A Sound Source Separation System with the Ability of Automatic Tone Modeling

Kunio Kashino and Hidehiko Tanaka

Bldg.#13, Department of Electrical Engineering, Faculty of Engineering, The University of Tokyo, 7-3-1, Hongo, Bunkyo-Ku, Tokyo, 113, Japan.
kashino@MTL.t.u-tokyo.ac.jp tanaka@MTL.t.u-tokyo.ac.jp

Abstract

We describe the configuration, implementation and evaluation of a sound source separation system which has the ability of automatic tone modeling. Input is assumed to be a monaural audio signal containing the sounds of several kinds of musical instruments. The system separates the input by instrument and generates MIDI data with each MIDI channel assigned to one kind of instrument. The output data is also displayed on the screen in a score-like format. A prototype system has been implemented and evaluated by a benchmark test. It is shown that the performance of the proposed system is better than that of a conventional bottom-up based system.

1 Introduction

Sound source separation refers to extracting information which originates in one sound source under the condition that multiple sound sources are simultaneously present. This is a key technique to realize an automatic transcription system for ensemble music, a speech separation system, and an auditory scene analysis [Bregman, 1990a] system.

Here a physical sound source and a perceptual sound are distinguished, and the concept of perceptual sound source separation is proposed. A perceptual sound stands for the sound which humans hear as one sound, while a physical sound means an actual sound source itself. For example, when one hits a key on piano keyboard, usually a hammer strikes more than one piece of wire at once. In this case, there are multiple physical sound sources, though one usually hears the sound as one sound. Another example is to listen to an ensemble of instruments through one loudspeaker. In this case, there is a single physical sound source while we hear separate sound sources.

Over the past years, several approaches have been taken on sound source separation; most of them have aimed at separation of a physical sound. A broad classification of them would be (1) the approach using the information of sound localization [Mitchell et al., 1971] [Flanagan et al., 1985] [Nagata et al., 1991] and (2) the one based on template matching. Both approaches are inflexible: the former is sensitive to the location of sources or microphones, and cannot be applied to a monaural source; the latter approach requires the advance registration of templates.

This paper addresses a process model of perceptual sound source separation to realize flexible processing which has compatibility with human auditory characteristics. The flexibility has two aspects: the process model is (1) sound localization-free and (2) template registration-free. That is, the input of the system we describe is assumed to be a monaural sound signal, and the system runs without advance registration of the sound models.

Sound enhancement by means of harmonic selection [Parsons, 1976] [Nehorai et al., 1986] and some bottom-up approaches [Brown, 1992] in the literature can be thought as examples of perceptual sound source separation. However, treatment of the cues for perceptual sound separation was, so far, limited to qualitative one. Thus we pay attention to quantitative treatment of the cues.

If the complete template set of musical instruments were available, we not require a registration-free approach. However, one musical sound differs by instrument, and player, and varies significantly according to the performance conditions. Moreover, there are a number of "new" musical sounds generated electronically. Therefore we think that a template registration-free approach plays an important role in achieving flexibility of the processing.

In the following section, we introduce the perceptual sound source separation problem. After that, important modules in the proposed system will be discussed in detail, followed by the experimental results for evaluation of the system. Finally some remarks and future direction of the work will
2 Perceptual Sound Source Separation

In this section, perceptual sound source separation is formalized. Here, perceptual sound source separation is viewed from a standpoint of creating a system which simulates human perceptual system in performance, not in mechanisms.

Let $S(t)$ denote a sound signal, which can be represented by a set of parameters $P(t)$:

$$P(t) = \{p_1(t), p_2(t), \ldots, p_N(t)\},$$  \hspace{1cm} (1)

where $p_k(t)$ are parameters. Suppose the parameter set $P_{\text{F}}(t)$ represents monaural sound signal $S_{\text{F}}(t)$, where $S_{\text{F}}(t)$ is a mixture of sounds which arise from $M$ sound sources. Then sound source separation can be viewed as a problem of obtaining a parameter set $P_{\text{S}}(t)$ which represents the $i$–th sound source $S_{\text{S}}(t)$ from $P_{\text{F}}(t)$.

For example, assume that waveform of a sound signal is employed as the parameter. Then sound source separation is to determine the waveform of each sound source $s_i(t)$ from the mixture waveform $y(t)$, where

$$\sum_{i=1}^{M} s_i(t) = y(t).$$  \hspace{1cm} (2)

Mathematically, it is obvious that Equation (2) is underdetermined. Here, frequency components are used as parameters which represents a sound signal. That is, assume that $S(t)$ can be represented using a set of frequency components $F_j(t)$:

$$F(t) = \{F_1(t), F_2(t), \ldots, F_L(t)\},$$  \hspace{1cm} (3)

where $F_j(t)$ can be written as

$$F_j(t) = \{p_j(t), f_j(t), \varphi_j(t)\},$$  \hspace{1cm} (4)

where $p_j(t)$ and $f_j(t)$ stand for the power and frequency of the spectral peak; $\varphi_j(t)$ is the parameter which corresponds to bandwidth of the spectral peak. We should note that some sound sources are insufficiently represented by Equation (4), such as white noise, and are not considered here. A sound source separation can be formulated as follows:

**Problem 1:** Extraction of frequency components $F_j(t)$ on a sound spectrogram derived from a composite sound signal (Extraction of frequency components).

**Problem 2:** Clustering of $F_j(t)$ into a certain number of groups according to a certain criteria (Clustering of frequency components).

In problem 2, each cluster corresponds to an individual sound source. Frequently one extracted frequency component is actually a mixture of overlapping multiple frequency components which originate in different sources. Thus the clustering should be a duplex grouping, which refers to the clustering which allows a frequency component to be a member of multiple clusters at the same time. This corresponds to decomposition of overlapping harmonics.

Problem 2 can be devided into two sub problems. One is clustering to group frequency components which humans tend to hear as one sound. We call this the “sound formation clustering” problem, and the results of this clustering we call a “sound cluster”. Another sub problem is to group sound clusters which originate in the same sound source. We call this problem “source identification clustering”, and the results of this clustering “source cluster”.

In terms of perceptual sound source separation, the results of these clustering should be compatible with human auditory characteristics. Therefore it is important to investigate what humans use as clues in realizing auditory separation, not only for psychological validity but also from engineering point of view. In the field of psychological acoustics, it is claimed that the cues or clues which promote segregation or fusion of frequency components include the ones listed in Table 1 [Moore et al., 1986] [Moore, 1989] [McAdams, 1989] [Bregman, 1990a] [Bregman et al., 1990b] [Hartmann et al., 1990] [Darwin et al., 1992]. Among them, we have so far implemented four cues:

1. Harmonic mistuning of frequency components,
2. Onset asynchrony of frequency components,
3. Memory of timbres, and
4. Old-plus-new heuristic.

The automatic tone modeling has been realized in terms of the old-plus-new heuristic, as discussed in Section 7.

3 General Description of the System

Figure 1 gives a block diagram of the perceptual sound source separation system described in this paper.

Input of this system is assumed to be a monaural audio signal, which is a mixture of sounds of several kinds of musical instruments. Output of this system is MIDI data which includes several MIDI channels, each of which is assigned to one kind of instrument.
Table 1: Cues for Human Auditory Separation

<table>
<thead>
<tr>
<th>Simultaneous Spectral Features</th>
<th>Others</th>
</tr>
</thead>
<tbody>
<tr>
<td>(Seg.) Harmonic Mistuning of Freq. Comp.</td>
<td>(Seg./Fus.) Localization</td>
</tr>
<tr>
<td>(Seg.) Onset Asynchrony of Freq. Comp.</td>
<td>(Seg./Fus.) Sequential Grouping</td>
</tr>
<tr>
<td>(Seg.) Offset Asynchrony of Freq. Comp.</td>
<td>(Seg./Fus.) Memory of Timbres</td>
</tr>
<tr>
<td>(Fus.) Common FM of Freq. Comp.</td>
<td></td>
</tr>
<tr>
<td>(Fus.) Common AM of Freq. Comp.</td>
<td></td>
</tr>
<tr>
<td>(Seg./Fus.) Old-Plus-New Heristic</td>
<td></td>
</tr>
</tbody>
</table>

Note: Seg. : Promoting Segregation  
Fus. : Promoting Fusion  
Freq. Comp. : Frequency Components

Graphics is also generated in a simplified score-like format for visual monitoring of the results.

The following sections describe the important modules of the system.

4 Extraction of Frequency Components

The input audio signal is first digitized at 16bit/48KHz, and then frequency analysis is performed by a bank of bandpass filters. Each filter is a 2-order IIR (Infinite Impulse Response) type, and the center frequency of each filter is set linearly in a log frequency scale.

The next step is the extraction of frequency components (Problem 1). One of the simple methods may consist of peak-picking along the frequency axis with a threshold, and peak-tracking (concatenation) along time axis. In reality, since noise or adjacent peaks will interfere, it is difficult to achieve practical accuracy by a simple threshold method. Thus we have developed the pinching plane method in peak picking and tracking.

Figure 1: Configuration of the System

![Diagram of the system configuration](image)

Figure 2: Extraction of Frequency Components Using Pinching Planes

As illustrated in Figure 2, this method uses two planes pinching a spectral peak in order to decide the direction of concatenation of peaks. The planes are the regression planes of the peak, calculated by a least-squares fitting.

The pinching plane $z$ can be written as

$$ z = at + bf + c, \quad a, b, c : const., $$

(5)

where $t$ and $f$ are the time and frequency. Letting $z_{ij}$ denote the power at $(t_i, f_j)$ on the spectrogram, the normal vector $(a, b, -1)$ can be calculated by
the following equation:

\[
\begin{pmatrix}
\sum_{i=1}^{m} t_i^2 & \sum_{i=1}^{m} t_i f_j & \sum_{i=1}^{n} t_i \\
\sum_{i=1}^{m} t_i f_j & \sum_{j=1}^{n} f_j^2 & \sum_{j=1}^{m} f_j \\
\sum_{i=1}^{m} t_i & \sum_{j=1}^{n} f_j & mn
\end{pmatrix} \begin{pmatrix}
a \\
b \\
c
\end{pmatrix}
\]

\[
= \begin{pmatrix}
\sum_{i=1}^{m} t_i z_{ij} \\
\sum_{j=1}^{n} f_j z_{ij} \\
\sum_{i=1}^{m} z_{ij}
\end{pmatrix},
\]

where \( m \) and \( n \) are width and height of the planes.

Thus we get two normal vectors \( \vec{n}_1 \) and \( \vec{n}_2 \) of pinching planes \( \alpha \) and \( \beta \), respectively. The direction vector \( \vec{t} \) and angle \( \phi \) formed by \( \alpha \) and \( \beta \) are given by

\[
\vec{t} = \frac{\vec{n}_1 \times \vec{n}_2}{|\vec{n}_1 \times \vec{n}_2|}, \quad \text{and} \quad \cos \phi = \frac{\vec{n}_1 \cdot \vec{n}_2}{|\vec{n}_1| |\vec{n}_2|}.
\]

This \( \phi \) corresponds to \( \phi_2 \) in Equation (4).

In practice, two parameters have been introduced: \( \theta_p \), a threshold for peak detection, and \( \theta_e (\theta_p \leq \theta_e) \), a threshold for effective peaks. When a peak which is greater than \( \theta_e \) is found, the calculation described above starts, (forward as well as backward along the time axis), finding peaks and concatenating them, then stops when the peaks become weaker than \( \theta_p \). By this method, the accuracy of extraction of frequency components has been improved as compared with a simple threshold method.

5 Bottom-Up Processing

5.1 Clustering for Sound Formation

The next step is a bottom-up clustering of frequency components. As shown in Figure 3, the task is to group frequency components using their features, which is a part of Problem 2. The result of this clustering roughly corresponds to a sound human tends to hear as one.

To perform this clustering, we have employed the evaluation–integration model of bottom–up clustering, as illustrated in Figure 4. This model consists of an independent evaluation of features and their integration.

In the current implementation, (1) harmonic mistuning and (2) onset asynchrony between every two frequency components are evaluated in terms of probability of auditory separation. We performed psychoacoustic experiments, and as an approximation of their results, two functions for feature evaluation have been derived.*

The probability of auditory separation by harmonic mistuning \( c_h \) is evaluated by

\[
c_h(u) = \begin{cases} 
\frac{1}{p} & \text{for } u < p, \\
1 & \text{otherwise},
\end{cases}
\]

where \( u (\geq 0) [\%] \) is an amount of mistuning from harmonic relations. The value of parameter \( p \) is found to be 2.60 [%] by a least-square fitting of the results of psychoacoustic experiments.

The probability of auditory separation by onset asynchrony \( c_o \) is evaluated by

\[
c_o(t) = \begin{cases} 
\frac{40}{S_p} t & \text{for } t < (S_p/40), \\
1 & \text{otherwise}.
\end{cases}
\]

*The psychoacoustic experiments and derivation of approximate functions based on statistical analyses will be described in a separate paper.
where $t \geq 0$ [s] is an amount of onset asynchrony and $S_p$ [dB] is given by

$$S_p = a \frac{b}{f} + \frac{c}{g}$$

(11)

where $f$ and $g$ are the frequency in [Hz] and an average onset gradient in [dB/ms] of the earlier frequency component. Using multiple regression analysis, the values of parameters have been chosen to be:

$$\begin{align*}
  a &= 250, \\
  b &= 1.11, \\
  c &= 0.317.
\end{align*}$$

(12)

Integration of the evaluated values is performed by Equation (13):

$$m = 1 - (1 - c_b)(1 - c_k)$$

(13)

which is a reduced form of Dempster's law of combination, where $m$ is the integrated probability of auditory separation [Kashino et al., 1992].

In the clustering, this $m$ is used as a distance measure. At first, frequency components are roughly grouped by their onset time. In each group:

1. The component of lowest frequency is designated as the center of cluster $C_1$;

2. Scanning the components from low frequency to high, the component whose $m$ value is greater than $\theta_m$ is found, and designated as the center of another cluster $C_2$;

3. By scanning the rest of components, the component that all $m$ values (against existing centers of clusters) are greater than $\theta_m$ is found and designated as the center of cluster $C_k$;

4. Scanning continues until no more centers of clusters can be found;

5. Each remaining component is grouped in clusters if the $m$ values against their centers are less than $\theta_m$;

and we get sound clusters.

5.2 Clustering for Source Identification

In the case that a sound cluster consists of one note, it is possible to identify sound sources by global characteristics of the sound clusters. This corresponds to another part of Problem 2. In the current implementation, the hierarchical and agglomerative method of clustering has been employed. Distance $D_s$, which is used in the clustering, is defined as

$$D_s = c_1 f_p + c_2 f_q + c_3 t_a + c_4 t_s$$

(14)

where

- $f_p$ : Peak power ratio of the second harmonic to the fundamental component
- $f_q$ : Peak power ratio of the third harmonic to the fundamental component
- $t_a$ : Attack time
- $t_s$ : Sustain time

and each $c_k$ is a coefficient for each parameter.

6 Tone Model - Based Processing

As mentioned in the previous section, each sound cluster roughly corresponds to one sound. However, there are some cases where a sound cluster contains more than one note. For example, it is difficult to separate two simultaneous notes which are an octave apart in pitch solely by bottom-up processing. Tone model-based processing tries to deal with such complicated situations. This process requires stored tone models; and the automatic acquisition of models will be discussed in the next section.

One unit of the input of the tone model based processing is called a “processing scope”. A processing scope basically consists of one sound cluster, but in the case that multiple sound clusters share at least one frequency component as a member, these sound clusters are included in one processing scope. As tone models, we use such $T_k$ as

$$T_k = \{\tilde{a}_{ij}\},$$

$$\tilde{a}_{ij} = \left( \frac{p_{ij}}{p_m}, \frac{f_{ij}}{f_m} \right),$$

(15)

where $p_m$ is the maximum value of $p_{ij}$ in $T_k$, and $f_m$ is an average frequency of the fundamental frequency component of $T_k$. That is, tone model $T_k$ is a matrix of a two-dimensional vector which has normalized power and normalized frequency as elements. Each row of the matrix corresponds to a frequency component, and each column is proportional to time.

In the tone model based processing, “mixture hypotheses” are generated from $T_k$ and matched with a processing scope one by one, to find the mixture hypothesis with the closest fit. In hypothesis generation, the number of maximum simultaneous notes, time / phase shift, and amplitude adjustment are treated as follows:

- **Number of Maximum Simultaneous Notes**: Currently limited to be three.
- **Time Shift of Tone Models**: Each tone model is relatively shifted along time axis from $-20$ms to $20$ms by $10$ms step, and a mixture hypothesis itself is also shifted from
-20ms to 20ms by 10ms step against the processing scope.

- **Relative Phase of Overlapping Harmonics**
  In case of overlapping harmonics, the relative phase of each frequency component should be taken into account. Letting $p_1$, $p_2$, and $p_3$ be amplitude of frequency components $C_1$, $C_2$, and $C_3$, the amplitude of overlapping component $p_a$ is given by

  $$p_a = \left( (p_1 + p_2 \cos \varphi_{12} + p_3 \cos \varphi_{13})^2 + (p_2 \sin \varphi_{12} + p_3 \sin \varphi_{13})^2 \right)^{1/2} \quad (16)$$

  where $\varphi_{ij}$ is a relative phase between $C_i$ and $C_j$. In the current implementation, the phase of each component is relatively shifted from -90 degrees to 90 degrees by 15 degree steps.

- **Amplitude Adjustment of Tone Models**
  The amplitude of the tone model is adjusted so that the amplitude of the lowest frequency component in the tone model fits the amplitude of the corresponding component in the processing scope.

  In matching, distance $D_t$ is defined as

  $$D_t = \sum_{i=1}^{F} \sum_{j=1}^{T} |p_{ij}' - p_{ij}| \cdot f_{ij}^t \quad (17)$$

  where $F$ is the number of frequency components in the processing scope (two corresponding components between a hypothesis and a processing scope are counted as one), $T$ is the number of time samples in the hypothesis, $p_{ij}$ and $f_{ij}$ are the power and frequency of the component in the hypothesis, and $p_{ij}'$ is the power of the component in the processing scope.

  Through matching, the hypothesis which gives the minimum $D_t$ value is regarded as the result of sound source separation and identification. In terms of a perceptual sound source separation, this can be viewed as solving Problem 2 based on "memory of timbres" as mentioned in Section 2.

### 7 Automatic Tone Modeling

Tone model-based processing requires stored tone models, but is effective in recognizing chords. To realize a registration-free system with high accuracy, we discuss automatic acquisition of tone models.

The perceptual basis of automatic tone modeling is a "old-plus-new heuristic" [Bregman, 1990a], which refers to a hypothesis that a complex sound is interpreted as a combination of old (already encountered) sounds as much as possible, and the remainder of the interpretation is perceived as a new sound.

Figure 5: Pattern Matching Module with the Automatic Tone Modeling Mechanism

The process model of this mechanism is shown in Figure 5. As described in the previous section, the matching module calculates the distance $D_t$ in Equation (17) for each hypothesis. In this calculation, the correspondence of frequency components between the processing scope and a hypothesis should be considered. As shown in Figure 6, there are five cases:

**Case 1:** Frequency components correspond; the component in the hypothesis is a non-overlapped one
  (Hit, Non-overlap)

**Case 2:** Frequency components correspond; the component in the hypothesis is an overlapped one
  (Hit, Overlap),

**Case 3:** No corresponding components in the processing scope; the component in the hypothesis is a non-overlapped one
  (No hit, Non-overlap),

**Case 4:** No corresponding components in the processing scope; the component in the hypothesis is an overlapped one
  (No hit, Overlap), and

**Case 5:** No corresponding components in the hypothesis.

Considering these cases, we extend the distance $D_t$ in Equation (17) as

$$D'^2_t = \frac{1}{\sqrt{H(G - H_o)}} \cdot D_{ts} \quad (18)$$

where $G$ is the number of frequency components in the hypothesis, $H$ is the number of "hits", $H_o$ is the number of "hit and overlap" components, and

$$D_{ts} = \sum_{i=\theta}^{G} \sum_{j=1}^{T} |p_{ij}' - p_{ij}| \cdot f_{ij} \cdot z \quad (19)$$
where \( z \) is defined as 0 for the component of "Hit, Overlap", otherwise 1, and the value of \( p'_{ij} \) is regarded as 0 in Case 3 and Case 4.

In the matching module, the hypothesis which minimizes \( D'_{ij} \) is found; if the minimum \( D'_{ij} \) value is greater than a threshold, the remainder of the interpretation (that is, such components in the processing scope as Case 5) are grouped as a new model candidate.

Thus new tone model candidates are generated. The number of candidates is reduced in two ways: one is a "max set" reduction, meaning that if a candidate is judged as a subset of already existing tone models, that candidate will not become a new tone model. Another reduction is a kind of equivalence clustering, similar to the one described in section 5.2.

In the current implementation, the number of new model candidates generated in each processing scope is limited to one or less (no new model candidates or one). Another heuristic in use is that new model candidates are assumed not to share any overlapping harmonics with the hypothesis.

8 Evaluation of the Model

8.1 Benchmark Test

The process model has been evaluated by a benchmark test. The test sound signal was a random note pattern (125 chords) generated by a PCM sound module, which is controlled by a computer. The note range was limited from 65 to 79 in MIDI note number. An example of the random note pattern is shown in Figure 7.

The test has been performed in three ways: (1) only using bottom-up processing without advance registration of tone models, (2) using automatic tone modeling without advance registration of tone models, and (3) using advance registration of tone models. In the second case, tone models obtained by the bottom-up source identification module were removed manually except for one model, to evaluate the function of automatic tone modeling. The number of simultaneous notes was chosen to be 2 (in piano tone and flute tone) and 3 (2 in piano tone, and 1 in flute tone). Duration of each note was fixed to 720 ms.

Recognition rate \( R \) [%] is defined as

\[
R = 100 \cdot \frac{\text{right} - \text{wrong}}{\text{total}} \cdot \frac{1}{2} + \frac{1}{2}
\]  

(20)

where \( \text{right} \) is the number of correctly separated and identified notes, \( \text{wrong} \) is the number of spurious (surplus) notes in the output and incorrectly identified notes, and \( \text{total} \) is the number of notes in the input. Since it is sometimes difficult to distinguish surplus notes from incorrectly identified notes, both are included together in \( \text{wrong} \). Scale factor 1/2 is for normalizing \( R \): when the number of output notes is the same as the number of input notes, \( R \) becomes 0 [%] if all the notes are incorrectly identified and 100 [%] if all the notes are correctly identified by this normalization.

8.2 Result

The result of a benchmark test is shown in Figure 8. In the case of two simultaneous notes, the recognition rate for method two is significantly better than that of the bottom-up processing, and is comparable to that of the processing based on manual (advance) registration of tone models. This clearly shows that the automatic tone modeling works effectively. In the case of three simultaneous notes, recognition rate of the automatic tone modeling system deteriorates to the approximately same value as that of the bottom-up processing. The main reasons for this phenomenon can be considered as (1) errors in clustering for sound formation, (2) inaccuracy in distance measure given by Equation (18), and (3) an effect of overlapping harmonics: we expect the recognition rate will be improved by further study on these points.
9 Conclusion

We described a configuration, implementation and evaluation of a perceptual sound source separation system. In the system, four cues which humans use for sound separation have been implemented. Specifically, our implementation of the old-plus-new heuristic has permitted automatic tone modeling. It is shown by experimental results that the tone modeling is succeeded when the number of simultaneous notes is two, and consequently the recognition rate has been improved in comparison with the system based on only bottom-up processing. This paper is our first report of an automatic tone modeling approach, and formal benchmark tests and detailed analysis of their results will be reported later. Future work will include effective cooperation of bottom-up processing and model-based processing, a study on stability of automatic tone modeling, further experiments on treatment of overlapping harmonics, and adaptive control of the threshold to extract new tone model candidates.

Acknowledgement

The authors would like to thank Kazuhiro Nakadai for valuable suggestions and for his help in implementation of the system described in this paper.

References


