

UAVM: A Unified Model for Audio-Visual Learning

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Abstract—Conventional audio-visual models have independent audio and video branches. We design a *unified* model for audio and video processing called Unified Audio-Visual Model (UAVM). In this paper, we describe UAVM, report its new state-of-the-art audio-visual event classification accuracy of 65.8% on VGGSound, and describe the intriguing properties of the model.

Index Terms—audio-visual learning, unified model

I. INTRODUCTION

Humans perceive and understand their world by combining different sensory input modalities including sound and vision. To enable AI systems to have a similar ability, *audio-visual multi-modal learning* has been extensively studied [1], [2]. Due to the inherent differences between audio and video, conventional audio-visual learning methods typically either use handcrafted modality-specific features or have two *independent* audio and visual branches with different architectures, training schemes, and model weights [3], [4], [5], [6].

This modal-specific approach began to change after the Transformer [7] architecture was found to be effective for various tasks and modalities including audio [8], [9] and video [10], [11]. With Transformer, we are witnessing a trend of unifying increasingly more model components for processing different modalities. Specifically, the Perceiver [12] model shows that it is possible to handle arbitrary configurations of different modalities using a *unified model architecture*, although the training method is still modality-specific. In addition to unified model architecture, data2vec [13] further demonstrates that a *unified training scheme* can be applied to different modalities to obtain state-of-the-art performance. Despite the unified model architecture and training scheme, the models of different modalities are trained independently and have independent weights. To go one step even further, SkillNet [14], EAO [15], VATT [16], and PolyViT [17] share part of the *model weights* among different modalities. Specifically, SkillNet [14] and EAO [15] are single models that handle multiple modalities, but different parts of the model weights are specialized for processing different modalities. VATT [16] uses a modality-agnostic, single-backbone Transformer and shares weights among different modalities. However, the training scheme of VATT is modality-asymmetric. PolyViT [17] shares model weights except the input tokenizer and task head.

The motivations for cross-modal model unification are multifold. First, it minimizes the use of handcrafted priors and inductive biases for each individual modality, which is more data-driven and saves manual effort. Second, a unified model that handles multiple modalities can be more parameter-efficient than a set of modality-specific models. Third, unified

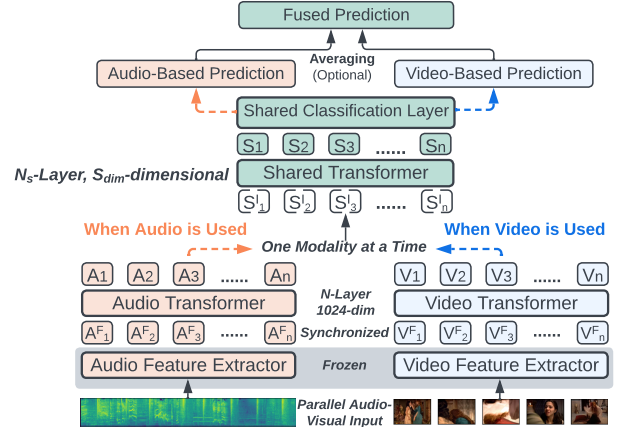


Fig. 1. An illustration of the unified audio-visual model (UAVM).

models are ideal *foundation models* [18] that can be adapted to a wide range of downstream tasks of multiple modalities.

However, existing efforts only unify part of the model components, and the differences between a unified model and a modal-independent model remain unclear. Are unified models just two modal-independent models “glued” together? In this paper, we build a Unified Audio-Visual Model (UAVM) that unifies almost everything: 1) the audio and visual branches have the same architecture; 2) the audio and visual branches share weights for high layers including the final decision layer; 3) both branches are trained with a unified algorithm. On VGGSound [19], the UAVM model achieves a new state-of-the-art (SOTA) accuracy for the audio-visual event classification task. Besides the performance, we conduct a series of analyses and show the unique properties of the unified model.

II. THE UNIFIED AUDIO-VISUAL MODEL

A. UAVM Model Architecture

The unique aspect of the UAVM is the shared Transformer and classification layer whose weights are shared between different modalities. This feature enables the UAVM to process both audio and video independently. Before the shared Transformer, we still have modal-specific feature extractors and optional modal-specific Transformers for each modality.

As shown in Figure 1, the audio or video is first input to the corresponding modality-specific feature extractor. Both audio and video feature extractors are ConvNeXt-Base [20]. For video, we use ImageNet pretrained ConvNeXt as the feature extractor. We uniformly sample RGB frames from the video at 3 FPS, input each frame to the ConvNeXt-Base, and get the mean-pooled penultimate layer representation of 1024-dimensions, i.e., for each 10-second video, the video features are 30 1024-dimensional vectors $\{V_1^F, \dots, V_{30}^F\}$. For audio, we train a ConvNeXt using in-domain audio data (AudioSet or VGGSound) with ImageNet initialization [21] and then use it

Algorithm 1 UAVM Model Training and Inference**Require:** Dataset $\mathcal{D} = \{A, V, \text{Label}\}$, UAVM Model $\mathcal{M} = \{\theta_a, \theta_v, \theta_s\}$ **Training** ($\mathcal{D}, \mathcal{M}, \lambda_{\text{MT}}$)

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1: while  $i < \max \text{ training iteration}$  do
  ▷ One modality is used to train the model at an iteration
2:   if  $\text{unif}(0, 1) < \text{modality training weight } \lambda_{\text{MT}}$  then
3:     sample a batch of audio  $\{A_i, \text{Label}_i\}$ 
4:      $\text{Pred}_a = \mathcal{M}_{\{\theta_a, \theta_s\}}(A_i)$ 
5:      $\mathcal{L} = \text{Loss}(\text{Label}, \text{Pred}_a)$ 
6:     backpropagate and update  $\{\theta_a, \theta_s\}$ 
7:   else
8:     sample a batch of video  $\{V_i, \text{Label}_i\}$ 
9:      $\text{Pred}_v = \mathcal{M}_{\{\theta_v, \theta_s\}}(V_i)$ 
10:     $\mathcal{L} = \text{Loss}(\text{Label}, \text{Pred}_v)$ 
11:    backpropagate and update  $\{\theta_v, \theta_s\}$ 
12: return  $\mathcal{M}$ 

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Inference ($\{A, V\}, \mathcal{M}$)

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12: if  $A \neq \text{None}$  and  $V \neq \text{None}$  then
13:    $\text{Pred} = (\mathcal{M}_{\{\theta_a, \theta_s\}}(A) + \mathcal{M}_{\{\theta_v, \theta_s\}}(V))/2$ 
14: else if one modality is missing then
15:    $\text{Pred} = \mathcal{M}_{\{\theta_v, \theta_s\}}(V)$  when  $A == \text{None}$ 
16:    $\text{Pred} = \mathcal{M}_{\{\theta_a, \theta_s\}}(A)$  when  $V == \text{None}$ 
17: return  $\text{Pred}$ 

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as the audio feature extractor. Each 10-second waveform is first converted to a 1000×128 log Mel filterbank (fbank) feature vector computed with a 25ms Hanning window every 10ms and input to the feature extractor. The output of the penultimate layer of ConvNeXt is a 30 (time) \times 4 (frequency) \times 1024 tensor. We apply a frequency mean pooling to produce 30 1024-dimension audio features $\{A_1^F, \dots, A_{30}^F\}$. Note that we intend to make the audio and video features synchronized. Both feature extractors are frozen during training.

We then input the L2-normalized $\{A^F\}$ or $\{V^F\}$ to corresponding N -layer modality-specific Transformers to produce $\{A_1, \dots, A_{30}\}$ or $\{V_1, \dots, V_{30}\}$. After that, either $\{A_1, \dots, A_{30}\}$ or $\{V_1, \dots, V_{30}\}$ is used as an input $\{S_1^I, \dots, S_{30}^I\}$ to the shared, modality-agnostic Transformer of N_s layers. This approach is fundamentally different from concatenating audio and visual tokens as input to the shared Transformer as in MBT [22] and Merlot Reserve [23]. We mean-pool the output of the shared Transformer $\{S_1, \dots, S_{30}\}$ and input it to a shared linear classification layer. When only one modality is input, we use the output of the shared linear classification layer as the single-modality prediction; when both modalities are input, we run one forward pass for each modality and average the outputs of the two passes as a fused prediction (Algorithm 1). In other words, UAVM is robust to missing modalities.

Throughout the experiments, all Transformer layers have 4 attention heads and the modality-specific Transformer always has an embedding dimension of 1024. We tune the embedding dimension of the shared Transformer S_{dim} from 16 to 1024 to control its capacity. We fix the total number of Transformer layers to 6 ($N + N_s = 6$), and tune the number of modality-specific Transformer layers, N , and shared Transformer layers, N_s , from 0 to 6 to control the model unification level. When $N = 0$ and $N_s = 6$, the audio and visual models are maximally unified; when $N = 6$ and $N_s = 0$, the audio and visual models are completely independent including the classification layer. We set the model with $N = N_s = 3$ and $S_{\text{dim}} = 1024$ as the base UAVM model.

B. UAVM Model Training

We train UAVM with Algorithm 1. In each training iteration, only one modality is used. This is implemented by using a modality training weight λ_{MT} and uniform sampling, i.e., audio and video have λ_{MT} and $1 - \lambda_{\text{MT}}$ probability to be used, respectively. In other words, UAVM does not explicitly leverage the audio-visual correspondence information and can be trained with unpaired audio and video data (though in this paper, we focus on parallel datasets for simplicity). By default, we use modality training weight $\lambda_{\text{MT}} = 0.5$, i.e., the audio and video have the same chance to be used.

As in prior work [8], [21], [22], we train the model with mixup [24], balanced sampling, label smoothing, and random time shifts. These training techniques are applied to both modalities with exactly the same hyperparameters, i.e., we use a unified training pipeline for the two modalities.

III. EXPERIMENT AND DISCUSSION**A. Experiment Settings**

1) *Dataset*: We use two widely-used datasets for audio and video event classification: AudioSet [25] and VGGSound [19]. AudioSet is a collection of 2M 10-second YouTube video clips labeled with the sounds that the clip contains from a set of 527 labels. Due to changes in video availability, we downloaded 1,772,023 training and 17,249 evaluation audio and video samples for our experiment. VGGSound [19] is a collection of 200K 10-second YouTube video clips annotated with 309 classes. We download 183,727 training and 15,446 test samples for our experiments. One advantage of VGGSound is that the sound source is always visually present in the video clip. Also, its moderate size allows us to conduct extensive experiments with our computational resources. Therefore, while we compare UAVM performance with existing methods on both AudioSet and VGGSound, we conduct all ablation studies and analysis on VGGSound only.

2) *Training Details*: For all experiments, we train the model with a batch size of 144 and the Adam optimizer [26]. For the main experiments, we use an initial learning rate of $1e-5$ and $5e-5$ for AudioSet and VGGSound, respectively, and decrease the learning rate with a factor of 0.5 every epoch. We train the model for 10 epochs and report the last epoch result. For ablation studies, we tune hyper-parameters to ensure a fair comparison. We repeat all experiments 3 times with different random seeds and report the mean and standard deviation. The standard deviation is shown as shaded areas in Figure 2-5.

B. Model Performance Comparison

We compare the performance of three base models:

1. **UAVM**. The base UAVM described in Section II-A with 3 modal-specific Transformer layers and 3 shared Transformer layers ($N = N_s = 3$) and $S_{\text{dim}} = 1024$.

2. **Modal-Independent Model**. The base UAVM without shared Transformer layers (i.e., $N = 6, N_s = 0$), so the audio and visual models are completely independent including the classification layer.

3. **Cross-Modal Attention Model**. The base UAVM model with 3 modal-specific Transformer layers and 3 shared Transformer layers ($N = N_s = 3$) but instead of inputting one

TABLE I
MODEL PERFORMANCE COMPARISON ON VGGSound AND AudioSet.
(* SINGLE-MODAL MODEL TRAINED INDEPENDENTLY.
† MODALITY-MISSING RESULTS OF A MULTI-MODAL MODEL.)

VGGSound (Top-1 Accuracy, %)	Audio	Video	Fusion
Chen <i>et al.</i> [19]	48.8	-	-
AudioSlowFast [27]	50.1	-	-
MBT [22]	52.3*	51.2*	64.1
Our Cross-Modal Attention Model	-	-	62.9±0.2
Our Modal-Independent Model	56.5±0.1*	49.7±0.2*	65.7±0.2
Our UAVM Model	56.5±0.1†	49.9±0.2†	65.8±0.1
Full AudioSet (mAP)	Audio	Video	Fusion
GBLend [28]	32.4*	18.8*	41.8
Attn Audio-Visual [29]	38.4*	25.7*	46.2
Perceiver [12]	38.4*	25.8*	44.2
MBT [22] (w/ 500k training samples)	44.3*	32.3*	52.1
Our Cross-Modal Attention Model	-	-	50.4±0.1
Our Modal-Independent Model	45.5±0.0*	26.8±0.1*	48.1±0.1
Our UAVM Model	45.6±0.0†	27.4±0.1†	48.0±0.0

modality at a time to the shared Transformer, this model *concatenates* the outputs of modal-specific Transformers $\{A\}$ and $\{V\}$ together and inputs $\{A, V\}$ to the shared Transformer. This model only works when both modalities are input.

We show the results in Table I. Key conclusions are as follows: First, compared with the modal-independent model, UAVM achieves almost the same fusion performance when both modalities are input, and even slightly better results when a single modality is input, demonstrating the feasibility of using a single network for two different modalities. Second, comparing UAVM with the “MBT-style” cross-modal attention model, we find a performance discrepancy between the datasets, i.e., UAVM is noticeably better on VGGSound while the cross-modal attention model is noticeably better on AudioSet. This is potentially because the sound source is always visually present in VGGSound videos but not in AudioSet videos. The UAVM strategy of giving equal weight to both modalities performs better than cross-modal attention models that could be dominated by one modality on tasks like VGGSound (video is always informative), but worse on AudioSet (video is not always informative). Finally, Transformer models with pretrained frozen features are a strong baseline with low computational cost. As a consequence, UAVM achieves a new SOTA performance on VGGSound, and outperforms all previous methods except MBT [22] on AudioSet. Note MBT also uses ImageNet pretraining. Compared with MBT, the UAVM Transformer input sequence length is much shorter (30 vs 1,500+), making it more computationally efficient since the Transformer has quadratic complexity w.r.t. the input.

We then conduct a series of ablation studies. We set models with $N = N_s = 3$ and $S_{dim} = 1024$ as the base UAVM and change one factor at a time to observe the performance change. First, we constrain the shared Transformer embedding dimension S_{dim} to smaller values to lower its capacity for UAVM and compare it with other models. For a fair comparison, the modal-independent and cross-modal attention models also have three 1024-dimensional Transformer layers and three S_{dim} -dimensional Transformer layers. With a limited capacity, the shared Transformer is forced to share neurons for two modalities. As shown in Figure 2 (upper), we find model performance generally improves with larger S_{dim} , but even

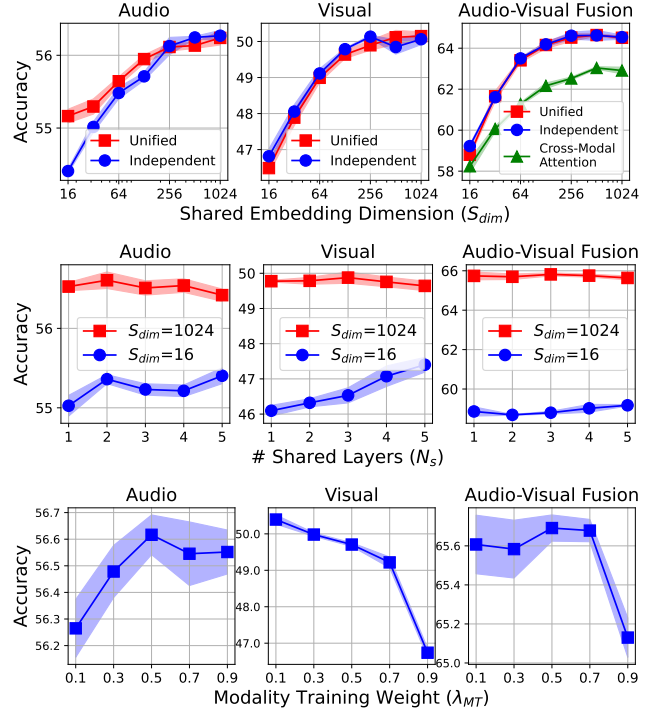


Fig. 2. The audio-based, video-based, and fused accuracy on VGGSound with various shared embedding dimension S_{dim} (upper), number of shared layers N_s (middle), and modality training weight λ_{MT} (lower).

when S_{dim} is very small, UAVM still performs similarly or even better than the modal-independent model. Second, we tune the number of shared layers N_s to change the level of model unification. Note that we fix the total number of Transformer layers to be 6, i.e., $N + N_s = 6$. As shown in Figure 2 (middle), when S_{dim} is small, the single-modality accuracy improves with more shared layers while when S_{dim} is large, the number of shared layers does not impact the performance much. Finally, as shown in Figure 2 (lower), even though audio and video are not equally informative for the event classification task (reflected in different accuracies), using a 0.5 λ_{MT} leads to optimal fusion results.

C. Unified Audio-Visual Representation Space

One core question about the unified model is if it indeed encodes two modalities in a unified latent space, or just processes each modality with a part of its parameters. We explore this by checking if a logistic regression model can successfully classify the input modality based on the mean-pooled penultimate layer representation of the UAVM model. Specifically, we use half of the VGGSound test set to train a logistic regression model and use the other half for testing. As shown in Figure 3.A, when the shared Transformer has a small S_{dim} and limited capacity, a logistic regression model is unable to predict the input modality based on the penultimate layer representation, in other words, the two modalities are encoded in a unified space. However, the modality classification accuracy gradually increases with S_{dim} , indicating that the model, though with shared weights, tends to encode two modalities in separate spaces when it has redundant capacity. Nevertheless, we do not see one or a small number of dimensions of the representation specifically encoding the input modality information because the classifier does not

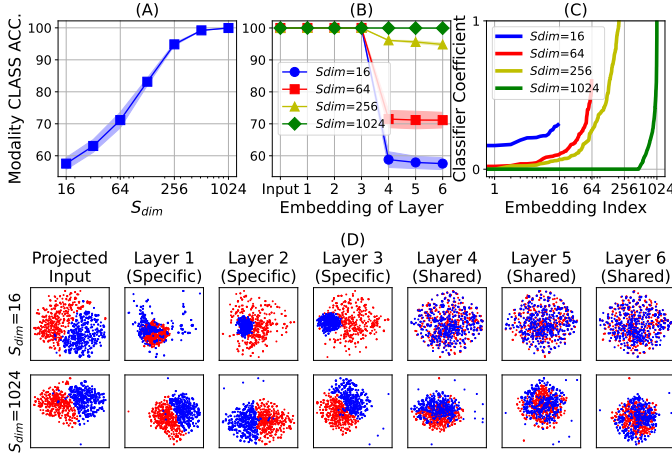


Fig. 3. (A) Modality classification accuracy based on the last layer representation of UAVM with various S_{dim} . (B) Modality classification accuracy based on intermediate representations of each layer of UAVM models. (C) The sorted modality classifier coefficients. (D) t-SNE plot of the intermediate representations of each layer with audio input (red) and video input (blue).

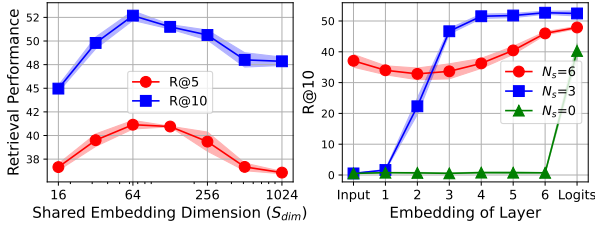


Fig. 4. (Left) A-V retrieval performance based on the last shared Transformer layer representation of UAVM models ($N_s = 3$) with various shared embedding dimensions S_{dim} . (Right) A-V retrieval performance based on the representation of various layers of models with $N_s = 0$ (fully independent model), $N_s = 3$, and $N_s = 6$ (fully unified model) and $S_{dim} = 64$.

have a dominant coefficient (Figure 3.C). Further, as shown in Figure 3.B, the modality of the input and modal-specific layer (layer 1-3) representations can be perfectly classified but the modality classification accuracy drops suddenly with the first shared layer (layer 4) representation, indicating that it is the shared Transformer that maps the two very different inputs to a unified space. The above findings can also be confirmed with the t-SNE plot of the representations of each layer with audio and video input shown in Figure 3.D.

D. Audio-Visual Representation Correspondence

Another interesting difference between UAVM and modal-independent models is the audio-visual representation correspondence. As a probing task, we calculate the A-V retrieval recall based on cosine similarity on a 1.5k subset of the VGGSound evaluation set (309 classes, 5 samples per class). Note that with reasonable single-modal classification accuracies, the audio and video of the same class can be naturally retrieved from each other. However, as shown in Figure 4 (right), for the modal-independent model ($N_s = 0$), the A-V retrieval recall is close to 0 for representations of all layers except the final linear classification head (i.e., prediction logits) while for UAVMs, the A-V retrieval recall is much higher for representations for front layers (the more shared layers, the earlier a high A-V retrieval recall is achieved), indicating that the shared Transformer layers more effectively propagate the supervision signal to front layers. More interestingly, the A-V retrieval

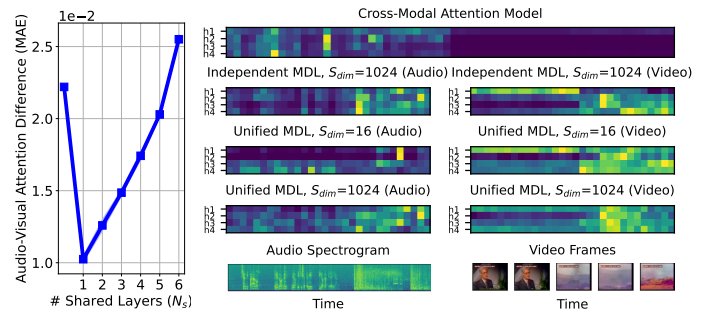


Fig. 5. (Left) Difference between the last layer attention maps with (paired) audio and video input for UAVM with various N_s . (Right) Temporal attention heatmap of 4 attention heads for a sample input (label: firing cannon).

recall of UAVMs ($S_{dim} = 64$) is about 20% higher than the modal-independent model while their classification accuracies are similar, indicating that the unified models learn A-V correspondences beyond intra-class correspondence. Note that the UAVM does not see simultaneous audio and video pairs, nor has any explicit audio-visual correspondence loss been applied during training. As shown in Figure 4.A, the A-V retrieval recall decreases with the increase of shared embedding dimension S_{dim} although the classification accuracy improves with larger S_{dim} . This is consistent with the conclusion in Section III-C that a UAVM with redundant capacity tends to be closer to modal-independent models.

E. Attention Maps

Even when the audio and video features are temporally synchronized, the informative part for event classification could be different. We show the temporal attention heatmap of 4 attention heads of the last Transformer layer in Figure 5 (right). The UAVM, even when the shared Transformer has very limited capacity ($S_{dim} = 16$), can pay attention to different parts of the audio and video, demonstrating its ability to process two modalities simultaneously. A cross-modal attention model, however, could be dominated by one modality while the other modality still contains useful information. Interestingly, by quantitatively calculating the mean absolute error (MAE) between attention maps of paired audio and video inputs, we find that the more modal-specific layers (i.e., fewer shared layers N_s), the smaller the difference between the audio and video attention maps (Figure 5 (left)), indicating the modal-specific Transformers layers are forced to temporally align the informative part of the two modalities, so that succeeding shared Transformer layers can have a relatively more modal-agnostic attention map.

IV. CONCLUSION

In this work, we explore unifying model architecture, weight, and training for the audio and video branches of a multi-modal model. Performance-wise, UAVM is similar to a modal-independent model, and outperforms cross-modal attention models on VGGSound. The capacity of the weight-sharing network impacts the behavior of UAVM. When the capacity is constrained, UAVM shows its unique properties, e.g., it maps audio and video in a unified latent space, and the intermediate representations of paired audio and video input have higher correspondence than modal-independent models.

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