POSITIVE TRANSFER OF THE WHISPER SPEECH TRANSFORMER TO HUMAN AND ANIMAL VOICE ACTIVITY DETECTION

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ABSTRACT

This paper introduces WhisperSeg, utilizing the Whisper Transformer pre-trained for Automatic Speech Recognition (ASR) for human and animal Voice Activity Detection (VAD). Contrary to traditional methods that detect human voice or animal vocalizations from a short audio frame and rely on careful threshold selection, WhisperSeg processes entire spectrograms of long audio and generates plain text representations of onset, offset, and type of voice activity. Processing a longer audio context with a larger network greatly improves detection accuracy from few labeled examples. We further demonstrate a positive transfer of detection performance to new animal species, making our approach viable in the data-scarce multi-species setting.[1]

***Index Terms*—** Voice activity detection, audio segmentation, Transformer, Whisper

1. INTRODUCTION

Voice Activity Detection (VAD) is the task of identifying the onset, offset, and type of speech in extended audio recordings, foundational in speech recognition [1], speaker diarization [2], and speech analysis tasks that require precise speech segmentations at word or utterance levels [3, 4, 5]. This task extends to the domains of animal communication [6] and vocal learning research [7], where vocalizations must be segmented against background noises [8, 9]. Given the intrinsic similarities between human and animal VAD, there is potential to develop a universal vocal segmentation tool that can effectively handle a wide range of species. Such a tool would streamline segmentation efforts and potentially contribute to comparative research on vocal communication.

Recent approaches to human and animal VAD typically use frame-by-frame classification strategies [10, 11, 12], where an audio spectrogram is divided into fixed-length frames (e.g., 20 to 30 columns within the spectrogram). A neural network such as a Convolutional Neural Network (CNN) then processes each frame to produce a hidden state. This state is subsequently processed by a recurrent neural network, like an LSTM, generating a final vector representation for each frame that is used for frame-wise classification. Depending on a selected threshold (e.g., 0.5), each frame is classified as non-voice (0) or voice (1), with additional classes to resolve vocalization type if necessary. Analyzing the resulting sequence of classifications allows the extraction of vocal activity details such as onset, offset, and type.

Traditional frame-wise classification approaches come with inherent challenges: 1) The classification of each frame requires an empirically chosen decision threshold. This threshold, applied universally to all frames, can falter in real-world recordings where nonstationary environmental noises and varying recording quality introduce substantial audio heterogeneity. 2) Audio frames may lack sufficient contextual information for accurately detecting vocal activity. Particularly with human speech, discerning between a pause within a structured vocal unit (e.g. an utterance) and true silence can be difficult without a broader context.

The recent Whisper Transformer is a speech-to-text encoder-decoder model trained on massive data [13]. In principle, Whisper overcomes the above challenges due to its threshold-free textual output and long input spectrograms with rich contexts. While prior studies have leveraged Whisper for human speech timestamp prediction [13, 14], its broader application remains relatively uncharted. This work investigates the transfer of Whisper to universal voice detection, targeting both human and animal vocalizations, elevating its potential as an all-encompassing VAD solution.

2. METHOD

Our methodology comprises the following steps:

Preprocessing and Adaptation: From an audio clip with sampling rate f_s and duration D, we compute the log-Mel spectrogram using 80 Mel bands within a species-specific frequency range (Table 1) and hop length L_{hop} between FFT windows, to yield a spectrogram with exactly $\frac{D \times f_s}{L_{\text{hop}}} = 1000$

[1]Our code and data are at https://github.com/nianlonggu/WhisperSeg

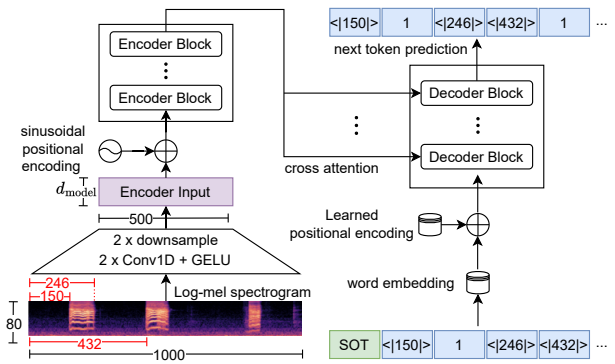
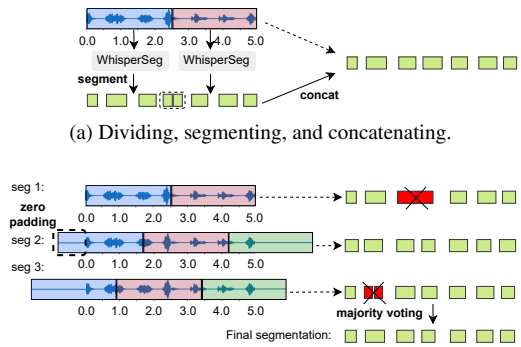


Fig. 1: WhisperSeg is trained to generate transcripts of vocal segments with onsets and offsets specified as column indices.



(a) Dividing, segmenting, and concatenating.

(b) Multiple segmentation variants and majority voting.

Fig. 2: Strategies for segmenting extended audios.

columns. D and L_{hop} are tailored for each species to account for varying utterance durations, such that one spectrogram contains a similar number of vocal segments across species (e.g., up to 30 segments per audio clip) and each segment spans a consistent number of spectrogram columns (e.g., 20 to 300 columns). By reducing inter-species differences in the input spectrogram, we can fine-tune a universal WhisperSeg model to detect onsets and offsets using spectrogram column indices rather than raw timestamps. This strategy obviates the need to add or adapt timestamping tokens for individual species, promoting a uniform segmentation procedure irrespective of the utterance duration and vocalizing species.

During conversion of onset and offset timestamps t into column indices for fine-tuning, we round $\frac{t}{L_{\text{hop}}/f_s}$ to the nearest even number. This aligns with Whisper's native time resolution, which is half that of the spectrogram due to the 2x downsampling in Whisper's convolution layers (Figure 1), which effectively halves the input size to Whisper encoder [13].

Model Fine-tuning: Utilizing the spectrograms and column indices obtained, we finetune Whisper to generate text sequences representing vocal activity. In the example in Figure 1, the target output is: `SOT<|150|>1<|246|><|432|>1<|518|><|810|>1<|854|>`, where `SOT` marks the start of decoding. The subsequent `<|150|>1<|246|>` indicates the onset of a vocal segment type (ID) 1 spanning from the 150th to the 246th spectrogram column.

Segment Extraction and Temporal Mapping: Segments are extracted from generated sequences using regular expressions. The onset and offset column indices are then mapped to timestamps relative to the beginning of the audio clip by multiplying the column index with the spectrogram's temporal resolution L_{hop}/f_s. The extracted segment type IDs are converted to the original segment type using a mapping table.

Segmentation of Extended Audios: To process extended sound files, we divide them into fixed-length clips (of duration D outlined in Table 1) and apply WhisperSeg to each. Consecutive segments from adjacent clips of the same type are concatenated (Figure 2a). However, the dividing point of sound clips may affect accuracy, especially when it inadvertently bisects a vocalization. To counter this effect, we process a sound file 3 times, each time with a distinct division point that we incrementally shift by 1/3 of the clip duration, producing for each sound file three segmentation variants. To obtain final segments, we use a majority voting strategy: a segment is considered valid if, in another variant, there is a counterpart segment with a mean absolute difference (of both onsets and offsets) below a predetermined threshold ϵ_{vote}. The final onset and offset are then defined as the averages across all valid segments. Segments without support from another co-valid segment are discarded (Figure 2b).

3. DATASET

We evaluate WhisperSeg on five species datasets detailed in Table 1. The zebra finch [15] and Bengalese finch [12, 16] datasets represent bird recordings in controlled, quiet environments. The marmoset [12, 17] and mouse [12, 18] datasets cover mammalian sounds; notably, mouse vocalizations occur in the ultrasonic domain, while marmoset recordings feature elevated noise levels, measured by signal-to-noise power ratio (SNR), due to their origin from a housing room with multiple animals. For human speech, the AVA-Speech dataset, derived from film audio [19, 10], captures diverse noisy conditions. These datasets help assess WhisperSeg's performance over various species, frequencies, and noise levels.

Table 1 shows the multi-species dataset statistics and settings for log-Mel spectrogram computation. For mice vocalizations, due to their ultrasonic nature, the log-Mel spectrogram contains 80 Mel bands starting at a minimum frequency of 35 kHz. The voting threshold ϵ_{vote} is empirically determined based on the segment length and WhisperSeg's time resolution. To prevent naming conflicts, we prefixed each vocalization type with a token representative of the respective species. This prefixing ensures that each vocalization type from every species is given a unique type ID. As a result, the WhisperSeg trained on multi-species datasets can discern both the type of vocalization and the species producing it.

Dataset	No. segments		Segment length (s)		SNR (dB)	No. segment types	audio clip duration D (s)	sampling rate f_s (kHz)	nfft	hop length L_{hop}	minimun frequency (kHz)	maximum frequency (kHz)	WhipserSeg time resolution (s)	ϵ_{vote} (s)
	train	test	train	test										
Zebra Finch (adults)	36,226	3,909	0.15 ± 0.14	0.15 ± 0.14	20.9	1	2.5	32	512	80	0	16	0.005	0.02
Zebra Finch (juveniles)	12,312	980	0.13 ± 0.07	0.13 ± 0.07	11.3	1	2.5	32	512	80	0	16	0.005	0.02
Bengalese Finch	28,562	2,074	0.07 ± 0.03	0.07 ± 0.03	9.0	1	2.5	32	512	80	0	16	0.005	0.02
Marmoset	2,535	702	0.13 ± 0.14	0.11 ± 0.12	6.0	4	2.5	48	1024	120	0	24	0.005	0.02
Mouse	1,807	229	0.06 ± 0.05	0.05 ± 0.04	17.5	1	0.5	300	4096	150	35	150	0.001	0.02
Human (AVA Speech)	21,958	93	3.39 ± 5.17	3.47 ± 3.95	4.3	1	10	16	512	160	0	8	0.02	0.2

Table 1: Statistics for the multi-species dataset and settings when generating log-Mel spectrograms.

Model	No. parameters	Zebra Finch (adults)		Zebra Finch (juveniles)		Bengalese Finch		Marmoset		Mouse		Human (AVA Speech)	
		$F1_{seg}$	$F1_{frame}$	$F1_{seg}$	$F1_{frame}$	$F1_{seg}$	$F1_{frame}$	$F1_{seg}$	$F1_{frame}$	$F1_{seg}$	$F1_{frame}$	$F1_{seg}$	$F1_{frame}$
MLPSeg	554k	0.9601	0.9791	0.7403	0.9374	0.8895	0.9341	0.0752	0.1853	0.8352	0.9033	0.0513	0.8173
DAS	180k	0.9797	0.9883	0.8489	0.9685	0.8977	0.9564	0.3026	0.7120	0.9536	0.9687	0.0461	0.8218
VoxSeg	619k	0.9894	0.9830	0.9171	0.9723	0.9187	0.9450	0.1534	0.8106	0.9491	0.9769	0.2837	0.9300
WSeg-base-S1	74M	0.9940	0.9878	0.9130	0.9776	0.9710	0.9656	0.6651	0.7738	0.9190	0.9613	0.3694	0.9311
WSeg-base-S3	74M	0.9957	0.9901	0.9543	0.9793	0.9741	0.9737	0.7265	0.8123	0.9332	0.9658	0.3968	0.5980
WSeg-large-S1	1550M	0.9945	0.9883	0.9345	0.9800	0.9793	0.9723	0.7744	0.8573	0.9398	0.9798	0.3730	0.9366
WSeg-large-S3	1550M	0.9962	**0.9905**	**0.9583**	**0.9830**	**0.9819**	**0.9781**	0.7988	**0.8742**	0.9437	0.9801	**0.4195**	0.7184
WSeg-large-MS-S1	1550M	0.9942	0.9882	0.9364	0.9796	0.9766	0.9695	0.7933	0.8662	0.9437	0.9810	0.3797	0.9349
WSeg-large-MS-S3	1550M	**0.9966**	**0.9905**	0.9551	0.9827	0.9789	0.9760	**0.8095**	0.8741	**0.9545**	**0.9813**	0.3995	0.6386
Whisper-Large-Framewise	1550M	0.9773	0.9856	0.8353	0.9740	0.9295	0.9654	0.1482	0.5336	0.8172	0.9246	0.3593	**0.9505**

Table 2: Comparison of animal sound segmentation and VAD performance. "WSeg-large-S3" is the large WhisperSeg variant employing a 3-variant majority voting strategy (Figure 2b), while "S1" corresponds to a simple concatenation of (1-variant) segments as shown in Figure 2a. "MS" denotes a universal model trained across all species with species-specific test scores. For the zebra finch dataset, we combined the training datasets of both adults and juveniles but presented test results separately for each group. Except for DAS, models were trained and tested 5 times, with average test scores reported.

4. EXPERIMENTS AND DISCUSSION

We fine-tuned WhisperSeg for next-token prediction using cross-entropy loss. We used two model sizes: a base version with 74M parameters (whisper-base) and a larger 1.55B parameter variant (whisper-large) [13]. We explored training species-specific WhipserSeg models and training a single WhipserSeg on all five species. Training parameters encompassed three epochs, a 3e-6 learning rate introduced with a linear warm-up over 100 model update steps followed by a linear decay to zero, and a batch size of 4. We augmented the training process by randomly choosing spectrogram onsets during training, to increase the number of positions at which vocalizations are sampled. Subsequent sections detail baselines, evaluation metrics, and performance analysis across species, assessing WhisperSeg's adaptability.

Baselines: We compared WhisperSeg with 1) VoxSeg [10], which uses a time-distributed CNN to convolve 25-column wide frames, shifting by 2 columns each time. Given a padded 1000-column spectrogram, this results in 500 hidden vectors, which are subsequently processed by a Bi-LSTM to produce 500 encoded outputs for frame-wise classification (e.g., voice vs noise); 2) DAS [12], which introduces a learned FFT block to transform audio samples into features that are subsequently processed by Temporal Convolutional Networks (TCNs) [20];

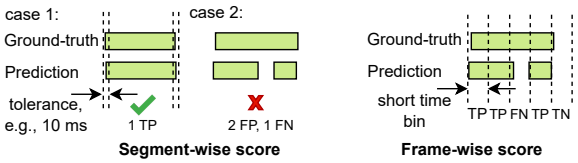


Fig. 3: Illustrations of evaluation metrics.

3) MLPSeg, an additional simple baseline we implemented that classifies a 25-column wide spectrogram frame through four fully connected layers. While these models diverge in contextual scope, they commonly predict labels for brief time frames, from which they derive onset, offset, and vocalization types. Contrarily, WhisperSeg directly infers these parameters through token generation without explicitly resorting to a serial computation at a high temporal resolution.

Evaluation Metrics: We evaluated segmentation performance using segment-wise ($F1_{seg}$) and frame-wise ($F1_{frame}$) F1 scores. $F1_{seg}$ aligns model predictions with ground truth, considering a tolerance of segment onset, offset, and type consistency of 200 ms for humans and 10 ms for all other species. From this alignment, we compute numbers of true positive (TP), false positive (FP), and false negative (FN) test samples to obtain the F1 score (Figure 3). $F1_{frame}$ divides time into short bins of 10 ms for humans and 1 ms for other species

and compares predicted and actual vocal segment types bin-by-bin. Notably, $F1_{seg}$ is more stringent, prioritizing segment integrity and penalizing minor segmentation errors.

Results and Discussion: WhisperSeg outperformed baseline models in both $F1_{seg}$ and $F1_{frame}$ on human and animal datasets, especially on noisy datasets from Marmosets and Humans. While baseline models achieve commendable $F1_{frame}$ scores, they falter on $F1_{seg}$ scores under noisy conditions, often producing fragmented predictions or brief false positives. In contrast, WhisperSeg's performance is stable on both metrics, emphasizing its accurate onset and offset predictions and strong resilience to noise. Notably, a universal WhisperSeg model (WSeg-large-MS), trained across all species, matches the performance of species-specific models (Table 2), highlighting its versatile generalization capability. By exception, the fine-tuning of a given WhisperSeg model on animal datasets provides no clear advantage for human VAD, which could be either due to the intricacies of the human data or the lack of pre-training on animal data.

WhisperSeg's superiority arises from its transformer architecture and numerous parameters, excelling particularly in noisy environments. In addition, when compared with a frame-wise Whisper solution, where both the encoder and decoder of Whisper receive the spectrogram and the decoder's output is used for frame-wise classification, the "Whisper-Large-Framewise" model underperformed compared to "WSeg-Large-S1", Table 2. This suggests that the token-based approach of WhisperSeg is a key factor in its superior performance, rather than just its large model size.

We observed that longer input audio contexts significantly enhance vocal segmentation accuracy. Models with extended audio contexts, such as VoxSeg and WhisperSeg, consistently outperformed those with shorter contexts like MLPSeg, especially in low signal-to-noise ratio (SNR) scenarios. In further experiments with WhisperSeg using varying input spectrogram lengths, we noted consistent improvements in performance. Increasing the context lengths from 100, 200, to 1000 columns resulted in a steady rise in $F1_{seg}$ scores: from 0.081, 0.222, to 0.380 in humans, and from 0.712, 0.832, to 0.944 in mice. In other species, the improvements were also exhibited but less substantial. These findings underscore the benefits of longer audio contexts for WhisperSeg's performance.

In addition, the majority voting approach enhances $F1_{seg}$ scores when comparing WSeg-xxx-S3 to WSeg-xxx-S1 (Table 2). A drop in $F1_{frame}$ for the Human dataset after majority voting could be due to the large variations in segment duration, which influence predictions and may lead to segment exclusion during voting.

We examined WhisperSeg's adaptability to new datasets and its reliance on training set size by fine-tuning WhisperSeg on a new species, starting with small subsets (50 to 1000 segments) and eventually using the full set. Notably, when selecting a zebra finch training subset of N segments, we ensured an even split between adult and juvenile vocal segments to enforce an age balance in training examples. We trained three models on the new species: a) VoxSeg, b) WhisperSeg-large initialized from a pretrained ASR checkpoint [13] with tri-variant majority voting (WS-L-S3), and c) WhisperSeg-large pretrained on multi-species data (excluding the new species data currently being tested) with majority voting (WS-L-MS-S3, Figure 4).

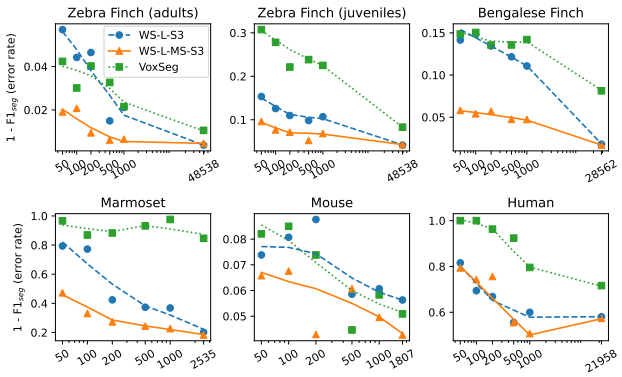


Fig. 4: Comparison of performance on the complete test dataset of a given new species. The x-axis shows the training subset size of the new species (in segment count), with the rightmost point representing the full training set size. The y-axis depicts $1 - F1_{seg}$ as an error rate measure.

The multi-species WhisperSeg (WS-L-MS-S3) consistently outperformed baselines, even with minimal training data. Notably, by leveraging a mere 50 segments from a novel species, WS-L-MS-S3 achieves an error rate reduction of over 50% when compared with baseline models on zebra finch, Bengalese finch, and marmoset datasets. On zebra finch and marmoset datasets, WS-L-MS-S3 converges to performance levels seen in full-size training but only requires 200 segments, demonstrating its efficiency in rapid adaptability. This evidence underscores the profound effectiveness of our proposed multi-species Whisper-based segmenter, showcasing its promise in real-world applications where data from unfamiliar individuals or species is introduced.

5. CONCLUSION

In this study, we successfully adapted the Whisper Transformer, originally designed for human ASR, to Voice Activity Detection (VAD) for humans and animals. Our strategy leverages the prediction of onset and offset timestamps, expressed as column indices within fixed-size input spectrograms. This spectrogram-level preprocessing facilitates the universal application of the Whisper model across diverse animal species. Empirical results underscore a consistent superiority across all tested species and emphasize the advantages of crafting a multi-species vocalization segmenter with enhanced generalization and adaptation capabilities.

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