

# Exploiting Music Source Separation for Automatic Lyrics Transcription with Whisper

Jaza Syed<sup>\*†</sup>, Ivan Meresman Higgs\*, Ondřej Cífka<sup>†</sup> and Mark Sandler\*

\*School of Electronic Engineering and Computer Science, Queen Mary University of London

<sup>†</sup>AudioShake

Email: j.syed@qmul.ac.uk, i.meresman-higgs@qmul.ac.uk, ondrej@audioshake.ai, mark.sandler@qmul.ac.uk

**Abstract**—Automatic lyrics transcription (ALT) remains a challenging task in the field of music information retrieval, despite great advances in automatic speech recognition (ASR) brought about by transformer-based architectures in recent years. One of the major challenges in ALT is the high amplitude of interfering audio signals relative to conventional ASR due to musical accompaniment. Recent advances in music source separation have enabled automatic extraction of high-quality separated vocals, which could potentially improve ALT performance. However, the effect of source separation has not been systematically investigated in order to establish best practices for its use. This work examines the impact of source separation on ALT using Whisper, a state-of-the-art open source ASR model. We evaluate Whisper’s performance on original audio, separated vocals, and vocal stems across short-form and long-form transcription tasks. For short-form, we suggest a concatenation method that results in a consistent reduction in Word Error Rate (WER). For long-form, we propose an algorithm using source separation as a vocal activity detector to derive segment boundaries, which results in a consistent reduction in WER relative to Whisper’s native long-form algorithm. Our approach achieves state-of-the-art results for an open source system on the Jam-ALT long-form ALT benchmark, without any training or fine-tuning. We also publish MUSDB-ALT, the first dataset of long-form lyric transcripts following the Jam-ALT guidelines for which vocal stems are publicly available.

**Index Terms**—Automatic Lyrics Transcription, Voice Activity Detection, Source Separation

## I. INTRODUCTION

Lyrics transcription is the task of obtaining lyrics from the singing voice in a musical recording. Automatic lyrics transcription (ALT) systems enable lyrics from large musical datasets to be obtained without requiring manual transcription. Recently, the open source state-of-the-art (SOTA) automatic speech recognition (ASR) model Whisper [1] has been found to perform well for ALT [2], significantly outperforming domain-specific approaches [3] and earlier ASR models fine-tuned for ALT [4]. LyricWhiz [5] achieved SOTA performance by combining multiple rounds of inference with Whisper with a closed source large language model (LLM) as a text post-processor. Whisper outputs have been used for downstream music information retrieval tasks that require lyrics transcription as a processing step [6], motivating the investigation of Whisper’s characteristics as an ALT system.

<sup>†</sup>Corresponding author

This work was supported by UKRI - InnovateUK [Grant Number 10102804]

A key challenge in ALT is interference from musical accompaniment [7], as vocal stems are not generally available. Both commercial [2] and open source [4] ALT systems use music source separation (MSS) to obtain separated vocals as a preprocessing step to reduce interference. In [4], an ASR model is fine-tuned and evaluated on vocals separated using Hybrid Demucs [8]. However [2] finds that using separated vocals with Whisper for long-form transcription can cause significant degradation in transcription quality. Vocal stems with no musical accompaniment are expected to be easier to transcribe, however no long-form ALT dataset with corresponding vocal stems is publicly available. As vocal stems are the ultimate target of source separation, comparing performance with stems and separated vocals may reveal areas for improving source separation for ALT.

Although Whisper has been successfully used for ALT, the model makes systematic errors in ALT due to fundamental differences in the goals of ASR and ALT systems. One such difference is the presence of “backing vocals” in music, which are vocals secondary to the lead vocal line. It is common practice to transcribe these, as they may contribute to the lyrical content of the song, and to enclose them in parentheses to distinguish them from the lead vocals [2]. In contrast, in ASR it is often desirable for a model to ignore background speech – Whisper is evaluated on robustness to additive “pub noise” [1] which may contain speech. Song lyrics also contain non-lexical vocables [9] such as *ooh*, *ah*, and *la*. Recent work [10] has shown that such vocalizations are often discarded by ASR models such as Whisper.

In ALT datasets such as JamendoLyrics MultiLang [11] and DALI [12], non-lexical vocables are often inconsistently annotated, using variable numbers of vowels to emphasize syllable length or with inconsistent numbers of syllables of multi-syllable vocalizations. wh Jam-ALT [2] is the first dataset for long-form ALT with consistently transcribed backing vocals and non-lexical vocables. Jam-ALT also has accompanying annotation guidelines that align with industry standards for lyrics transcription and formatting. Lyrics annotations [13] have been published for the widely used MSS dataset MUSDB18 [14], however they are only provided for short segments which may overlap in time, so cannot be used for long-form transcription. They also do not follow the Jam-ALT guidelines for backing vocals and non-lexical vocables.

In addition, ALT systems often process extended recordings

exceeding their underlying model’s input length. While the ASR and ALT literature uses the terms “short-form” and “long-form” inconsistently, in Whisper’s context, these terms have specific meanings: the underlying neural network accepts audio segments under 30 s (“short-form”), requiring segmentation strategies for longer recordings (“long-form”). Whisper’s performance on long-form transcription tasks is sensitive to the segmentation algorithm used [1]. WhisperX [15] improves on the performance of Whisper’s native long-form segmentation algorithm for long-form ASR transcription tasks by using Voice Activity Detection (VAD) to determine segments, but has not been evaluated for long-form ALT.

In short-form ALT, the effect of MSS on Whisper’s output can be observed independently of potential interactions with the long-form transcription algorithm, which may affect the sections of audio transcribed. Since short-form ALT is a sub-task of long-form, findings should be transferable, however the two have not been systematically compared.

In this work, we study the effect of MSS on both short-form and long-form ALT. In order to determine the effect of MSS artifacts relative to “perfect” source separation, we construct a new dataset **MUSDB-ALT** of long-form lyrics transcripts of a subset of the MUSDB18 test set, following the Jam-ALT guidelines. We evaluate Whisper’s performance on Jam-ALT and MUSDB-ALT using the original audio, separated vocals for both datasets, and vocal stems for MUSDB-ALT. Our experimental code is published open source<sup>1</sup>. The contributions of our work are:

- We create and publish MUSDB-ALT<sup>2</sup>, the first consistently annotated dataset of lyrics for long-form ALT for which vocal stems are publicly available
- We publish line timings and annotations for non-lexical vocables in Jam-ALT<sup>3</sup> and MUSDB-ALT
- We show that short-form performance is strongly affected by sample length and propose a method of merging samples so that performance is similar to long-form
- We propose using separated vocals to obtain segment boundaries for long-form transcription, achieving SOTA performance on Jam-ALT for an open source system

## II. METHODS

In the following sections, we introduce our evaluation datasets, preprocessing methods for audio and lyrics, metrics, and specific experimental methods for short-form and long-form transcription.

### A. Data

Our work uses Jam-ALT [2], an updated version of the JamendoLyrics MultiLang [11] benchmark. Jam-ALT contains 79 songs in English, French, Spanish and German with full-song transcripts. To enable using Jam-ALT for short-form transcription, we enriched it with line-level timings based on JamendoLyrics. Specifically, we computed the optimal word

TABLE I  
SUMMARY OF DATASETS

Dataset	Songs	Duration (min)	Non-lexical%	Backing%
Jam-ALT	79	283	4.64	4.66
MUSDB-ALT	39	166	3.27	4.90

alignment (edit-distance-based, as explained below in Section II-B) between the JamendoLyrics and Jam-ALT transcripts, and used this alignment, along with the word-level timings in JamendoLyrics, to automatically derive line timings for Jam-ALT. We subsequently manually revised the timings to account for ambiguity in the automatic assignment (often caused by missing words or lines in JamendoLyrics), as well as inaccurate timing annotations in JamendoLyrics.

As another evaluation set, we propose MUSDB-ALT, a novel dataset of long-form transcripts with line-level timings following the Jam-ALT guidelines. We produced MUSDB-ALT manually based on the MUSDB lyrics extension [13]. The major changes we made were ensuring that non-lexical vocables were consistently transcribed, backing vocals were enclosed in parentheses and that line and section breaks in the transcripts corresponded to musically significant sections.

MUSDB-ALT consists of 39 of the 45 English-language songs in the MUSDB18 test set. Of the six songs excluded from the dataset, three contained primarily instrumental music with minimal lyrical content derived from highly processed vocal samples, two featured extended sections with three or more interacting vocal lines that could not be categorized as lead and backing vocals, and one was omitted due to screamed vocals with unintelligible lyrical content. We use audio from MUSDB18-HQ [14], the uncompressed version of MUSDB18.

In both datasets backing vocals are identifiable as they are within parentheses. We manually produced extra annotations of which words are non-lexical vocables. We use these annotations to define metrics specific to these words in Section II-B2. The dataset duration, number of songs and percentage of words in the data that are non-lexical vocables or backing vocals are shown in Table I.

### B. Metrics

The standard metric to evaluate ASR performance is the Word Error Rate (WER), which is calculated as the normalized minimum edit distance (Levenshtein distance) between what an ALT system produces (the hypothesis) and the ground-truth transcript (the reference). WER considers the minimum number of word-level edit operations (deletions, insertions, and substitutions) required to transform the reference into the hypothesis. Denoting the numbers of deletions as  $D$ , substitutions as  $S$ , insertions as  $I$ , and the number of correctly recognized words (hits) as  $H$ , the WER is defined as:

$$WER = \frac{S + D + I}{S + D + H} \quad (1)$$

Defining the total length of the reference  $N = S + D + H$  for brevity, we can disaggregate the WER into  $SR = S/N$ ,

<sup>1</sup><https://github.com/jaza-syed/mss-alt>

<sup>2</sup><https://huggingface.co/datasets/jazasyed/musdb-alt>

<sup>3</sup><https://huggingface.co/datasets/audioshake/jam-alt>

I went to the +++ park yesterday evening
I came to the new pool yesterday -----

Fig. 1. Example of an alignment with reference above and hypothesis below. Substitutions are in blue, hits are in black, insertions are denoted by + in green and are deletions denoted by – in red.

$DR = D/N$  and  $IR = I/N$  for substitutions, deletions and insertions respectively.

When computing the WER, we apply lyrics-specific punctuation normalization and tokenization from the `alt-eval`<sup>4</sup> package to both the reference and hypothesis, and remove all non-word tokens, following [2]. We compute the WER for multiple songs or short-form samples by summing the component edit counts  $H$ ,  $D$ ,  $S$ , and  $I$  over all the songs or samples. The minimal sequence of edits produced by the edit distance calculation can be interpreted as an *alignment*, where each hit or substitution aligns a hypothesis word to a reference word; insertions and deletions correspond to unpaired words in the hypothesis and the reference respectively. An example of an alignment is shown in Figure 1. Using the alignment, we define additional rates presented in the sections below.

1) *Hallucination rate*: Whisper can produce hallucinations - text totally unrelated to the audio input or repetition loops of the same short text segment [16]. As they are unrelated to the audio, these hallucinations frequently appear in the alignment as sequential insertions. We define  $I_{10}$  as the number of insertions within blocks of 10 or more sequential insertions, and define a proxy hallucination rate  $IR_{10} = I_{10}/N$ . Manual inspection confirmed these blocks are hallucinations with a low false positive rate, though the method fails with short hallucinations or when chance matches disrupt insertions.

2) *Deletions*: The most common cause of error for non-lexical vocables and backing vocals is deletions. Counting the number of deletions where the reference words are non-lexical vocables  $D_{NL}$  and backing vocals  $D_{BV}$ , we define  $DR_{NL} = D_{NL}/N$  and  $DR_{BV} = D_{BV}/N$  to disaggregate the deletion rate by reference word type.

### C. Source separation

We use two Hybrid Demucs [8] pre-trained models: `mdx` (trained on the MUSDB18 train set only) and `mdx_extra` (trained on the MUSDB18 train and test sets plus 800 additional songs). These models achieved Signal-to-Distortion Ratios (SDR) for vocals of 7.97 dB and 8.76 dB respectively on the MUSDB18 test set, demonstrating that `mdx_extra` improves the quality of separated vocals over `mdx`. When using `mdx_extra` for source separation with MUSDB-ALT, we are evaluating on the training set, meaning we likely overestimate performance on unseen data.

### D. Whisper

We use the *Faster Whisper*<sup>5</sup> implementation of Whisper. *Faster Whisper* features batched transcription over segments

TABLE II  
SUMMARY STATISTICS OF MERGED AND GROUPED LINES

Dataset	Type	Duration (s)				Count
		Mean	Std. Dev.	Min.	Max.	
Jam-ALT	Merged Line	3.52	1.80	0.44	17.66	3445
	Group	20.44	5.51	3.64	29.93	613
MUSDB-ALT	Merged Line	4.59	2.39	1.22	23.82	1488
	Group	19.84	5.51	2.27	29.93	359

of the same audio file, which we use for speed where possible. We use Whisper model `large-v2` and provide the language of the song at transcription time, following [2]. For decoding, we use a beam size of 5. We compute average error rates over 5 runs to account for Whisper’s stochastic decoding algorithm.

### E. Short-form transcription

Some lyric lines may overlap in time, so lines cannot directly be used as samples for short-form transcription. We use two methods to merge lines to produce non-overlapping samples of different average lengths, in order to investigate the effect of segment length on ALT performance.

1) *Merging*: Using a threshold of 0.2 s overlap between lines, we identify sets of transitively overlapping lines. To define a new short-form sample for each set, we concatenate the lines to obtain a transcript and use the earliest line start time and latest line end time as the timings. We exclude any samples longer than 30 s, of which there are none in Jam-ALT and only one in MUSDB-ALT. We refer to these samples as “merged lines”. Since this process excludes so few lyrics, metrics for short-form and long-form transcription are comparable.

2) *Groups*: To produce a second set of short-form samples of greater average duration, we apply an additional two-step grouping procedure to the merged lines for each song:

- We split the lines into groups on gaps of longer than 7 s between the end and start of consecutive lines.
- We split the groups into one or more subgroups of consecutive lines such that the minimum subgroup duration is maximized and the maximum is under 30 s.

Each group of merged lines produces a short-form sample, where the earliest start time and latest end time are the timings of the new sample and the concatenated transcripts are the new transcript. The durations of the merged lines and groups are summarized in Table II.

### F. Long-form transcription

In the following section, we present a novel segmentation algorithm for long-form ALT and describe considerations specific to our use of Whisper’s native long-form algorithm.

1) *RMS-VAD*: Applying a threshold to the root mean square (RMS) amplitude of separated vocals is an effective VAD for singing voice [17]. For a signal  $x[n]$  where  $n$  is the sample index, we compute the time-varying RMS amplitude with frame size  $N$  as

$$\text{RMS}[n] = \sqrt{\frac{1}{N} \sum_{m=n-N+1}^n x[m]^2}. \quad (2)$$

<sup>4</sup><https://github.com/audioshake/alt-eval>

<sup>5</sup><https://github.com/SYSTRAN/faster-whisper>

TABLE III  
SHORT-FORM RESULTS

Type	Audio	Jam-ALT							MUSDB-ALT						
		WER	SR	DR	IR	IR <sub>10</sub>	DR <sub>NL</sub>	DR <sub>BV</sub>	WER	SR	DR	IR	IR <sub>10</sub>	DR <sub>NL</sub>	DR <sub>BV</sub>
Group	Original Mix	<b>20.99</b>	9.69	9.54	<b>1.76</b>	0.05	3.54	3.07	23.59	9.24	12.31	2.03	<b>0.12</b>	1.77	3.31
	Separated (mdx)	21.17	9.25	10.14	1.78	<b>0.00</b>	3.40	3.12	23.98	9.56	10.47	3.96	1.80	1.68	3.03
	Separated (mdx_extra)	21.08	<b>9.02</b>	<b>9.04</b>	3.01	1.29	<b>3.24</b>	<b>3.04</b>	<b>20.00</b>	<b>8.16</b>	<b>9.81</b>	<b>2.03</b>	<b>0.12</b>	<b>1.64</b>	<b>2.77</b>
	Vocal Stem	-	-	-	-	-	-	-	14.19	4.94	7.75	1.51	0.21	1.51	2.90
Merged Line	Original Mix	29.46	17.86	<b>7.67</b>	3.93	<b>0.00</b>	1.95	2.89	28.57	15.17	10.22	<b>3.18</b>	<b>0.00</b>	<b>1.40</b>	2.96
	Separated (mdx)	29.88	17.69	8.13	4.07	0.46	<b>1.90</b>	<b>2.84</b>	29.89	14.65	9.40	5.84	2.63	<b>1.40</b>	2.76
	Separated (mdx_extra)	<b>28.06</b>	<b>16.53</b>	7.88	<b>3.66</b>	<b>0.00</b>	1.96	2.88	<b>25.40</b>	<b>13.30</b>	<b>8.10</b>	4.00	1.39	1.44	<b>2.60</b>
	Vocal Stem	-	-	-	-	-	-	-	14.93	6.88	6.27	1.78	0.00	1.22	2.63

We then define the normalised score  $VAD[n]$  as

$$VAD[n] = \frac{RMS[n]}{\max_m RMS[m]}. \quad (3)$$

The “Cut & Merge” segmentation algorithm defined in [15] converts  $VAD$  scores into non-overlapping vocal activity segments optimized to approach a specified maximum length. Whisper’s output for each segment can then be concatenated to obtain the final transcript. We extend the approach in [17] to define a new segmentation algorithm **RMS-VAD**:

- 1) Obtain separated vocals with a source separation model
- 2) Compute  $VAD[n]$  for the separated vocals
- 3) Obtain boundaries from  $VAD[n]$  using “Cut & Merge”

For “Cut & Merge”, we use onset and offset thresholds of 0.1, a minimum silence duration of 1s and a maximum segment length of 30s, matching Whisper’s input duration.

2) *Whisper*: Whisper groups its text output into sections with predicted start and end timestamps. Its native long-form transcription takes the last predicted timestamp in a 30s window as the start of the next window. This approach prevents batched transcription of sections of a single song unlike when segment timings are provided by algorithms such as RMS-VAD. In long-form transcription, it is possible to use Whisper’s prompt

to condition the output for a segment on the text predicted for the previous segment. Since this conditioning is incompatible with batched transcription, we disabled it in the native long-form algorithm. This change had no significant impact on performance.

### III. EXPERIMENTS

In the following sections, we present the results of our experiments on both short-form and long-form transcription tasks. In both experiments, we report the metrics described in Section II-B as percentages.

#### A. Short-form transcription

We evaluate Whisper’s short-form performance, using both the merged lines and groups described in Section II-A as samples. For Jam-ALT and MUSDB-ALT, we evaluate with the original audio, vocals separated with `mdx` and vocals separated with `mdx_extra`. We also evaluate on vocal stems for MUSDB-ALT. We show the results in Table III with the lowest error rates in bold, except for vocal stems.

1) *Sample length*: We observe that evaluating on groups yields lower WER than evaluating on merged lines in almost all cases. This improvement stems from reduced substitutions, highlighting the importance of extended audio context to increase the intelligibility of the vocals to Whisper. Deletions are mostly lower for merged lines than for groups, except for backing vocals. Interestingly, the lower deletions are mostly accounted for by a reduction in  $DR_{NL}$  for Jam-ALT. Group and line evaluations show similar performance only for MUSDB-ALT stems, suggesting that musical accompaniment or source separation artifacts are what cause sample durations to affect performance significantly.

Error rates for groups are similar to those for long-form transcription in Table IV, whereas line-level performance is significantly worse. This is likely because the segment lengths in long-form transcription are close to 30s. As longer sample length is crucial to Whisper’s short-form ALT performance, we only consider group-level evaluation in the remaining analysis.

2) *Source separation*: For Jam-ALT the WER remains essentially unchanged for both original and separated audio. However, the WER for `mdx_extra` is skewed higher due to the presence of a significant amount of hallucinations.  $SR$  is improved by MSS and is better for `mdx_extra` than `mdx`,

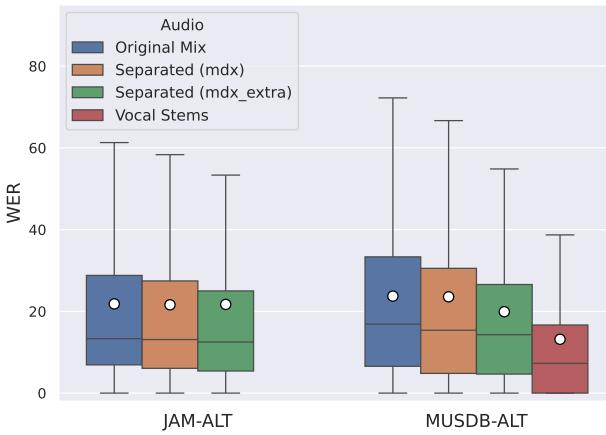


Fig. 2. Group-level WER distribution by dataset and audio. Means are shown by white circles. Outliers which affect the means occur but are not shown. WER medians and quartiles improve slightly with separation quality for JAM-ALT and more significantly for MUSDB-ALT.

TABLE IV  
LONG-FORM RESULTS

Audio	Algorithm	Jam-ALT							MUSDB-ALT						
		WER	SR	DR	IR	IR <sub>10</sub>	DR <sub>NL</sub>	DR <sub>BV</sub>	WER	SR	DR	IR	IR <sub>10</sub>	DR <sub>NL</sub>	DR <sub>BV</sub>
Original Mix	Native	23.02	10.84	9.16	3.03	1.31	3.29	<b>2.69</b>	25.82	10.39	12.25	3.19	1.09	1.97	3.31
	RMS-VAD	<b>20.35</b>	9.94	<b>8.74</b>	1.67	<b>0.35</b>	3.25	3.05	22.72	9.24	11.01	2.47	<b>0.12</b>	1.89	3.30
Separated Vocals	Native	22.87	9.64	10.06	3.18	1.35	<b>3.21</b>	2.88	21.90	9.07	<b>9.71</b>	3.12	0.80	<b>1.47</b>	<b>2.92</b>
	RMS-VAD	20.72	<b>8.65</b>	10.48	<b>1.58</b>	0.41	3.32	3.04	<b>20.07</b>	<b>7.87</b>	10.14	<b>2.06</b>	0.24	1.93	3.08
Vocal Stem	Native	-	-	-	-	-	-	-	17.51	6.31	8.26	2.93	0.95	1.54	2.87
	RMS-VAD	-	-	-	-	-	-	-	14.98	4.85	8.53	1.60	0.12	1.89	3.09

demonstrating that increasing MSS quality improves vocal intelligibility for Jam-ALT. Subtracting  $IR_{10}$ , the same trend applies to the WER. Source separation has little impact on reducing deletions for Jam-ALT and shows negligible effects on insertions, especially when hallucinations are excluded.

For MUSDB-ALT, the overall WER and components  $SR$ ,  $DR$  and  $IR$  consistently decrease as separation quality improves. Notably, the `mdx_extra` model, which was trained on MUSDB-ALT songs, shows a substantial improvement. Deletions are significantly lower for stems than for separated vocals, indicating that MSS artifacts cause Whisper to fail to transcribe some vocals.

Figure 2 shows that the mean and quartiles of the WER distribution follow the same trend for both datasets, with reduced effect for Jam-ALT. Improved separation quality reduces substitutions across both datasets, indicating that source separation increases vocal intelligibility for Whisper.

3) *Hallucinations*: A low proportion of the WER is accounted for by  $IR_{10}$ , particularly for Jam-ALT.  $IR_{10}$  is consistently low for the original audio and vocal stems but is higher for some cases with MSS, indicating that source separation artifacts can trigger hallucinations.

4) *Deletions*: Non-lexical vocables and backing vocals constitute a high proportion of total deletions. Comparing their occurrence rates in Table I with  $DR_{NL}$  and  $DR_{BV}$  in Table III shows that over half of both are deleted in both datasets, revealing a shortcoming in Whisper. For Jam-ALT, MSS does not impact these rates. While improved separation quality

reduces both rates for MUSDB-ALT, the proportionate decrease is significantly less than the proportionate decrease in  $DR$ . The high values observed even with vocal stems indicate that this issue cannot be resolved by improved source separation.

### B. Long-form transcription

For long-form transcription, we use only `mdx_extra` for source separation because it demonstrated superior performance for short-form transcription. We evaluate on original mixes, separated vocals and vocal stems using two segmentation approaches: (1) Whisper’s native long-form transcription algorithm, which determines boundaries by predicting timestamps in the audio being transcribed and (2) our approach RMS-VAD, which determines boundaries using the amplitude of separated vocals. When evaluating with RMS-VAD, we use the boundaries obtained from the separated vocals even when evaluating on original mixes and vocal stems. This design allows us to isolate the effect of RMS-VAD boundaries from the effect of the audio type. We show the results in Table IV with the lowest error rates in bold, except for vocal stems.

1) *Effect of RMS-VAD*: For Jam-ALT, source separation provides little improvement when used with the native algorithm, which is consistent with the short-form Jam-ALT results that indicated little improvement. Using RMS-VAD improves the WER over Whisper’s native algorithm for both the original mix and the separated vocals. RMS-VAD shows similar performance with original and separated audio, indicating that the improvement comes from better segment boundaries rather than improved vocal quality after separation.

For MUSDB-ALT, RMS-VAD demonstrates consistent improvements over the native algorithm on original mixes, separated vocals and even stems, which highlights the consistency of the segmentation approach in enhancing transcription accuracy. Figure 3 shows that the quartiles and medians of the song-level WER distribution are also improved by RMS-VAD.

2) *Hallucinations and deletions*: The use of RMS-VAD has no significant effect on  $DR_{NL}$  or  $DR_{BV}$  however it does reduce  $IR_{10}$  relative to the native algorithm for all audio types. With RMS-VAD,  $IR_{10}$  values are similar to those for short-form transcription with the same audio, with the exception of `mdx_extra` on JAM-ALT. Whisper’s native algorithm uses its predicted timestamps to avoid cutting speech with segment boundaries [1], but these timestamps can be inaccurate [18]. In contrast, short-form transcription uses gold-standard human-annotated line boundaries and RMS-VAD is unlikely to cut

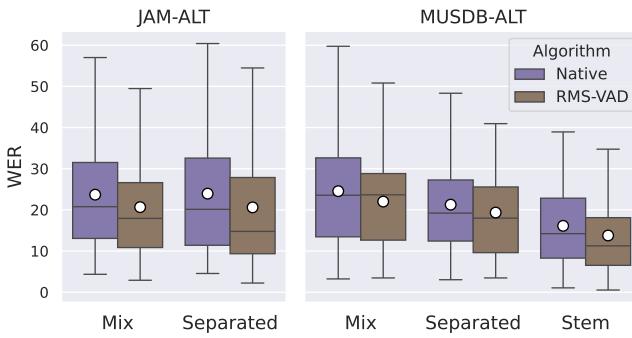


Fig. 3. Song-level WER distribution for each dataset, by audio and long-form transcription algorithm used. Means are shown by white circles. Outliers which affect the means occur but are not shown. Medians and quartiles of the WER are improved by RMS-VAD for all audio types and datasets.

TABLE V  
LONG-FORM WER FOR JAM-ALT BY LANGUAGE

Audio	Algorithm	Language			
		DE	EN	ES	FR
Original Mix	Native	18.14	26.92	17.07	24.98
	RMS-VAD	16.12	<b>24.73</b>	<b>14.28</b>	<b>24.41</b>
Separated Vocals	Native	15.75	25.41	19.80	25.68
	RMS-VAD	<b>13.03</b>	<b>24.73</b>	17.86	24.50
LyricWhiz		-	23.70	-	-

words as it uses a low vocal amplitude threshold. Therefore, we hypothesize that segment boundaries interrupting vocal activity can cause hallucinations.

3) *Comparison with SOTA*: We report the long-form WER by song language for Jam-ALT in Table V. We include the WER for LyricWhiz, which is only evaluated on the original mixes of the English songs, for comparison. The previous open source SOTA for Jam-ALT was Whisper using its native algorithm with provided language [2], which corresponds to the top row of both and Table IV and V. RMS-VAD boundaries improve the WER relative to the native algorithm for all languages on both original and separated audio. RMS-VAD evaluated on original mixes achieves a new open source SOTA WER of 20.35. For English, RMS-VAD provides an improvement on Whisper close to the effect of LyricWhiz without the use of multiple inference rounds or an LLM.

#### IV. CONCLUSION

In this work, we show that Whisper’s performance on line-level short-form transcription (under 30s) is not indicative of performance on long-form transcription (over 30s) for ALT due to the short duration of lines.

We introduce a method for grouping lines into longer short-form samples, which improves the utility of short-form evaluation for assessing the impact of audio preprocessing methods such as source separation on downstream performance. For long-form transcription, we introduce a new method RMS-VAD to obtain segment boundaries using source separation, which yields consistent improvements over Whisper’s native long-form algorithm. In practice, an ALT system may be run over large datasets of hundreds of thousands of songs. As source separation is computationally expensive, developing less resource-intensive music-specific VAD systems to obtain segment boundaries directly may therefore be a valuable direction for future work. In addition, future work on ALT systems would benefit from considering the importance of sample length and segment boundary placement demonstrated in this study.

We also show that Whisper systematically deletes non-lexical vocables and backing vocals, regardless of source separation quality. These deletions constitute a high proportion of the total, indicating that this is an area for significant potential improvement.

#### ACKNOWLEDGMENTS

We thank Tom Andersson and Yasmin Gapper for feedback on the manuscript.

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