EMOVA^C: EMPOWERING LANGUAGE MODELS TO SEE, HEAR AND SPEAK WITH VIVID EMOTIONS

Anonymous authors

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ABSTRACT

GPT-4o, an omni-modal model that enables vocal conversations with diverse emotions and tones, marks a milestone for omni-modal foundation models. However, empowering Large Language Models to perceive and generate images, texts, and speeches end-to-end with publicly available data remains challenging in the opensource community. Existing vision-language models rely on external tools for the speech processing, while speech-language models still suffer from limited or even without vision-understanding abilities. To address this gap, we propose **EMOVA** $(\underline{EM}$ -otionally **Omni-present Voice Assistant**), to enable Large Language Models with end-to-end speech capabilities while maintaining the leading vision-language performance. With a *semantic-acoustic disentangled* speech tokenizer, we notice surprisingly that omni-modal alignment can further enhance vision-language and speech abilities compared with the corresponding bi-modal aligned counterparts. Moreover, a lightweight style module is proposed for flexible speech style controls (*e.g*., emotions and pitches). For the first time, EMOVA achieves state-of-the-art performance on both the vision-language and speech benchmarks, and meanwhile, supporting omni-modal emotional spoken dialogue. Demos are available in the project page: <https://emova-anonymous.github.io/>.

1 INTRODUCTION

034 035 036 037 038 039 040 041 OpenAI GPT-4o [\(OpenAI,](#page-13-0) [2024\)](#page-13-0), a new milestone for omni-modal foundation models, has rekindled people's attentions on intelligent assistants that can *see* (*i.e*., perceiving fine-grained visual inputs), *hear* (*i.e*., understanding vocal instructions) and *speak* (*i.e*., generating vocal responses) simultaneously. Most existing Multi-modal Large Language Models (MLLMs) focus on two modalities only, either vision-language [\(Bai et al.,](#page-10-0) [2023;](#page-10-0) [Li et al.,](#page-12-0) [2024a\)](#page-12-0) or speech-language [\(Chu et al.,](#page-10-1) [2024;](#page-10-1) [Xie &](#page-14-0) [Wu,](#page-14-0) [2024\)](#page-14-0), demonstrating severe demands for omni-modal models with visual, language and speech abilities. How to effectively empower Large Language Models (LLMs) to process omni-modal data in an end-to-end manner remains an open question.

042 043 044 045 046 047 048 049 050 051 052 053 Existing omni-modal LLMs [\(Chen et al.,](#page-10-2) [2024b;](#page-10-2) [Fu et al.,](#page-11-0) [2024b\)](#page-11-0) generally build upon Vision LLMs and integrate the speech modality by adopting a speech encoder like Whisper [\(Radford et al.,](#page-14-1) [2023\)](#page-14-1), which extracts **continuous** features from speech, similar to how images are processed, and enables speech understanding. However, these models still rely on external Text-to-Speech (TTS) tools for generating speech responses, limiting their ability to support real-time interactions. AnyGPT [\(Zhan](#page-14-2) [et al.,](#page-14-2) [2024\)](#page-14-2), instead, opts for a fully discretization manner, which first discretizes all data modalities (*i.e*., images, texts, and speeches), followed by omni-modal auto-regressive modeling. This enables AnyGPT to handle multiple modalities with a **unified end-to-end** framework, facilitating **real-time** interactions with the help of streaming decoding. However, the discrete vision tokenizer adopted by AnyGPT struggles to capture visual details, especially for high-resolution images, making it far behind its continuous counterparts on vision-language benchmarks. Moreover, none of the existing works explore speech style controls (*e.g*., emotions and pitches) with LLMs. Therefore, our question arises: *How to build an end-to-end omni-modal LLM enabling spoken dialogue with vivid emotions while maintaining state-of-the-art vision-language performance?*

and quantized into discrete units. Given the omni-modal inputs, **EMOVA** can generate both textual Figure 1: Model architecture of EMOVA. The vision encoder extracts continuous visual features, which are projected into the text embedding space as visual tokens, while the input speech is encoded and speech responses with vivid emotional control. Check Sec. [3](#page-2-0) for more details.

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074 075 076 077 078 079 080 081 082 083 084 085 086 087 088 In this paper, we propose EMOVA (EMotionally Omni-present Voice Assistant), a novel end-toend omni-modal LLM with state-of-the-art vision-language and speech capabilities while supporting emotional spoken dialogue. Fig. [1](#page-1-0) shows an overview of the model framework. A *continuous* vision encoder captures the fine-grained visual details, while the *discrete* speech tokenizer and detokenizer enable the end-to-end speech understanding and generation. Specifically, the speech-to-unit (S2U) tokenizer converts the input speech waveforms into discrete speech units as LLM inputs, while the unit-to-speech (U2S) detokenizer reconstructs the speech waveforms from the LLM's output speech units. To seamlessly integrate the speech modality with LLMs, we meticulously design a semanticacoustic disentangled speech tokenizer to decouple the semantic contents and acoustic styles of the input speeches [\(Tao et al.,](#page-14-3) [2024\)](#page-14-3), where 1) *semantic content* (*i.e*., what it says) captures the semantic meanings of input speeches, which is finally discretized and aligned with LLMs, while 2) *acoustic style* (*i.e*., how it says) captures the diverse speech styles (*e.g*., emotions and pitches). Utilizing the semantic-acoustic disentanglement of our speech tokenizer, we further introduce a lightweight style module to support spoken dialogue with vivid emotions and pitches. As in Sec. [4.1,](#page-4-0) this innovative disentanglement design better facilitates the modality alignment between texts and speeches while maintaining flexibility for diverse speech style controllability and personalization.

090 091 092 093 094 095 096 With the end-to-end omni-modal architecture of **EMOVA**, we empirically demonstrate that publicly available bi-modal image-text and speech-text data are sufficient for omni-modal alignment, utilizing the text modality as a bridge. This eliminates the need for omni-modal data (*i.e*., image-text-speech), which is usually scarce. Surprisingly, we find that omni-modal alignment can further improve both vision-language and speech capabilities through joint optimization, even when compared with their bi-modal aligned counterparts. Finally, only a small amount of mixed-modality samples are required to teach the model to respond in the desired format. For the first time, EMOVA achieves state-ofthe-art performance on both vision-language and speech benchmarks (see Table [1](#page-2-1) for comparisons).

- **097 098** The main contributions of this work contain three parts:
	- 1. We propose EMOVA, a novel end-to-end omni-modal LLM that can see, hear and speak. We use a continuous vision encoder and a semantic-acoustic disentangled speech tokenizer for seamless omni-modal alignment and diverse speech style controllability.
	- 2. We introduce an efficient text-centric omni-modal alignment which can further improve the vision-language and speech capabilities, even compared with the corresponding bi-modal aligned counterparts (*i.e*., image-text only and speech-text only alignment).
- **105 106 107** 3. For the first time, our EMOVA achieve state-of-the-art comparable performance on both the vision-language and speech benchmarks simultaneously, while supporting flexible spoken dialogues with vivid emotions.

108 109 110 Table 1: Comparison among Multi-modal Large Language Models. Our EMOVA is the very first unified Omni-modal Large Language Model capable of emotional spoken dialogue with stateof-the-art vision-language and speech capabilities simultaneously.

2 RELATED WORK

125 126 127 128 129 130 131 132 133 134 135 136 Vision Large Language Models (VLLMs) integrate vision modality into Large Language Models (LLMs) [\(Touvron et al.,](#page-14-4) [2023;](#page-14-4) [Chen et al.,](#page-10-3) [2023b\)](#page-10-3), enabling advanced understanding and reasoning over visual instructions [\(Liu et al.,](#page-12-1) [2024b;](#page-12-1) [Bai et al.,](#page-10-0) [2023;](#page-10-0) [Gou et al.,](#page-11-1) [2023;](#page-11-1) [2024\)](#page-11-2). Recent efforts in VLLMs can be broadly categorized into three directions, including 1) *Vision encoders* [\(Oquab et al.,](#page-13-1) [2023;](#page-13-1) [Chen et al.,](#page-10-4) [2021b;](#page-10-4) [2023a\)](#page-10-5) are enhanced and aggregated for robust representations [\(Lin et al.,](#page-12-2) [2023;](#page-12-2) [Li et al.,](#page-12-3) [2024b;](#page-12-3) [Tong et al.,](#page-14-5) [2024\)](#page-14-5). 2) *High-resolution* methods are proposed to overcome the fixed resolution of pre-trained vision encoders (e.g., 336×336 for CLIP [\(Radford et al.,](#page-14-6) [2021\)](#page-14-6)), empowering LLMs to perceive fine-grained visual information [\(Liu et al.,](#page-12-4) [2024a;](#page-12-4) [Dong et al.,](#page-10-6) [2024;](#page-10-6) [Huang et al.,](#page-11-3) [2024;](#page-11-3) [Luo et al.,](#page-13-2) [2024\)](#page-13-2). 3) *High-quality instruction data* is essential for the VLLMs to generate accurate and well-formed responses following instructions (Laurencon et al., [2024;](#page-12-5) [Li et al.,](#page-12-0) [2024a;](#page-12-0) [Chen et al.,](#page-10-2) [2024b\)](#page-10-2). In this paper, besides achieving state-of-the-art vision-language performance, we further introduce speech understanding and generating capabilities into our **EMOVA**.

137 138 139 140 141 142 143 144 145 Speech Large Language Models (SLLMs) empower speech interaction with LLMs. *Continuous SLLMs* [\(Wu et al.,](#page-14-7) [2023;](#page-14-7) [Chu et al.,](#page-10-1) [2024\)](#page-10-1) utilize the speech encoders [\(Radford et al.,](#page-14-1) [2023\)](#page-14-1) to extract continuous speech embeddings for LLM, which, however, only support speech understanding, relying on external TTS modules for speech generation, and therefore, hampering real-time interaction. *Discrete SLLMs* [\(Zhang et al.,](#page-15-0) [2023a\)](#page-15-0), instead, first discretize speech signals with speech tokenizers, followed by auto-regressive modeling. Recent works [\(Fang et al.,](#page-11-4) [2024;](#page-11-4) [Xie & Wu,](#page-14-0) [2024\)](#page-14-0) further combine the continuous speech encoders with the discrete speech tokenizers for better performance. Although effective, none of the existing works explore speech style controllability in SLLMs (*e.g*., genders, emotions, and pitches), which is essential for real-life spoken dialogue.

146 147 148 149 150 151 152 153 Omni-modal Large Language Models support visual, text, and speech capabilities with a unified architecture simultaneously. Similar to the continuous SLLMs, InternOmni [\(Chen et al.,](#page-10-2) [2024b\)](#page-10-2) and VITA [\(Fu et al.,](#page-11-0) [2024b\)](#page-11-0) connect a speech encoder with VLLMs, supporting speech understanding only. Instead, AnyGPT [\(Zhan et al.,](#page-14-2) [2024\)](#page-14-2) proposes a unified architecture to discretize and conduct auto-regressive modeling for image, text, and audio simultaneously, which, however, suffers from inevitable information loss brought by discretization, especially for the high-resolution visual inputs. In this work, we propose EMOVA, the very first unified Omni-modal Large Language Models with state-of-the-art vision-language and speech performance at the same time.

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3 ARCHITECTURE

157 158 3.1 FORMULATION

159 160 161 Denote the Large Language Model (LLM) as $f(\cdot)$ and the text, visual and speech inputs as X_T, X_V and \mathbf{X}_S , respectively. \mathbf{X}_T is converted to discrete tokens \mathbf{U}_T via a text tokenizer [\(Gage,](#page-11-5) [1994\)](#page-11-5), while the visual input X_V is first encoded with a vision encoder $v(\cdot)$ as $\mathbf{E}_V = v(\mathbf{X}_V)$, and then projected into the text embedding space with a projector $p(\cdot)$ as $\mathbf{H}_V = p(\mathbf{E}_V)$. As for the speech input \mathbf{X}_S , a

162 163 164 165 *Speech-to-Unit* (S2U) procedure is required. Specifically, \mathbf{X}_S first passes through a speech encoder $s(\cdot)$ as $\mathbf{E}_S = s(\mathbf{X}_S)$, which is then discretized by the quantizer $q(\cdot)$ as $\mathbf{U}_S = q(\mathbf{E}_S)$. The LLM $f(\cdot)$ is then trained to compute the joint probability of the output text and speech units $\dot{\mathbf{U}}_T^o$, \mathbf{U}_S^o as

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\mathbb{P}(\mathbf{U}_T^o, \mathbf{U}_S^o | \mathbf{U}_T, \mathbf{U}_S, \mathbf{H}_V) = \prod_{i=1}^L \mathbb{P}(\mathbf{x}_i | \mathbf{U}_{T, < i}^o, \mathbf{U}_{S, < i}^o, \mathbf{U}_T, \mathbf{U}_S, \mathbf{H}_V),\tag{1}
$$

168 169 170 where $x_i \in U_T^o \cup U_S^o$ and $L = |U_T^o| + |U_S^o|$. The output response units U_S^o are then recovered into the output speech waveform Y_S^o via a *Unit-to-Speech* (U2S) decoder $d(\cdot, \cdot)$ together with an emotion style embedding \mathbf{E}_{style}^o to realize vivid emotional spoken dialogue controllability (Sec. [3.2\)](#page-3-0).

172 173 174 LLM. We adopt the LLaMA-3.1-8B [\(Dubey et al.,](#page-11-6) [2024\)](#page-11-6) as our base LLM $f(\cdot)$, due to its superior performance among publicly available checkpoints, which is equipped with a tiktoken text tokenizer and a vocabulary size of 128,256, supporting both multilingual textual inputs and outputs.

175 176 177 178 179 180 Vision encoder and projector. We utilize InternViT-6B [\(Chen et al.,](#page-10-2) [2024b\)](#page-10-2) as our visual encoder $v(\cdot)$ with 448 × 448 base resolution, and C-Abstractor [\(Cha et al.,](#page-10-7) [2024\)](#page-10-7) with two ResBlocks (both before and after the pooling layer) and $4\times$ downsample rate as vision projector $p(\cdot)$. To process the high-resolution inputs, the high-resolution image-slicing [\(Liu et al.,](#page-12-4) [2024a\)](#page-12-4) is used, where visual tokens for one image are concatenation with a low-resolution thumbnail and the origin high-resolution image with separators in each line, allowing a maximum of nine tiles during training.

182 3.2 SPEECH TOKENIZATION

183 184 185 186 187 188 189 Speech-to-unit (S2U) tokenizer. Following [Tao et al.](#page-14-3) [\(2024\)](#page-14-3), we adopt the SPIRAL [\(Huang et al.,](#page-11-7) [2022\)](#page-11-7) architecture for the speech encoder $s(\cdot)$ to capture both phonetic and tonal information, which is then discretized by the quantizer $q(\cdot)$ utilizing the finite scalar quantization (FSQ) [\(Mentzer et al.,](#page-13-3) [2023\)](#page-13-3). The size of the speech codebook is 4,096, while the sample rate is 25 tokens per second. Once discretized, the speech modality can be simply integrated into LLMs by concatenating the text vocabulary and speech codebook.

190 191 192 193 194 Our S2U tokenizer provides the following advantages: 1) *Data efficiency*: after pre-training on largescale unlabeled speech data, it requires only a small amount of speech-text pair data for fine-tuning. 2) *Bilingual*: the speech codebook is shared among different languages (*i.e*., English and Chinese), sharing unit modeling abilities across languages. Check more training details and comparisons with other speech tokenizers [\(Zhang et al.,](#page-15-1) [2023b\)](#page-15-1) in Appendix [A.1.](#page-16-0)

195 196 197 198 199 200 201 Semantic-acoustic disentanglement. To seamlessly align speech units with the highly semantic embedding space of LLMs, we opt for decoupling the semantic contents and acoustic styles of input speeches. Specifically, given input speechs \mathbf{X}_{S} , both semantic embedding $\mathbf{E}_{semantic}$ and style embeddings \mathbf{E}_{style} are extracted separately, while only the $\mathbf{E}_{semantic}$ is quantified by $q(\cdot)$ to generate speech units U_S . By changing E_{style} while maintaining the same $E_{semantic}$, we can easily control speech styles without disturbing the semantic contents of recovered speeches. Moreover, the disentanglement facilitates modality alignment among speeches and texts, as later shown in Sec. [4.1.](#page-4-0)

202 203 204 205 206 207 208 209 Unit-to-speech (U2S) detokenizer with style control. Building on VITS [\(Kim et al.,](#page-12-6) [2021\)](#page-12-6), our U2S detokenizer adopts a conditional VAE architecture (see Fig. [7\)](#page-17-0). To achieve vivid style controls, we utilize the semantic-style disentanglement of our S2U tokenizer (as discussed above) and adopt a novel style embedding to control the speech styles (*e.g*., speaker identities, emotions, and pitches). Specifically, the LLM $f(\cdot)$ is trained to generate both the output speech units \mathbf{U}_S^o and a style label. The speech units \mathbf{U}_S^o are converted to unit embeddings $\mathbf{E}_{semantic}^o$, while the style label is utilized to generate a unique style prototype \mathbf{E}_{style}^o . Both $\mathbf{E}_{semantic}^o$ and \mathbf{E}_{style}^o are taken as inputs to speech decoder $d(\cdot, \cdot)$ to synthesize the output speech waveform Y_S^o . See Appendix [A.2](#page-17-1) for more details.

210 211 212 213 214 215 Our U2S detokenizer is pre-trained on LibriTTS [\(Zen et al.,](#page-14-8) [2019\)](#page-14-8) and AISHELL-1 [\(Bu et al.,](#page-10-8) [2017\)](#page-10-8) and subsequently fine-tuned on synthetic style-rich speech data. Specifically, due to the scarcity of real-life style-rich data, we utilize TTS tools [\(Du et al.,](#page-11-8) [2024\)](#page-11-8) to synthesize speech samples diverse in genders, pitches, and emotions. As for style prototypes, Emotion2Vec [\(Ma et al.,](#page-13-4) [2023\)](#page-13-4) is adopted to select the most representative samples with the highest confidence in conveying the desired style. Our empirical results reveal that even one representative style reference speech has been sufficient

to control the speech styles flexibly and precisely.

Figure 2: Comparison between omni-modal alignment paradigms. 1) Joint training achieves consistent improvements over VL and Speech, suggesting omni-modal alignment can be beneficial across modalities. 2) Joint training outperforms both VL→Speech and Speech→VL, revealing that joint training is more superior and efficient than sequential training. 3) Joint is superior to Joint-entangled, highlighting the effectiveness of the semantic-acoustic disentanglement.

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4 OMNI-MODAL ALIGNMENT AND INSTRUCTION TUNING

234 235 236 237 238 239 240 241 To achieve the omni-model alignment, it is ideal to utilize large-scale omni-modal image-text-speech data, which, however, is either without reach due to copyright [\(Nagrani et al.,](#page-13-5) [2022\)](#page-13-5) or limited in the quality [\(Miech et al.,](#page-13-6) [2019\)](#page-13-6). An alternative is to use existing image-text data with TTS-synthesized speeches, which is not only computationally expensive but also hampers data diversity, as most TTS tools generate speech in similar patterns. Recent works [\(Chen et al.,](#page-10-2) [2024b;](#page-10-2) [Fu et al.,](#page-11-0) [2024b\)](#page-11-0) choose to integrate the speech modality into a well-structured VLLM via a sequential training manner with bi-modal alignment datasets. However, the relationships between different modalities and how to effectively leverage multiple bi-modal alignment datasets remain an open question.

242 243 244 245 246 In this work, we explore omni-modal text-centric alignment by utilizing publicly available bi-modal alignment datasets, including both image-text (*e.g*., captioning) and speech-text (*e.g*., ASR and TTS) datasets. With the text modality as a bridge, EMOVA ultimately becomes a unified system capable of understanding and generating multiple modalities in a coherent and integrated manner. Specifically, in Sec. [4.1,](#page-4-0) we explore the following three questions:

- 1. *Does the integration of the speech modality conflict with the vision-language capabilities?*
- 2. *How to represent speech modality to foster omni-modal alignment?*
- 3. *Is sequential alignment of multiple modalities optimal?*

Then we introduce the omni-modal instruction tuning pipeline and the overall training paradigm of our EMOVA in Sec. [4.2](#page-5-0) and Sec. [4.3,](#page-7-0) respectively.

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4.1 OMNI-MODAL TEXT-CENTRIC ALIGNMENT

255 256 257 258 259 260 261 262 263 264 265 266 Settings. To answer the questions above, we experimentally compare the following omni-modal training paradigms: 1) VL→Speech conducts image-text alignment first followed by speech-unittext alignment using the full speech data and 10% of image-text alignment data to avoid catastrophic forgetting, similar to InternOmni [\(Chen et al.,](#page-10-2) [2024b\)](#page-11-0) and VITA [\(Fu et al.,](#page-11-0) 2024b). 2) Speech \rightarrow VL instead performs speech-unit-text alignment first and then aligns images with texts using 10% of the speech unit-text data and the full image-text data. 3) Joint aligns both modalities simultaneously. Note that unless otherwise specified, we use the S2U tokenizer introduced in Sec[.3.2](#page-3-0) to extract speech units for all speech data, which effectively disentangles semantic and acoustic features. Joint-entangled derives speech units using HuBERT [\(Hsu et al.,](#page-11-9) [2021\)](#page-11-9), which does not achieve semantic-acoustic disentanglement effectively with only Kmeans clustering. 4) VL and Speech only align the vision and speech modalities with texts, respectively, serving as baselines (see Appendix [B.1](#page-18-0) for more details).

267 268 269 Evaluation. For speech capabilities, we directly evaluate the aligned model's performance on the ASR task of LibriSpeech [\(Panayotov et al.,](#page-13-7) [2015\)](#page-13-7), while for vision-language, we fine-tune the model using a small amount of high-quality visual instruction data (*i.e*., the 665K SFT data from ShareGPT4V [\(Chen et al.,](#page-10-9) [2023d\)](#page-10-9)) and evaluate the fine-tuned model on common vision-language

Figure 3: Demonstration of the omni-modal instruction tuning. 1) To empower emotional spoken dialogues, EMOVA is trained to explicitly select the speech style labels (*e.g*., emotions and pitches) with output speech units. 2) For the ease of parsing, data components are arranged in JSON format.

benchmarks. [C](#page-18-1)heck Appendix C for evaluation details. Fig. [2](#page-4-1) shows the comparison among different paradigms on vision-language (left and middle) and ASR (right, where we report the 1 − WER value for better readability) benchmarks, from which we can derive the following observations:

297 298 299 300 301 302 303 304 305 306 Observation 1: image-text and speech-unit-text data benefit each other. Contrary to the common assumption that multiple modalities might compete and create conflicts, we find that introducing additional modalities is actually beneficial. As illustrated in Fig. [2,](#page-4-1) Joint consistently outperforms both VL and Speech across vision-language and speech benchmarks. Moreover, even models aligned sequentially, such as VL→Speech and Speech→VL, which are typically prone to catastrophic forgetting, demonstrate superior performance on most vision-language tasks. We speculate that the requirement to align multiple modalities with text leads to more robust representations, which in turn generalize better across different downstream tasks. This finding aligns with the results from ImageBind [\(Girdhar et al.,](#page-11-10) [2023\)](#page-11-10), where the joint alignment of audio and depth modalities with images resulted in improved downstream performance.

307 308 309 310 311 Observation 2: semantic-acoustic disentanglement benefits omni-modal alignment. We find 1) Joint outperforms Joint-entangled on vision-language benchmarks, and 2) in the speech tasks, Joint maintains significant advantages over its entangled counterpart. This can be attributed to the semantic-acoustic disentanglement which makes speech units more analogous to languages, a domain LLMs are particularly specialized.

312 313 314 Observation 3: sequential alignment is not optimal. We notice that $Joint$ consistently outperforms its sequential counterparts (*i.e*., VL→Speech and Speech→VL) on both vision-language and speech benchmarks, probably due to catastrophic forgetting when integrating a new modality.

315 316 317 318 In light of these observations, we have chosen to pursue the ultimate alignment strategy that simultaneously aligns image-text and speech-unit-text for EMOVA, which offers two important benefits, 1) it fosters mutual enhancement between vision-language and speech capabilities, and 2) it avoids the issue of catastrophic forgetting during sequential alignment of multiple modalities.

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- 4.2 OMNI-MODAL INSTRUCTION TUNING
- **322 323** After the omni-modal text-centric alignment, the model learns fundamental vision-language (*e.g*., captioning) and speech capabilities (*e.g*., ASR and TTS). However, instruction tuning is essential to better follow complicated user instructions and respond with vivid emotions.

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Figure 4: Overview of the data composition for EMOVA omni-modal instruction tuning. (Left) Distribution of instruction data across categories, with the outer circle representing overall categories and the inner circle depicting subset distributions. (Right) Quantitative breakdown of data sources.

Table 2: Detailed configuration for different training stages of EMOVA. The table illustrates the vision configurations, dataset characteristics, and training hyperparameters.

	Settings	Stage-1	Stage-2	Stage-3
Vision	Resolution # Tokens	$448\times \{\{1\times2\},\cdots,\{3\times3\}\}\$ Max $256 \times (1+9)$	$448 \times \{\{1\times2\}, \cdots, \{3\times3\}\}\$ Max $256 \times (1+9)$	$448\times \{\{1\times2\},\cdots,\{3\times3\}\}\$ Max $256 \times (1+9)$
Data	Dataset # Samples	LCS 558K	EMOVA-Alignment (Fig. 8) 7.4M	EMOVA-SFT (Fig. 4) 4.4M
Training	Trainable Batch Size	Projector 256	Full Model (Half ViT) 256	Full Model 128
	LR : $p(\cdot)$ LR: $v(\cdot)$	1×10^{-3}	2×10^{-5} 2×10^{-5}	2×10^{-5} 2×10^{-6}
	LR: $f(\cdot)$ Epoch		2×10^{-5}	2×10^{-5}

352 353 354 355 356 357 358 359 360 361 362 363 364 Emotion-enriched instruction data synthesis. Due to the scarcity of omni-modal instruction data (*i.e*., dialogues involving images, speeches, and texts simultaneously), we opt for synthesizing omnimodal instruction data from existing text and visual instruction datasets. First, we select instruction data suitable for the vocal expression by filtering out the non-vocal data (*e.g*., code and mathematical formulas). Second, we clean the selected data to be more vocal by removing text formatting elements $(e.g., **$ and \n\n). Then, we obtain style labels for the remaining dialog contexts, including genders (male, female), pitches (normal, low, high), and emotions (neutral, happy, sad, angry), resulting in 24 different speech styles. The style labels are obtained by prompting $GPT-4o¹$ $GPT-4o¹$ $GPT-4o¹$ to make reasonable inferences given the dialogue context. Finally, we convert the textual instructions and responses into speeches using the latest TTS tools (*i.e*., CosyVoice [\(Du et al.,](#page-11-8) [2024\)](#page-11-8) and Azure AI Speech), and the style labels are used to control the style of the synthesized speech data. To further improve the diversity of the data, each instruction is synthesized by randomly selecting one of the 39 available speakers. Ultimately, we gather 120K speech-text and 110K speech-image data pairs in total. More details can be found in Appendix [B.2.](#page-18-3)

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367 368 369 370 371 372 373 374 375 Data organization and chain of modality. The omni-modal instruction data can be represented as $D_{\text{omni}} = \{(x_V, u_S, x_T^o, c_{\text{style}}^o, u_S^o)\}_{i=1}^N$, where the input consists of the optional queried image x_V and the speech units of the instruction u_S , while the output consists of the textual response x_T^o , the predicted speech style labels c_{style}^o , and the output speech unit u_S^o . Note that we train **EMOVA** to explicitly select styles (*e.g*., emotions and pitches), which are utilized to determine the corresponding style embedding for the U2S detokenizer (Sec. [3.2\)](#page-3-0). Moreover, since directly generating the speech responses is challenging, we decompose the speech response procedure into three primary steps: 1) recognizing user instructions into texts; 2) generating textual responses based on the recognized instructions; 3) generating the style labels and response speech units based on the textual responses. For ease of parsing during deployment, the target outputs are formatted as JSON, as shown in Fig. [3.](#page-5-1)

¹ https://chatgpt.ust.hk

4.3 OVERALL TRAINING PARADIGM

Inspired by [Chen et al.](#page-10-9) [\(2023d\)](#page-10-9), a three-stage training paradigm is adopted for EMOVA,

- Stage-1: Vision-language pre-alignment. The purpose is to align visual features into the embedding space of LLMs. Only the vision projector $p(\cdot)$ is trained.
- Stage-2: Omni-modal text-centric alignment. This stage jointly performs the visionlanguage and speech-language alignment simultaneously. We train the LLM $f(\cdot)$, vision projector $p(\cdot)$, and the deeper half of vision encoder $v(\cdot)$ layers.
- Stage-3: Omni-modal instruction tuning. To empower EMOVA to respond accurately to omni-modal instructions, we organize different datasets with various types of instructions to enforce EMOVA to learn generalization across tasks, as detailed in Sec. [5.1.](#page-7-1)

411 5 EXPERIMENTS

412 413 5.1 TRAINING CONFIGURATION

414 415 416 Stage-1. In this stage, we only train the parameters of the vision projector $p(\cdot)$ for vision-language pre-alignment with the LCS-558K dataset [\(Liu et al.,](#page-12-1) [2024b\)](#page-12-1), with the high-resolution image-slicing strategy [\(Liu et al.,](#page-12-4) [2024a\)](#page-12-4) adopted.

417 418 419 420 421 422 423 424 425 Stage-2. We assemble a unified dataset with 7.4M samples for both the image-text and speech-text alignment, as summarized in Fig. [8.](#page-18-2) Specifically, we utilize pre-training datasets from ShareGPT4V [\(Chen et al.,](#page-10-9) [2023d\)](#page-10-9), ALLaVA [\(Chen et al.,](#page-10-10) [2024a\)](#page-10-10) (both the original English version and the Chinese version translated on our own), and ShareGPT-4o [\(Cui et al.,](#page-10-11) [2023\)](#page-10-11) for general perception, while for the OCR capabilities, we leverage SynthDog [\(Kim et al.,](#page-11-11) [2022\)](#page-11-11), MMC-Alignment [\(Liu et al.,](#page-12-7) [2023a\)](#page-12-7), K12 Printing, and UReader Text Reading subset [\(Ye et al.,](#page-14-9) [2023\)](#page-14-9). Moreover, we use the 2,000 hours of ASR and TTS data from LibriSpeech [\(Panayotov et al.,](#page-13-7) [2015\)](#page-13-7) and AISHELL-2 [\(Du et al.,](#page-10-12) [2018\)](#page-10-12) for speech-text alignment, and to preserve the language capabilities of LLMs, we further incorporate the text-only data from Magpie Pro [\(Xu et al.,](#page-14-10) [2024\)](#page-14-10).

426 427 428 429 430 431 Stage-3. We collect the EMOVA-SFT dataset consisting of 4.4M multi-task omni-modal samples (see Fig. [4\)](#page-6-0). We start by gathering high-quality open-sourced visual instruction datasets, including ShareGPT4V [\(Chen et al.,](#page-10-9) [2023d\)](#page-10-9), InternVL [\(Chen et al.,](#page-10-2) [2024b\)](#page-10-2), Meteor [\(Lee et al.,](#page-12-8) [2024\)](#page-12-8), Idefics-2 (Laurençon et al., [2024\)](#page-14-5), Cambrian [\(Tong et al.,](#page-14-5) 2024), and LLaVA-Onevision [\(Li et al.,](#page-12-0) [2024a\)](#page-12-0), followed by quality checking, re-formatting all data samples to a unified template, and removing the duplicates. For speech, we include the training split of EMOVA omni-model instruction data (*c.f* ., Sec. [4.2\)](#page-5-0), with 10% of speech alignment datasets to maintain ASR and TTS performance. We adopt 128 Ascend 910 Training NPUs in parallel for model training (check more details in Table [2\)](#page-6-2).

Figure 5: EMOVA engages in emotional spoken dialogue expressing sadness.

feel the frost, to feelthe snow, To find the beauty in its woe.

451 5.2 COMPARISON TO THE SOTA MODELS

452 453 454 455 456 8B [\(Chen et al.,](#page-10-2) [2024b\)](#page-10-2), together with the Speech LLM Mini-Omni [\(Xie & Wu,](#page-14-0) [2024\)](#page-14-0) and omni-Experimental results are provided in Table [3.](#page-7-2) We compare a wide range of state-of-the-art VLLMs, including proprietary ones like Gemini Pro 1.5 [\(Reid et al.,](#page-14-11) [2024\)](#page-14-11), GPT-4V [\(OpenAI,](#page-13-8) [2023\)](#page-13-8), GPT-4o [\(OpenAI,](#page-13-0) [2024\)](#page-13-0), and open-sourced ones like LLaVA-OneVision-7B [\(Li et al.,](#page-12-0) [2024a\)](#page-12-0), InternVL2 modal LLM AnyGPT [\(Zhan et al.,](#page-14-2) [2024\)](#page-14-2) and VITA-8x7B [\(Fu et al.,](#page-11-0) [2024b\)](#page-11-0).

457 458 459 460 461 462 Comparison with SOTA VLLMs. As an omni-modal model, EMOVA achieves comparable performance with state-of-the-art open-sourced VLLMs across multiple benchmarks. Specifically, our EMOVA outperforms both InternVL2 and LLaVA-OV on MMBench, SEED-Image, TextVQA, and OCRBench consistently, while demonstrating exceptional proficiency in solving math problems requiring precise visual content interpretation. **EMOVA** achieves a 2.8% improvement on MathVista compared with InternVL2 and 1.6% higher score on Mathverse compared to LLaVA-OneVision.

463 464 465 466 Furthermore, **EMOVA** shows competitive performance even compared with the proprietary models. EMOVA exceeds both GPT-4V and Gemini Pro 1.5 significantly on 10 out of 14 benchmarks, while for GPT-4o, EMOVA outperforms on both SEEDBench-Image and OCRBench, reaching over 95% of GPT-4o's performance on ALL evaluated benchmarks except RealWorldQA.

468 469 470 471 472 473 474 475 Comparison with SOTA omni-modal LLMs. Compared with VITA-8x7B, EMOVA shows substantial improvement on visual-language benchmarks. Specifically, EMOVA is 112 points higher than VITA on MME, and surpasses VITA by 21.5% on OCRBench, underscoring the effectiveness of our approach and the potential to push boundaries of omni-modal abilities. What's more, EMOVA significantly outperforms the most recent omni-modal model VITA, even surpassing its SLLM counterpart Mini-Omni, showing the effectiveness of the semantic-acoustic disentanglement and omni-modal mutual benefits. Qualitative results are shown in Fig. 5 and Appendix G . We also report TTS results in Table [6.](#page-20-0) For the first time, our EMOVA obtains state-of-the-art performance on both the vision-language and speech benchmarks simultaneously.

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5.3 EVALUATION OF EMOTION-RICH SPOKEN DIALOGUE

479 480 481 482 In this section, we evaluate the end-to-end spoken dialogue capabilities of EMOVA. As discussed in Sec. [4.2,](#page-5-0) the model takes an input image x_V and user instructions in the form of speech units u_S , and outputs *text responses*, *style labels*, and *corresponding speech units*. To ensure comprehensive evaluation, we propose the following evaluation metrics (see Appendix D for more details):

- **483** 1. End-to-end spoken dialogue score assesses the model's dialogue performance based on
- **484 485** the generated speeches, with a score ranging from 0 to 10, reporting the average.
	- 2. Unit-input-text-output score focuses on the quality of the text responses of LLM when the inputs are speech units, bypassing errors from speech synthesis.
- 3. Text-input-text-output score inputs the ground-truth text of the user instruction and evaluate the model's text output. This helps disentangle the impact of speech recognition errors and eliminates the influence of the JSON format.
	- 4. ASR and TTS evaluate how accurately EMOVA recognizes speech units and how effectively it generates speech units from text.
	- 5. Style label classification accuracy evaluates the accuracy of the model in selecting the appropriate speech style labels (Sec. [3.2\)](#page-3-0).
	- 6. Style controllablity assesses the style controllability of U2S detokenizer with the given conditional style labels via the confusion matrix between generated and recognized styles.

495 496 497 498 Due to the lack of emotionally rich spoken dialogue evaluation datasets, we split a test set from our synthesized omni-modal instruction-tuning data (see Sec. [4.1\)](#page-4-0). GPT-4o are used for automated evaluation. Details are provided in Appendix [D.](#page-19-0)

499 Results are shown in Table [4.](#page-8-1) As can be seen,

500 501 502 503 504 505 506 507 508 509 510 511 512 513 (i) By comparing the *end-to-end dialogue score* with the *unit-input-text-output score*, we notice that the two scores are closely aligned, with a maximum gap of only 0.22, except for Speech-Image-ZH. The TTS-WER/CER is generally low, revealing that EMOVA can synthesize accurate speech based on text responses. However, the Speech-Image-ZH is an outlier, which we attribute to its complexity. It includes tasks such as generating poetries and answering riddles, resulting in more intricate responses. When these answers are converted to speeches and then transcribed back to texts, multiple variations often arise, leading to discrepancies from the original responses.

Figure 6: Confusion matrix between the generated and recognized emotions.

514 515 516 517 518 519 520 521 (ii) Comparing the *unit-input-text-output* score with the *text-input-text-output* score, we observe that their differences correlate with the ASR results of speech instructions. Specifically, for Speech-Text-EN and Speech-Text-ZH, which involve more complex instructions, EMOVA reports inferior ASR performance (6.8 and 12.6, respectively) compared to other datasets (2.45 and 1.85). Consequently, when we replace speech instructions with ground-truth transcriptions, EMOVA shows significant improvements from *unit-input* to *text input* score. On the contrary, for datasets with accurate ASR performance, the results are quite similar, suggesting EMOVA retains robust dialogue capabilities when using the JSON format.

522 523 524 525 526 527 (iii) Examining the *classification accuracy of style labels*, we find that EMOVA performs satisfactorily in classifying emotions and pitch during speech conversations, achieving an accuracy of over 80%. The confusion matrix comparing the conditional and recognized emotion labels is shown in Fig. [6.](#page-9-0) The results indicate that the four emotions are recognized with high probabilities, with three achieving over 80% accuracy. This demonstrates that our U2S detokenizer effectively controls common emotions, endowing the synthesized speech with vivid emotional expression.

528 529 6 CONCLUSION

530 531 532 533 534 535 536 537 538 539 This work presents EMOVA, an innovative end-to-end omni-modal large language model that effectively aligns vision, speech, and text simultaneously. We employ a continuous vision encoder to capture fine-grained visual details, while a discrete, semantic-acoustic disentangled speech tokenizer and detokenizer enable end-to-end speech understanding and generation. A lightweight style module further supports spoken dialogue with vivid emotions. By using text as a bridge, we demonstrate that omni-modal alignment is achievable without relying on scarce omni-modal image-text-speech data, which not only enhances both vision-language and speech capabilities but also surpasses its bi-modal counterparts through joint optimization. For the first time, **EMOVA** achieves state-of-theart performance on both vision-language and speech benchmarks, setting a novel standard for the omni-modal models for versatile and expressive omni-modal interactions.

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APPENDIX

A MORE ON SPEECH TOKENIZER

A.1 SPEECH-TO-UNIT (S2U) TOKENIZER

871 872 873 874 Overview. To process the speech input X_S , our S2U tokenizer consists of a speech encoder $s(\cdot)$ with a quantization module $q(.)$. First, the speech input is passed through $s(.)$, producing a continuous latent representation $\mathbf{E}_S = s(\mathbf{X}_S)$. Then, the quantization module $q(\cdot)$ converts \mathbf{E}_s into discrete units $U_S = q(E_s)$. The final output is an ID sequence $U_S = [u_1, u_2, \dots]$, where each u_i corresponds to a unique speech unit.

876 877 878 879 880 881 882 After this S2U extraction process, the speech is represented by quantized IDs instead of quantized embeddings. For example, a speech signal is represented as an ID sequence like $[782, 463, 550, \dots]$, which can be treated as a special form of text. As a result, to integrate speech tokens into LLM $f(\cdot)$, we only need to expand the LLM's original vocabulary V_T by adding a set of speech unit tokens V_S , similar to [Zhang et al.](#page-15-0) [\(2023a\)](#page-15-0). The expanded vocabulary is thus the union $V = V_T \cup V_S$. In this work, the same codebook is shared across multiple languages, such as English and Chinese, enabling the unit modeling abilities to be shared across languages. This design enables simply integration of the speech modality into the LLM with minimal data requirements (see experiments in Sec. [4.1\)](#page-4-0).

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885 886 887 888 889 890 891 892 893 894 895 896 897 898 Training of S2U. The training of the S2U encoder involves three primary components: the speech encoder, the quantization module, and the phone decoder. First, the speech encoder is trained with a large amount of unlabeled speech with contrastive loss in a self-supervised learning manner [\(Tao](#page-14-3) [et al.,](#page-14-3) [2024\)](#page-14-3). The dataset utilized is the 10000-hour English speeches from GigaSpeech [\(Chen et al.,](#page-10-13) [2021a\)](#page-10-13) and the 10000-hour Chinese speeches from WenetSpeech [\(Zhang et al.,](#page-14-12) [2022\)](#page-14-12), both of which encode large variations in speakers, styles, and acoustic environments. Subsequently, the speech encoder, along with the quantization module and phone decoder, are optimized on a small amount of speech-text pair data, which is derived from the train-clean-100 subset of LibriSpeech [\(Panayotov](#page-13-7) [et al.,](#page-13-7) [2015\)](#page-13-7) in English and AISHELL-1 [\(Bu et al.,](#page-10-8) [2017\)](#page-10-8) in Chinese. The phone label is obtained using the phone set in Charsiu [\(Zhu et al.,](#page-15-2) [2022\)](#page-15-2). During training, the speech encoder encodes input speeches into continuous latent representations that are rich in both phonetic and tonal information. Then, the quantization module is used to convert the continuous outputs from the speech encoder into discrete units. Finally, the phone decoder converts the quantized embeddings into a sequence of non-tonal/tonal phones, ensuring that the speech units capture necessary information related to semantic contents in both non-tonal and tonal languages. After training, only the speech encoder and the quantization module are used in EMOVA.

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901 902 Comparisons with SpeechTokenizer in AnyGPT. Our S2U tokenizer differs from the Speech-Tokenizer [\(Zhang et al.,](#page-15-1) [2023b\)](#page-15-1) used in AnyGPT [\(Zhan et al.,](#page-14-2) [2024\)](#page-14-2), in the following aspects:

903 904 905 906 (1) SpeechTokenizer encodes both semantic contents and acoustic details of speeches, while our S2U tokenizer focuses solely on semantic contents. This design reduces the LLMs' burden of disentangling different aspects of speech information, facilitating the modality alignment between speech and text modalities during LLM training.

907 908 909 910 911 912 913 914 (2) Compared with SpeechTokenizer, our S2U tokenizer offers a more concise representation and helps to simplify and accelerate the generation of **EMOVA**. SpeechTokenizer employs tokens from eight RVQ layers with a 50Hz frame rate to represent speech, thus a 10-second speech corresponds to $500 \times 8 = 4000$ tokens. However, we reduce the frame rate from 50Hz to 25Hz and utilize only one token to represent each frame, and thus, a 10-second speech can be represented by only 250 tokens. Moreover, AnyGPT requires a two-stage generation process, involving autoregressive (AR) semantic token generation followed by the non-autoregressive (NAR) acoustic token generation. Instead, we only need to generate speech units capturing semantic contents in a fully AR manner.

915 916 917 (3) SpeechTokenizer lacks an explicit structure design to deal with tonal languages like Chinese, therefore, the processing ability in Chinese is not demonstrated in either SpeechTokenizer or AnyGPT. In contrast, our S2U tokenizer incorporates training constraints to better capture tone variation in phone, making it effective for both the non-tonal and tonal languages. This further en**918 919 920** hances **EMOVA**'s multilingual speech processing capabilities, enabling it to effectively handle both English and Chinese.

922 In summary, our S2U tokenizer improves the compactness and generality of speech representation, facilitates LLM training, and enhances its multilingual speech ability. Experimental results show that our model significantly outperforms AnyGPT in ASR tasks, as shown in Table [6.](#page-20-0)

A.2 UNIT-TO-SPEECH (U2S) DETOKENIZER WITH STYLE CONTROL

928 929 930 931 932 933 934 935 936 Overview. The LLM, along with the vision encoder and speech tokenizer, is trained endto-end to generate responses in the form of the speech units, given the input images and speeches. Specifically, the output speech units can be obtained via $\mathbf{U}_S^o = f(\mathbf{U}_T, \mathbf{U}_S, \mathbf{H}_V)$, followed by a U2S detokenizer to convert the discrete speech units \mathbf{U}_{S}° into the final output speech waveforms.

937 938 939 940 941 942 943 944 945 946 947 948 The proposed U2S detokenizer involves three core modules: the speech unit encoder $e(\cdot)$, the speech style encoder $q(\cdot)$, and the speech decoder $d(\cdot, \cdot)$. First, the speech unit encoder converts the speech units U_S^o into unit embeddings \mathbf{E}_{unit}^o . Meanwhile, the style encoder $g(\cdot)$, adopting the structure of Meta-StyleSpeech [\(Min et al.,](#page-13-9) [2021\)](#page-13-9), is utilized to extract a style embedding \mathbf{E}^o_{style} from the chosen reference speech. Lastly, the speech decoder $d(\cdot, \cdot)$ reconstructs the speech waveform Y_S^o from the unit embedding \mathbf{E}_{unit}^o and style embedding \mathbf{E}^o_{style} .

Figure 7: U2S detokenizer with style control.

Training of U2S. Training a U2S detokenizer with emotion controls is challenging considering the lack of labeled emotional speech data since most open-source speech data is predominantly neutral in emotion or lacks emotion labels. Due to the limited availability of emotion-rich data, we utilize TTS tools [\(Du et al.,](#page-11-8) [2024\)](#page-11-8) to generate a small set of style-rich speech samples diverse in speaker identities, genders, emotions, and pitches. Our U2S detokenizer is first pre-trained on LibriTTS [\(Zen](#page-14-8) [et al.,](#page-14-8) [2019\)](#page-14-8) and AISHELL-1 [\(Bu et al.,](#page-10-8) [2017\)](#page-10-8) to acquire fundamental speech synthesis capabilities, and subsequently, the synthesized style-rich speech data is utilized to fine-tune the U2S detokenizer, enhancing its controllability over diverse speech styles.

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961 962 963 964 965 966 967 968 969 970 971 Style Prototypes. To better facilitate controls of genders, emotions, and pitches, inspired by [Min](#page-13-9) [et al.](#page-13-9) [\(2021\)](#page-13-9) that a small number of style reference speeches can effectively transfer the target styles, we adopt a "store-for-usage" manner, *i.e*., we construct a style prototype codebook in advance for speech style assignation. Specifically, we synthesize K reference candidates with external TTS tools for each possible combination of the following styles: two genders (male, female), seven emotions (neutral, happy, sad, angry), and three pitches (normal, high, low), leading to 24 unique styles and $24 \times K$ candidates. Empirically we find that genders and pitches are easy to control using any of the candidate references, while the emotion intensity varies across speeches. To tackle this, we adopt Emotion2Vec [\(Ma et al.,](#page-13-4) [2023\)](#page-13-4), a powerful speech emotion recognition (SER) tool, to measure the emotion intensity of each candidate reference, and rank them in terms of the confidence of the desired emotion. We select the Top-1 candidate reference in each combination style to be the prototype of this condition. Finally, the most representative 24 reference speeches are selected from the $24 \times K$ candidates.

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C MORE ON BENCHMARK EVALUATION

1023 1024 1025 To thoroughly evaluate our model's vision-language abilities, 14 benchmarks covering four different aspects of real-life scenarios are utilized for a comprehensive assessment across multiple domains. Moreover, Automatic Speech Recognition (ASR) and Text-to-speech (TTS) are adopted to evaluate speech-language abilities.

Table 5: Statistics of the EMOVA speech instruction tuning datasets.

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1045 1046 1047 1048 1049 1050 Document/chart understanding and OCR abilities. Benchmarks including the TextVQA [\(Singh](#page-14-13) [et al.,](#page-14-13) [2019\)](#page-14-13), ChartQA [\(Masry et al.,](#page-13-10) [2022\)](#page-13-10), DocVQA [\(Mathew et al.,](#page-13-11) [2021\)](#page-13-11), InfoVQA [\(Mathew](#page-13-12) [et al.,](#page-13-12) [2022\)](#page-13-12), and OCRBench [\(Liu et al.,](#page-12-9) [2023c\)](#page-12-9), focus on recognition and understanding of structured data (*e.g*., charts, documents, and characters), challenging the model to extract, comprehend, and reason with structural and textual data. We adopt the corresponding test set for evaluation.

1051 1052 1053 1054 General image perception and understanding. MME [\(Fu et al.,](#page-11-12) [2024a\)](#page-11-12), MMBench [\(Liu et al.,](#page-12-10) [2023b\)](#page-12-10), SEED-Bench-Image [\(Li et al.,](#page-12-11) [2023a\)](#page-12-11), MM-Vet [\(Yu et al.,](#page-14-14) [2024\)](#page-14-14), and RealWorldQA [\(xAI,](#page-14-15) [2024\)](#page-14-15) evaluate models on perception and reasoning among general vision domains, providing a comprehensive measurement of models' real-world generalization abilities.

1055 1056 1057 Mathematical problem solving. MathVista (testmini) [\(Lu et al.,](#page-13-13) [2024\)](#page-13-13) and MathVerse (visionmini) [\(Zhang et al.,](#page-15-3) [2024\)](#page-15-3) evaluate the model's ability to deal with diverse mathematical problems, including both arithmetic and complex reasoning questions across different levels of complexity.

1058 1059 1060 1061 Science understanding. ScienceQA-Img [\(Lu et al.,](#page-13-14) [2022\)](#page-13-14) and AI2D [\(Kembhavi et al.,](#page-11-13) [2016\)](#page-11-13) are utilized to assess the model's ability to deal with scientific questions and diagrams, which requires reasoning across various subjects and interpreting structured visual content.

1062 1063 1064 1065 1066 Automatic speech recognition (ASR). We utilize the test-clean set of LibriSpeech [\(Panayotov](#page-13-7) [et al.,](#page-13-7) [2015\)](#page-13-7) for English, reporting the Word Error Rate (WER) as the evaluation metric. For Chinese, evaluation is conducted on the test set of AISHELL-2 [\(Du et al.,](#page-10-12) [2018\)](#page-10-12), using the Character Error Rate (CER). Both WER and CER assess ASR performance, calculated by comparing the recognized texts with the ground-truth transcripts.

1067 1068 1069 1070 1071 1072 Text-to-speech (TTS). To evaluate the TTS abilities, we first prompt **EMOVA** to generate speech units, which are then converted to speech waveforms by the U2S detokenizer. Using the synthesized speech as input, we conduct ASR with Whisper-large-v3 and Paraformer-zh for English and Chinese, respectively, to obtain transcribed texts. We then compute the WER and CER between the ground truth texts and the transcribed texts as metrics for TTS. The resulting metrics are denoted as TTS-WER and TTS-CER for English and Chines.

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1076 D.1 CALCULATION OF EVALUATION METRICS

1078 1079 End-to-end spoken dialogue score. We prompt GPT-40 with the original question x_T^{0} , the ground-truth text answer $x_T^{o^2}$ and the transcribed text from the generated speech, to obtain a score ranging from 0 to 10 and report an average of them. The prompt can be found in Fig. [24.](#page-32-0)

Comparison on the ASR and TTS benchmarks.

1088 1089 1090 1091 Unit-input-text-output score. Similar to end-to-end spoken dialogue score, but we use the predicted text response $x_T^{\tilde{O}^2}$ as answer instead of the transcribed text from the generated speech, to obtain a score ranging from 0 to 10 and report an average of them. See the prompt in Fig. [23.](#page-32-1)

1092 1093 Text-input-text-output score. The prompt can be found in Fig. [23.](#page-32-1)

1094 1095 1096 1097 Style label classification accuracy. We use GPT-4o to decide whether the style predictions c_{emo}^{σ} , \tilde{c}_{p}^{σ} are correct given the transcribed instruction x_{T}^{σ} and the predicted text response x_{T}^{σ} . The prompt can be found in Fig. [25.](#page-33-0)

1098 1099 1100 1101 1102 1103 1104 Emotion controllablity of our U2S detokenizer is assessed by providing texts to LLM to generate corresponding units (*i.e*., TTS), which, along with the given conditional emotion labels, are then fed into our U2S detokenizer to synthesize speech. We choose female voice due to its large variation of styles. We evaluate on 4 commonly-seen emotion, *i.e*.,"neutral", "happy", "sad", and "angry". We synthesize 200 speech utterances for testing, with 50 utterances per emotion. The output speeches are analyzed by a Speech Emotion Recognition (SER) model named Emotion2Vec [\(Ma et al.,](#page-13-4) [2023\)](#page-13-4), which identifies the emotion with the greatest likelihood among these four emotion.

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1106 1107 D.2 COMPARISON WITH OTHER OMNI MODELS

1108 1109 1110 1111 1112 Experimental results of ASR and TTS are reported in Table [3](#page-7-2) and [6.](#page-20-0) **EMOVA** achieves significant improvements over other omni-modal models (*i.e*., AnyGPT and VITA), even surpassing its SLLM counterpart Mini-Omni [\(Xie & Wu,](#page-14-0) [2024\)](#page-14-0), demonstrating the effectiveness of semantic-acoustic disentanglement and omni-modal mutual benefits. For the first time, our **EMOVA** obtains state-ofthe-art performance on both the vision-language and speech benchmarks simultaneously.

E MORE ON VISION-LANGUAGE

Table 7: Ablation on the ViT configurations and templates for vision-language alignment.

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1124 1125 1126 1127 1128 This section analyzes the pre-training configurations of the vision encoder and the prompt template during Stage 2, as shown in Table. [7.](#page-20-2) Our final selection is the colored setting. We find that training the ViT model with half of the deeper parameters [\(Chen et al.,](#page-10-9) [2023d\)](#page-10-9) with a learning rate of 2×10⁻⁶ [\(Liu et al.,](#page-12-4) [2024a\)](#page-12-4) yields the best performance. Furthermore, we compare the QA template with the plain template [\(Liu et al.,](#page-12-1) [2024b\)](#page-12-1) and find that the QA template is superior for pre-training.

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F LIMITATIONS

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1132 1133 Duplex modeling. In the current version, EMOVA can only process either visual/speech/text inputs or produce speech/text outputs at the same time. For a communication experience that mirrors human interaction, handling inputs and outputs simultaneously is crucial. Recent works like VITA

1134 1135 1136 1137 1138 1139 1140 [\(Fu et al.,](#page-11-0) [2024b\)](#page-11-0) and LSLM [\(Ma et al.,](#page-13-15) [2024\)](#page-13-15) have begun to explore duplex modeling. VITA focuses on recognizing speech in noisy environments during the generation process to facilitate timely responses. LSLM, on the other hand, attempts to halt speech production when it detects a com-mand or voice. Recently, a ground-breaking work named Moshi (Défossez et al., [2024\)](#page-10-14) develops a model supporting fully duplex modeling. The adeptness at simultaneously managing the information streams from both the user and the assistant allows Moshi to converse with human beings in real-time scenarios.

1141 1142 1143 1144 However, incorporating emotion into this duplex modeling presents additional challenges. Emotional states can fluctuate throughout a conversation, and how to generate appropriate responses given the user's previous and current emotional cues has not been thoroughly investigated. We will dive into this topic in the future work.

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1146 1147 1148 1149 1150 1151 1152 Direct unit-to-unit generation. Although speech units have served as speech representation, they are predominantly adopted in conjunction with text-based assistance [\(Zhang et al.,](#page-15-0) [2023a\)](#page-15-0). However, the direct generation from unit to unit without text assistance is an area that has not been extensively explored. In [Lee et al.](#page-12-12) [\(2021\)](#page-12-12), speeches from the source language are directly translated into speech units of the target language for speech-to-speech translation. Similarly, [Nguyen et al.](#page-13-16) [\(2023\)](#page-13-16) builds a language model directly on speech units, enabling spoken dialogue generation from raw audio. Both works develop models in speech-only data.

1153 1154 1155 1156 In the current version of **EMOVA**, the text modality is integrated into the speech generation process to transfer textual knowledge to the speech modality, thereby enhancing the correctness of speech responses. In the future, we will strengthen the model's direct unit-to-unit generation capabilities to boost the speed of speech generation and augment the model's comprehension of speech modality.

1157 1158 1159 1160 1161 1162 1163 Vision configurations. Currently, we only utilize a single vision encoder pre-trained via a visionlanguage manner, while recent works have shown effectiveness by combining vision encoders pretrained by different manners (*e.g*., self-supervised pre-training [\(Liu et al.,](#page-12-13) [2022\)](#page-12-13)) and architectures (*e.g*., MoE [\(Zhili et al.,](#page-15-4) [2023;](#page-15-4) [Liu et al.,](#page-13-17) [2024c\)](#page-13-17)). We prioritize visual understanding in this work, while the incorporation of (controllable) visual generation [\(Chen et al.,](#page-10-15) [2023c;](#page-10-15) [Gao et al.,](#page-11-14) [2023;](#page-11-14) [Li et al.,](#page-12-14) [2023b;](#page-12-14) [Wang et al.,](#page-14-16) [2024;](#page-14-16) [Liu et al.,](#page-13-18) [2023d;](#page-13-18) [Gao et al.,](#page-11-15) [2024\)](#page-11-15) is also appealing to better empower EMOVA in real-life applications [\(Li et al.,](#page-12-15) [2024c;](#page-12-15) [2022;](#page-12-16) [Han et al.,](#page-11-16) [2021\)](#page-11-16).

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G QUALITATIVE RESULTS

1167 1168 1169 1170 Advanced Vision-language Abilities. EMOVA exhibit advanced vision-language abilities in humor understanding (Fig. [9\)](#page-22-0), numerical calculations (Fig. [10\)](#page-22-1), coding (Fig. [11\)](#page-23-0), geometry problem solving $(Fig. 15)$ $(Fig. 15)$.

1171 1172 1173 1174 (Omni-modal) Emotional Spoken Dialogue. EMOVA can engage in omni-modal emotional spoken dialogue (Figs. [5,](#page-8-0) [12,](#page-23-1) [13,](#page-24-0) [17\)](#page-27-0). For demonstration purposes, we present only the speech transcriptions^{[2](#page-21-2)} of the instruction and EMOVA's response. The text highlighted in blue indicates the style labels predicted by EMOVA, reflecting the emotion and pitch of the generated speech.

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1176 1177 1178 1179 1180 1181 Omni-modal Spoken Dialogue with Structural Data Understanding. EMOVA can perform structural data understanding in spoken dialogue. The data types include PDF files (Fig. [14\)](#page-24-1), movie posters (Figs. 16 and 18), personal résumés (Fig. 19), charts (Fig. 20), and websites (Fig. 21). Remarkably, EMOVA is not specifically trained on spoken dialogue involving these data types. This suggests that our chain of modality data introduced in Sec. [4.2](#page-5-0) effectively activates both visionlanguage abilities and speech understanding acquired during the text-centric alignment stage.

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²For speech files, please refer to our project page.

	PORK BELLY Shelf life 30%! 9.80
User	First please perform reasoning, and think step by step to provide best answer to the following question: What is the original price for pork belly before discount?
EMOVA	To find the original price of pork belly before the discount, we need to use the information provided in the image. The image shows that the price after a 30% discount is \$9.80. This price represents 70% of the original price, since $100\% - 30\% = 70\%$. To find the original price, we can set up the equation:
	70% of original price $=$ \$9.80
	To solve for the original price, we can convert the percentage to a decimal and then divide:
	$0.70 *$ original price = \$9.80
	Now, divide both sides by 0.70 to isolate the original price:
	original price = $$9.80 / 0.70$ original price $= 14

Figure 10: EMOVA exhibits advanced vision-language abilities in numerical calculations.

Figure 12: EMOVA engages in spoken dialogue with a cheerful tone.

Figure 14: EMOVA engages in omni-modal spoken dialogue with structural data understanding.

Figure 16: EMOVA engages in omni-modal spoken dialogue with structural data understanding (*i.e*., poster).

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Figure 17: EMOVA engages in omni-modal emotional spoken dialogue.

Figure 18: EMOVA engages in omni-modal spoken dialogue with structural data understanding (*i.e*., movie posters).

 Figure 19: EMOVA engages in omni-modal spoken dialogue with structural data understanding (*i.e.*, résumé).

Figure 20: EMOVA engages in omni-modal spoken dialogue with structural data understanding (*i.e*., chart).

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For pitch, choose a single option from the following list: ['low', 'normal', 'high']

Please do not provide an option outside of the given list. Please output in the following JSON format:

{{ "user emotion":

"user pitch": ...,

 "assistant emotion": ..., "assistant pitch": ...

}}

Figure 22: Prompt used to obtain style labels of the speech instruction dataset.

Figure 25: Prompt used to obtain Classification Accuracy of Style Label.

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