Extended Source-Filter Model of Harmonic Instruments
for Sound Synthesis, Transformation and Interpolation

Henrik Hahn    Axel Röbel
henrik.hahn@ircam.fr

IRCAM - CNRS - UMR 9912 - STMS, Paris, France

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Sample Based Synthesis (Overview)

- an electronic instrument
- based on ‘playback’ of prerecorded instrument sounds
- playback is triggered by some input device (MIDI Keyboard)

- instrument characteristics are discretized
- synthesis sounds static
- no expressive control
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Sample Based Synthesis (State of the art)

- recordings on a semitone scale
- recordings at several intensities
- transformations based on local Source-Filter approach

- soundspace is does not contain knowledge about intermediate values.
- transformations do not account for real instrument characteristics.
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\[ I : \text{Intensity } \{a=1...A\} \]
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Sample Based Synthesis (Proposed Method)

- usage of **State of the Art** databases
- **parametric** model to describe the whole instrument sound characteristic along **pitch** / **global intensity**
- account for **temporal evolution** of a sound (**ASR**) denoted **local Intensity**
- separately treat **harmonic** and **noise** components
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- Sound synthesis with continuous pitch and intensity values
- Interpolation between sounds
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System Overview

- Harmonic/Noise Segregation
- Parameter Analysis
  - Global Intensity / Pitch
  - Local Intensity (ASR scheme)
- Model adaption for harmonic/noise component
- Remove estimated instrument sound from signal components
- Yields database of ‘flat’ residual sounds
- Interpolate 2 ‘flat’ residuals (harmonic / noise separately)
- Apply any parameter change to estimate new envelopes to use on ’flat’ residuals

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Signal Analysis

Harmonics/Noise Segregation
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- Partials are modeled as amplitude and frequency function per partial $k$ over time $n$:

$$A(k, n) \mid f(k, n)$$
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Harmonics/Noise Segregation

- Partials are modeled as amplitude and frequency function per partial $k$ over time $n$:

\[ A(k, n) \mid f(k, n) \]

- Noise is modeled as envelope using its smoothed Short Time Cepstrum $C(l, n)$

\[ C(l, n) \]
Parameter Analysis

Global Intensity / Pitch Analysis

- Obtained from meta data provided by the Database

Local Intensity

- Local intensity reflects amplitude envelope over time: $I_L(n)$.
- Threshold method to determine attack/release time frames $n_A, n_R$

Temporal Segmentation

- Segmentation using an overlapping scheme to define $n_s = \{n_a, n_r\}$
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Harmonic Model

Features by partial index $k$
- function for each $k$ depending on pitch $m$ (MIDI) and both intensities $I_G$ and $I_L$
- separate functions $s$ for attack-sustain and sustain-release
- may refer to a vibrating string / air pipe

Partial function $S^{k,s}(I_G, I_L, m)$

Features by frequency $f$
- invariant filter
- refers mainly to the instrument corpus

Resonance filter $R(f)$
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$S_{k,s}^{k}(I_G, I_L, m)$

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Resonance filter
$$R(f)$$
Harmonic Model

\[ \hat{A}^{k,s}(I_G, I_L, m, f(k, n)) = S^{k,s}(I_G, I_L, m) + R(f(k, n)) \]
Harmonic Model

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▶ Model of partial function using tensor-product B-splines:

\[ S^{k,s}(I_G, I_L, m) = \sum_{p,q,t} B_p(I_G)B_q(I_L)B_t(m) \cdot \gamma_{p,q,t}^{k,s} \]

B-Spline functions for \(B_p(I_G), B_q(I_L), B_t(m)\)

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Harmonic Model

\[ \hat{A}^{k,s}(l_G, I_L, m, f(k, n)) = S^{k,s}(l_G, I_L, m) + R(f(k, n)) \]

model of resonance filter using one-dimensional B-splines
Harmonic Model

\[ \hat{A}^{k,s}(l_G, l_L, m, f(k, n)) = S^{k,s}(l_G, l_L, m) + R(f(k, n)) \]

- model of resonance filter using one-dimensional B-splines

\[ R(f(k, n)) = \sum_{v} B_v(f(k, n)) \cdot \lambda_v \]

B-Spline functions for \( B_v(f(k, n)) \)
Noise Model

- Cepstral coefficients are described using a single tensor-product B-spline model:

\[
\hat{C}_{k,s}(I_G, I_L, m) = \sum_{p,q,t} P_{p,q,t} B_p(I_G)B_q(I_L)B_t(m) \cdot \delta_{p,q,t}
\]

B-Spline functions for \( B_p(I_G), B_q(I_L), B_t(m) \)
Parameter Estimation

Iterative method using Conjugate Gradient

\[ \mathcal{O}_h = \frac{1}{2} \sum_{s=1}^{2} \sum_{k,n_s}^K N_s |A(k, n_s) - \hat{A}^{k,s}(I_G, I_L(n_s), m, f(k, n))|^2 \]

\[ \mathcal{O}_n = \frac{1}{2} \sum_{s=1}^{2} \sum_{l,n_s}^L N_s |C(l, n_s) - \hat{C}^{k,s}(I_G, I_L(n_s), m)|^2 \]
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Model Results: Trumpet

$S^{k,s}$, $k = 1$

$S^{k,s}$, $k = 40$

$R(f)$:
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Subjective Evaluation

Tests have been made for trumpet and clarinet

- Interpolation between different pitches (12st and 24st)
- Interpolation between different intensities (pp-mf, mf-ff, pp-ff)

Sequence of 3 sounds has always been presented, framing the interpolated by their original sounds.

Each sequence was presented twice. Once containing the transformed and once the original counterpart.

Participants were asked to judge for any audible artifacts and convincingness.

Clarinet: \textit{mf}-ff  
Trumpet \textit{pp}-ff  
 Clarinet A\#3-A\#5
Subjective Evaluation

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Clarinet **A#3-A#5**
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Trumpet pp-ff

Clarinet A#3-A#5

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Subjective Evaluation: Results

- Measured the **Mean Opinion Score** for both instruments at once.
- **Org** represents original samples, **Mod1** and **Mod2** represent synthesized ones.

- MOS for original value way too low. Need for a new test with different setup.
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![Graphs showing MOS for different conditions](image)

**Pitch Interpolation:**
- 12st
- 24st

**IG Interpolation:**
- Mod1: *pp-mf* and *mf-ff*
- Mod2: *pp-ff*

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### Pitch Interpolation:
- **12st**
- **24st**

### $l_G$ Interpolation:
- **Mod1**: $pp-mf$ and $mf-ff$
- **Mod2**: $pp-ff$

<table>
<thead>
<tr>
<th>MOS</th>
<th>Org</th>
<th>Mod</th>
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<tbody>
<tr>
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<td>1.5</td>
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MOS for original value way too low. Need for a new test with different setup.
Subjective Evaluation: Results

- Measured the **Mean Opinion Score** for both instruments at once
- **Org** represents original samples, **Mod1** and **Mod2** represent synthesized ones.

Pitch Interpolation:
- 12st

Pitch Interpolation:
- 24st

\[ l_G \] Interpolation:
- Mod1: \textit{pp-mf} and \textit{mf-ff}
- Mod2: \textit{pp-ff}

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Introduction

Extended Source Filter Model

Model Results

Subjective Evaluation

Conclusions
We presented

- A parametric model for harmonic instruments
- A model which separately represents harmonic and noise components utilizing tensor-product B-splines
- An harmonic model separately representing features by partial index and frequency
- An objective function to estimate model parameters iteratively
- A subjective evaluation showing promising results

- More instruments need to be addressed (Strings, Piano, Guitar, ...)
- A subjective evaluation needs to be repeated with a different setup
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Thanks for listening