td>1.00</td>
<td>1.00</td>
<td>1.00</td>
</tr>
</tbody>
</table>
Teatro Olimpico, Vicenza, Italy

\[ T_{20} = 2.63 \text{ s} \]
Acoustical properties of the theatres

Reverberation Time T20 of 5 Italian theatres

Reverberation Time T20 (s) vs Frequency (Hz) for the following theatres:
- Paganini
- Olimpico
- Roma 700
- Valli
- Regio
Acoustical properties of the theatres
According to ISO 3382 standard

<table>
<thead>
<tr>
<th>Param.</th>
<th>Regio</th>
<th>Valli</th>
<th>Paganini</th>
<th>Roma-700</th>
<th>Olimpico</th>
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<tr>
<td>C50 [dB]</td>
<td>1.82</td>
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<td>-2.45</td>
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<td>0.83</td>
<td>1.20</td>
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<tr>
<td>D50 [%]</td>
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<td>84</td>
<td>42</td>
<td>37</td>
<td>50</td>
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<tr>
<td>Ts [ms]</td>
<td>48</td>
<td>28</td>
<td>115</td>
<td>144</td>
<td>110</td>
</tr>
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<td>EDT [s]</td>
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<td>1.26</td>
<td>2.09</td>
<td>1.98</td>
<td>2.43</td>
</tr>
<tr>
<td>T20 [s]</td>
<td>1.10</td>
<td>1.44</td>
<td>2.22</td>
<td>1.99</td>
<td>2.65</td>
</tr>
<tr>
<td>T30 [s]</td>
<td>1.11</td>
<td>1.55</td>
<td>2.24</td>
<td>1.99</td>
<td>2.64</td>
</tr>
<tr>
<td>LF</td>
<td>0.10</td>
<td>0.12</td>
<td>0.22</td>
<td>0.12</td>
<td>0.18</td>
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<td>0.88</td>
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<td>0.70</td>
<td>0.81</td>
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<td>TB</td>
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<td>1.70</td>
<td>1.20</td>
<td>1.11</td>
<td>0.91</td>
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<tr>
<td>BR</td>
<td>1.27</td>
<td>1.41</td>
<td>1.17</td>
<td>0.94</td>
<td>0.72</td>
</tr>
</tbody>
</table>
Sound Samples

Chosen in order to evidence the difference between rooms designed for opera and rooms for purely orchestral music:

ORCHESTRAL:
- G.Verdi, Preludio al Primo Atto de “la Traviata”
- W.A. Mozart, Overture de “le Nozze di Figaro”
- Strauss, “Pizzicate Polka”

VOCAL:
- Mozart, “Cosi’ fan tutte” (voice and piano)
- Tosti, “Non t’amo più” (voice and piano)
- “My Funny Valentine” (jazz, voice solo)
Listening seat in Casa della Musica (Parma-Italy):

**Instrumentation:**
- A liquid cooled Computer (*Futureclient*)
- Open dynamic headphones
  *Sennheiser HD 580 Precision*
- Audio-pro Subwoofer for very low frequencies (18-50 Hz)
Headphones

- Open dynamic headphones: *Sennheiser HD 580 Precision*
- Digital equalizing filters are employed for compensating the frequency response of the headphone+dummy head
- The computation of these inverse filters revealed to be very important for the “transparency” of the reproduction chain
Theory of inverse filters

Single input, single output system

Signal source (CD) → Filter → Amplifier → Loudspeaker

Microphone

Block diagram

Input signal: $x(\tau)$
Filtering coefficients: $f(\tau)$
System’s Impulse Response (Transfer function): $h(\tau)$
Output signal: $y(\tau)$
Combined transfer function

As all the stages are linear:

\[ y(i) = x(i) \otimes f(j) \otimes h(l) \]

The goal of the filter is to “equalize” – this means to make the output \( y(i) \) to be equal to the input \( x(i) \). This is obtained if:

\[ f(j) \otimes h(l) \Rightarrow \delta(i) \]

In which \( \delta(i) \) is the Dirac’s Delta function (a single sample having unit value, preceded and followed by thousands of zeroes) – this way the total effect of the filter+system is simply a delay of a few millisecond, with no other evident alteration.
Possible designe strategies for an equalizing filter

• Mourjopoulos – Least-squares recursive method in time domain – the whole frequency range is always completely inverted.
• Neely & Allen – the filter is designed in the frequency domain – only the magnitude of the transfer function is inverted, so the equalized system will have flat frequency response, but it will still be “smeared” in time.
• Nelson & Kirkeby – again in the frequency domain, but the whole complex value is inverted, adding a small regularization quantity at the denominator for avoiding instabilities and ensuring a finite length of the inverse filter
Theory of Kirkeby inversion

- **Step 1** – pass to frequency domain through FFT

\[ H(\omega) = \text{FFT}[h(\tau)] \]

- **Step 2** – make the complex reciprocal at each frequency:

\[ F(\omega) = \frac{\text{Conj}[H(\omega)]}{\text{Conj}[H(\omega)] \cdot H(\omega) + \varepsilon(\omega)} \]

- **Step 3** – go back to time domain through an IFFT:

\[ f(\tau) = \text{IFFT}[F(\omega)] \]

Parametro di regolarizzazione
Regularization parameter $\varepsilon(\omega)$ variable with frequency

Changing the value of the regularization parameter allows for a very accurate filter at central frequency, and progressively a less aggressive filtering at very low or very high frequencies.
Example

- A loudspeaker+microphone system was measured:
Inverse filter example

- System’s impulse response
Inverse filter example

- Convolution of inverse filter with the system’s impulse response
The software

- Real-time control over the theatre (ABCDE) and the music piece (123456)
- Collection of the questionnaires with a graphical user’s interface
- The user is allowed to switch at will the sound samples and theatres, and to change the responses
Statistical Analysis of the results

**Aim:** possible correlation between objective parameters measured inside the theatres and subjective descriptors.

Objective parameters choosen:

- **Monophonic parameters:** reverberation times $T_{10}, T_{20}, T_{30}, EDT$; clarity $C_{50}$ e $C_{80}$; center time $T_s$; $D_{50}$.

- **Spatial parameters:** *Lateral Fraction*(Lf) and *Inter Aural Cross Correlation* (IACC)

- **Tonal parameters:** *Tonal Balance* (TB) and *Bass Ratio* (BR)
Method of analysis: *linear regression* (multiple regression to manage the whole matrix of subjective/objective data).

Defining $r$ as coefficient of correlation:

$$r = \frac{\sigma_{xy}}{\sigma_x \sigma_y}$$

$|r| > 0.3$

Objective parameter $\rightarrow r \rightarrow$ Subjective parameter
Matrix 9x11 of “r”: objective parameters on the abscissa and subjective parameters on the ordinate.

<table>
<thead>
<tr>
<th>Coeff. Di Regressione Lineare ORCHESTRAL</th>
<th>C50</th>
<th>C80</th>
<th>D50</th>
<th>Ts</th>
<th>EDT</th>
<th>T20</th>
<th>T30</th>
<th>LF</th>
<th>IACC</th>
<th>TB</th>
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<td>0,15</td>
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<td>-0,05</td>
<td>0,01</td>
<td>-0,12</td>
<td>-0,11</td>
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</table>
Results

Bad correlation between objective and subjective parameters. Example: pleasant vs. T30 gives a correlation $r=-0.20$
Results

IACC 0.42 diffuse-localisable

Tonal Balance 0.31 treble boosted-treble reduced
   -0.34 bass boosted-bass reduced

Lateral Fraction -0.51 round-sharp
Although the results were inherently bad, they were useful for improving the methodology, along these findings:

- Future listening tests
- Statistical analysis of the results

Only one music sample for test
Short explanation of the attributes

Advanced statistical methods: *factor analysis* or *multivariate regression.*
The Present
THE NEW LISTENING TEST WITH 4 DIFFERENT SOUND SYSTEMS

• 4 reproduction sound systems being compared:
  1. Headphones
  2. Traditional normal stereo
  3. Stereo Dipole
  4. Dual Stereo Dipole

• Monophonic sound source (accordion) – no multiple soundtracks

• The same 5 virtual rooms as in the preliminary experiment

• Different source-receiver distances in each room
Recording of the anechoic music piece

(ASK Industries, Reggio Emilia)
THE NEW LISTENING ROOM

(“Casa della Musica” – Parma)

To reduce the reverberation:

- **High Frequencies:**
  traditional absorbing panel (glass wool, pyramid, ecc... )

- **Low Frequencies:**
  resonant open cavities (cartoon boxes and tube traps), double side rigid and vertical absorbing panels.
Modern multiple-format microphonic systems

- Microphones Neumann KS-140 in configuration ORTF (angle of 110°)
- Dummy head Neumann KU-100 with binaural microphones
- Soundfield ST-250 microphone probe
- Turn table for recordings to different angles
Binaural / HEADPHONES

- **Sennheiser HD 580 Precision**

-In principle, they should put the right pressure exactly where it was recorded, at the ears, maintaining perfect separation and not being affected by the room response.

- Each signal is passed through an inverse FIR filter, which is computed after a measurement performed with the headphones over the dummy head.
THE NORMAL STEREO

- *Dynaudio* self-powered studio monitors

- The sound is picked up by the two cardioid microphones

- Each of the two signals is passed through a proper inverse filter, computed after a measurement performed placing the ORTF microphones at the listening position, in front of the loudspeakers
The cross-talk cancellation allows for the replica of the recorded signals at the ears of the listener.
THE DUAL STEREO DIPOLE

Reproduction over a Dual-Stereo-Dipole loudspeaker rig

It is a four-channel system, in which a frontal stereo dipole is employed for reproducing the sound coming from directions located in the frontal hemispace, and the rear stereo dipole reproduces the sound coming from the rear hemispace.
Reverberation Time of the listening room
Speakers positioning

Genelec S30D
Dynaudio
QSC AD-S82H
Design of cross-talk canceling filters

- First, a binaural measurement is made in front of the Stereo Dipole loudspeakers

- Then, the cross-talk cancelling filters are computed, so that their convolution with the measured impulse responses reduces to the identity matrix
Stereo Dipole inverse filters

\[
\begin{align*}
    f_{ll} &= (h_{rr}) \otimes \text{InvDen} \\
    f_{lr} &= (-h_{lr}) \otimes \text{InvDen} \\
    f_{rl} &= (-h_{rl}) \otimes \text{InvDen} \\
    f_{rr} &= (h_{ll}) \otimes \text{InvDen} \\
    \text{InvDen} &= \text{InvFilter} (h_{ll} \otimes h_{rr} - h_{lr} \otimes h_{rl})
\end{align*}
\]

\[
C(\omega) = \text{FFT}(h_{ll}) \cdot \text{FFT}(h_{rr}) - \text{FFT}(h_{lr}) \cdot \text{FFT}(h_{rl})
\]

\[
\text{InvDen}(\omega) = \frac{\text{Conj}[C(\omega)]}{\text{Conj}[C(\omega)] \cdot C(\omega) + \epsilon(\omega)}
\]
Hardware and Software setup

- Anechoic tracks preconvolved with IRs of the theatres at the beginning of the chain
- Filtering through Audiomulch in real time
- Voxengo Pristine Space VST plugin employed as multichannel convolver
The Voxengo Pristine Space multichannel convolver

- Filtering through Audiomulch in real time

- Voxengo Pristine Space VST plugin employed as multichannel convolver
Software for collecting the answers

- Listening and simultaneous evaluation of 10 sounds, differing for position inside the theatre and system of reproduction
GOALS of the new subjective experiment

- Evaluation of spatiality and distance in auditoria through auralized tracks and different reproduction systems

- Evaluation of the best system for a realistic reproduction

- The subjective tests are undergoing in these months, the results will be published in 2005
Conclusions (preliminary yet)

• The loudspeakers are generally judged more natural than the headphones

• The frontal (single) stereo dipole is the system providing the better result/effort ratio: it works with a normal stereo system, it does not require a lot of computational power for running the inverse filters in real time, and is generally very natural and provides good localization

• Adding the second stereo dipole gives some advantages, but generally they are not worth the extra effort required (4-channels processing, etc.)

• The normal stereo gives very poor localization, and the perceived spectrum changes with the direction of the sound source, so the “colour” of the sound is not preserved

• It is very important to provide a comfortable user’s interface: large LCD screen, wireless mouse, comfortable seat, proper lighting, thermohygrometric confort.
The Future
Future improvement of the listening room

• Multichannel (surround) sound systems, including:
  • ITU 5.1 horizontal surround (2D) through Dolby AC3 or DTS
  • Advanced 7.1 horizontal surround (2D) through DTS-ES or Microsoft WMA
  • 1°-order periphonic (3D) Ambisonics (8 loudspeakers)
  • Hybrid Stereo Dipole + Ambisonics = Ambiophonics (3D)
• A single computer running both the playback tool and the filtering tool
• Large LCD widescreen display (also good for films!)
• Fanless, completely silent liquid-cooled computer
• 16 playback channels through an Apogee DAC 16 converter and an RME Hammerfall soundcard with 2 ADAT optical outputs
ITU 5.1 surround

- **Williams MMA**
  
  **Schematic of the setup**
  C : Cardioid, $0^\circ$
  L, R : Cardioid, $\pm 40^\circ$
  LS, RS : Cardioid, $\pm 120^\circ$

- **INA-5**
  
  **Schematic of the setup**
  C : Cardioid, $0^\circ$
  L, R : Cardioid, $\pm 90^\circ$
  LS, RS : Cardioid, $\pm 150^\circ$
ITU 5.1 surround

Schematic of the setup
C: Cardioid, 0°
L, R: Super Cardioid, ± 90°
LS, RS: Cardioid, ± 180°

OCT
Ambisonics 3D 1\textsuperscript{st} order

Reproduction occurs over an array of 8-24 loudspeakers, through an Ambisonics decoder.
The Soundfield microphone

• This microphone is equipped with 4 subcardioid capsules, placed on the faces of a tetraedron.

• The signals are analogically processed in its own special control box, which derives 4 “B-format” signals.

• These signals are:
  • $W$ : omnidirectional
  • $X,Y,Z$ : the three figure-of-eight microphones aligned with the ISO cartesian reference system.
The B-format components

• Physically, $W$ is a signal proportional to the pressure, $XYZ$ are signals proportional to the three Cartesian components of the particle velocity

• when a sound wave impinges over the microphone from the “negative” direction of the $x$-axis, the signal on the $X$ output will have polarity reversed with respect to the $W$ signal
Ambisonics decoding

Each speaker feed is simply a weighted sum of the 4 B-format signals.

The weighting coefficients are computed by the cosines of the angles between the loudspeaker and the three Cartesian axes.

\[
F_i = \frac{1}{2} \left[ G_1 \cdot W + G_2 \cdot (X \cdot \cos(\alpha) + Y \cdot \cos(\beta) + Z \cdot \cos(\gamma)) \right]
\]
A software Ambisonics decoder

Audiomulch VST host

Gerzonic bPlayer

Gerzonic Emigrator
Rooms for Ambisonics 3D 1st order

University of Bologna

University of Ferrara

ASK – Reggio Emilia
Virtual high-order microphones
(M. Poletti)

One of the two ORTF cardioid is employed, which samples 36 positions along a 110 mm-radius circumference.

From these 36 impulse responses it is possible to derive the response of cylindrical harmonics microphones (2D Ambisonics) up to 5th order.
Ambiophonics 3D (10 loudspeakers):
Ambiophonics Room at ASK
Ralph Glasgal’s Amphiophonics Room at the Amphiophonics Institute
New hardware tools

- Apogee DAC 16
- RME Hammerfall soundcard
- Liquid-cooled FutureClient PC
- Toshiba 23” LCD TV/Monitor
Internet resources

All the papers previously published by Angelo Farina can be downloaded from his personal web site:

www.angelofarina.it

The software system employed for this research is based on the following modules:

Adobe Audition (www.adobe.com)
Aurora Plugins (www.aurora-plugins.com)
Audio Mulch (www.audiomulch.com)
Voxengo Pristine Space (www.voxengo.com)