S2ST-Omni: Hierarchical Language-Aware SpeechLLM Adaptation for Multilingual Speech-to-Speech Translation

Yu Pan[1†] **Xiongfei Wu**[2] **Yuguang Yang**[4] **Jixun Yao**[5]
Maxime Cordy[2] **Lei Ma**[3] **Jianjun Zhao**[1]
[1]Kyushu University, Japan [2]University of Luxembourg, Luxembourg
[3]The University of Tokyo, Japan [4]Tencent, China [5]ByteDance Ltd., China
panyu.ztj@gmail.com

Abstract

Despite recent advances in speech-to-speech translation (S2ST), it remains difficult to achieve both high translation accuracy and practical flexibility. In this paper, we present *S2ST-Omni*, a compositional S2ST framework that integrates a high-accuracy speech-to-text translation (S2TT) frontend with a modular, plug-and-play text-to-speech (TTS) backend, enabling independent optimization of translation and synthesis. On the S2TT side, we introduce a hybrid adapter that follows a "local-then-global" strategy to bridge the pretrained Whisper encoder and Qwen3 LLM, yielding a hierarchical acoustic-to-semantic abstraction. Building on this bridge, we further propose a hierarchical language-aware architecture that injects source-language information at two complementary levels. At the acoustic level, *Language-Aware Dual-CTC* operates on intermediate adapter features and employs FiLM-style feature modulation with a learnable gate, encouraging the model to learn language-specific but content-faithful acoustic representations. At the linguistic level, *Language-Aware Prompting* dynamically constructs source-language-conditioned prompts that activate language-specific translation knowledge in the LLM. To enable efficient optimization, we design a task-specific progressive fine-tuning strategy that first stabilizes speech-text alignment and then improves translation via LoRA on top of this converged foundation. The TTS backend remains fully modular and can be instantiated with any state-of-the-art synthesizer without retraining the S2TT frontend. Experiments on CVSS-C show that *S2ST-Omni* consistently achieves the best BLEU and ASR-BLEU across French, German, and Spanish to English directions, outperforming strong recent S2ST baselines.

1 Introduction

Multilingual speech-to-speech translation (S2ST) aims to directly convert utterances from multiple source languages into target-language speech, and plays an important role in cross-language communication, healthcare, and education (Kikui et al., 2003; Lee et al., 2022b).

Recent years have witnessed rapid progress in S2ST, with existing methods broadly falling into two categories: end-to-end (E2E) and compositional. E2E approaches, exemplified by the Translatotron series (Jia et al., 2019, 2022a) and SeamlessM4T (Barrault et al., 2023), map source speech to target speech within a single unified network. While conceptually attractive, they must jointly handle speech understanding, cross-lingual translation, and speech generation, often leading to optimization trade-offs across sub-tasks. Moreover, the lack of an intermediate textual representation complicates error analysis and correction, while the scarcity of large-scale speech-to-speech parallel data further limits scalability (Fang et al., 2024).

To mitigate these issues, compositional S2ST architectures decompose the task into an S2TT frontend and a TTS backend. For example, DASpeech (Fang et al., 2023) adopts a two-pass design in which a linguistic decoder predicts target-side representations and an acoustic decoder (FastSpeech 2 (Ren et al., 2020)) generates speech conditioned on them. These models improve controllability and latency, but are still trained as direct S2ST systems and heavily rely on costly source-to-target speech pairs. In contrast, ComSpeech (Fang et al., 2024) explicitly integrates pretrained S2TT and TTS models via CTC-based vocabulary adaptors, thereby leveraging abundant S2TT/TTS corpora and exposing an interpretable intermediate text. However, its vocabulary adaptors and phoneme-level embeddings provide relatively shallow, weakly contextualized interfaces, limiting robustness on seman-

† denotes the corresponding author.

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tically complex or ambiguous utterances. More importantly, existing systems provide little explicit modeling of source-language information at either the acoustic or linguistic level, despite its importance for accurate multilingual translation.

In this paper, we propose **S2ST-Omni**, a high-accuracy and flexible compositional S2ST framework that couples a SpeechLLM-based S2TT frontend with a plug-and-play TTS backend. Our motivation is that existing compositional S2ST systems still fall short in two key aspects: they often lack a sufficiently strong interface between pretrained speech and language models, and they underexploit source-language information that is crucial for multilingual translation. To address these limitations, S2ST-Omni integrates three complementary designs: a **hybrid speech adapter** to better bridge a pretrained Whisper-large-v3-based speech encoder (Radford et al., 2023) and a Qwen3 LLM (Yang et al., 2025), a **hierarchical language-aware** (HLA) augmentation framework for modeling source-language information at both acoustic and linguistic levels, including FiLM-based language-aware supervision (Perez et al., 2018), and a task-specific **progressive fine-tuning** (PFT) strategy for stable and efficient optimization. Notably, HLA reuses Whisper encoder features for language identification, achieving near-oracle accuracy (99.67–99.80%) with only 22.8 ms additional latency and without requiring a dedicated classifier. Meanwhile, the framework remains fully compositional and can be combined with state-of-the-art TTS backends (Chen et al., 2024; Peng et al., 2025; Zhou et al., 2025a,b). Together, these designs improve translation fidelity while preserving the modularity, interpretability, and backend flexibility of the compositional paradigm.

Extensive experiments on CVSS-C (Jia et al., 2022b) show that S2ST-Omni consistently outperforms eight recent E2E and compositional S2ST baselines on French, German, and Spanish → English translation. In particular, it achieves the best BLEU, COMET, and ASR-BLEU scores across all evaluated directions, showing that stronger speech-to-LLM adaptation and explicit source-language modeling are highly effective for compositional multilingual S2ST.

In summary, we introduce *S2ST-Omni*, a compositional S2ST framework that combines a SpeechLLM-based S2TT frontend with a plug-and-play TTS backend. Our approach bridges Whisper and Qwen3 through a local-then-global hybrid speech adapter, and explicitly incorporates source-language information predicted from Whisper via HLA augmentation, including LA-Dual-CTC and LAP. Notably, language identity is obtained by reusing Whisper encoder features, achieving near-oracle accuracy with minimal latency overhead and without requiring a dedicated classifier. To enable stable and effective optimization, we further propose a task-specific PFT strategy that first establishes robust speech–text alignment and then enhances translation capability via LoRA-based adaptation (Hu et al., 2022). Experiments on CVSS-C (Jia et al., 2022b) show that *S2ST-Omni* achieves the best BLEU, COMET, and ASR-BLEU scores among strong E2E and compositional baselines on French, German, and Spanish → English translation.

2 Background

2.1 Speech-to-Speech Translation

Existing S2ST systems broadly fall into two paradigms: E2E and compositional. E2E models such as Translatotron series (Jia et al., 2019, 2022a) and SeamlessM4T (Barrault et al., 2023) map source speech directly to target speech within a single network, but must jointly learn speech understanding, translation, and generation, which complicates optimization and typically demands large amounts of speechspeech parallel data. Compositional approaches instead factorize S2ST into an S2TT frontend plus a TTS backend. ComSpeech (Fang et al., 2024) links pretrained S2TT and TTS models via CTC-based vocabulary adapters, while StreamSpeech (Zhang et al., 2024) builds a streaming cascaded system with online S2TT and expressive TTS. Despite their advantages in interpretability and modularity, these systems still rely on relatively shallow adapters and task-specific S2TT training, and they do not explicitly model source-language information at the acoustic or linguistic level, leaving room for more language-aware and SpeechLLM-centric designs.

2.2 Speech Large Language Models

Recent SpeechLLM frameworks extend text LLMs to audio either via discrete tokens (e.g., SpeechGPT (Zhang et al., 2023), AudioPaLM (Rubenstein et al., 2023)) or continuous adapters that connect speech encoders such as Whisper to LLM backbones (e.g., SALMONN (Tang et al., 2024), Qwen-Audio (Chu

et al., 2023), Qwen2-Audio (Chu et al., 2024)). These models achieve strong results on ASR and spoken understanding and can perform speech-to-text translation as part of a multitask setup. However, they are not tailored for high-accuracy multilingual S2TT and lack explicit mechanisms to inject source-language information into either the acoustic representations or the prompting interface, which is precisely the gap our HLA-based Speech-LLM framework is designed to address.

2.3 CTC for Speech Processing

Connectionist Temporal Classification (CTC) (Graves et al., 2006) provides a principled framework for sequence-to-sequence learning without explicit alignment supervision, which has been widely adopted as an auxiliary objective to improve speech-text alignment in encoder-decoder models (Kim et al., 2017; Watanabe et al., 2017; Yang et al., 2023). Given an input sequence $\mathbf{x} = (x_1, \ldots, x_T)$ and target sequence $\mathbf{y} = (y_1, \ldots, y_U)$ where $U \leq T$, CTC marginalizes over all valid alignments π that collapse to $\mathbf{y}$:

$$P(\mathbf{y}|\mathbf{x}) = \sum_{\pi \in \mathcal{B}^{-1}(\mathbf{y})} P(\pi|\mathbf{x}) \tag{1}$$

where $\mathcal{B}$ denotes the collapsing function that removes blanks and repeated tokens.

3 Methodology

3.1 System Overview

As illustrated in Fig. 1, *S2ST-Omni* adopts a compositional architecture that decomposes S2ST into two largely independently optimizable modules: a high-accuracy S2TT frontend and a flexible TTS backend. The S2TT frontend comprises four core components: (i) a Whisper Large-v3 encoder that provides robust multilingual speech representations; (ii) a Qwen3-4B LLM that serves as the translation backbone; (iii) a hybrid speech adapter that bridges Whisper and Qwen3 via a hierarchical "local-then-global" strategy; and (iv) a HLA architecture consisting of an acoustic-level LA-Dual-CTC and a linguistic-level LAP. The TTS backend is kept fully modular and can be instantiated with any SOTA synthesizer.

3.2 Hybrid Speech Adapter

Adapting continuous speech representations to an LLM requires modeling information across multiple scales, from phonetic-level acoustic patterns to sentence-level semantics. Conformer-style adapters (Dubey et al., 2024; Xu et al., 2025) that interleave convolution and attention in every layer offer a general solution; however, early global mixing can dilute fine-grained local cues, and computing self-attention on long speech sequences is computationally expensive. At the other extreme, MLP-style adapters (Fang et al., 2025; He et al., 2025) that apply only per-frame linear projections are parameter-efficient but lack explicit temporal modeling, thereby shifting the burden of capturing short-term phonetic patterns onto the LLM, which is primarily designed for higher-level semantic reasoning.

We therefore propose a *Hybrid Adapter* that follows a "local-then-global" hierarchy. Given Whisper encoder outputs $\mathbf{H} \in \mathbb{R}^{T \times d_s}$ with frame length T and hidden size $d_s = 1280$, we first project them into an adapter hidden space $d_h = 1024$ via a linear layer with LayerNorm and GELU, yielding $\mathbf{H}^{(0)} \in \mathbb{R}^{T \times d_h}$. On top of $\mathbf{H}^{(0)}$, we stack two depthwise separable convolution blocks. Each block applies LayerNorm, a depthwise-separable 1D convolution with kernel size $k = 7$ to capture local acoustic patterns, followed by a position-wise feed-forward network (FFN) with expansion ratio 4 and a residual connection.

To shorten the sequence before global attention, we apply a strided 1D convolution with kernel size of 5 and stride of 2 to the final output $\mathbf{H}^{(N)}$, followed by LayerNorm and GELU, obtaining a downsampled representation $\mathbf{H}^{\text{down}} \in \mathbb{R}^{T' \times d_h}$ with $T' = \lceil T/2 \rceil$. The learnable downsampling adaptively preserves translation-relevant information while discarding redundant frames. We then feed $\mathbf{H}^{\text{down}}$ into two standard Transformer blocks, each composed of multi-head self-attention with 4 heads and a FFN (expansion ratio 4), both wrapped with LayerNorm and residual connections, to model long-range semantic dependencies and produce a global representation $\mathbf{H}^{\text{glob}} \in \mathbb{R}^{T' \times d_h}$. Finally, we apply LayerNorm and a linear projection $\mathbf{W}_{\text{out}} \in \mathbb{R}^{d_h \times d_l}$ with $d_l = 3584$ to map $\mathbf{H}^{\text{glob}}$ into the LLM hidden space, yielding $\mathbf{Z} \in \mathbb{R}^{T' \times d_l}$. This sequence $\mathbf{Z}$ is fed to the LLM, while the intermediate features $\mathbf{H}^{\text{down}}$ are reused by LA-Dual-CTC for language-aware auxiliary supervision.

3.3 HLA Augmentation Architecture

In multilingual S2ST, we suppose that explicit source-language awareness is crucial for accu-

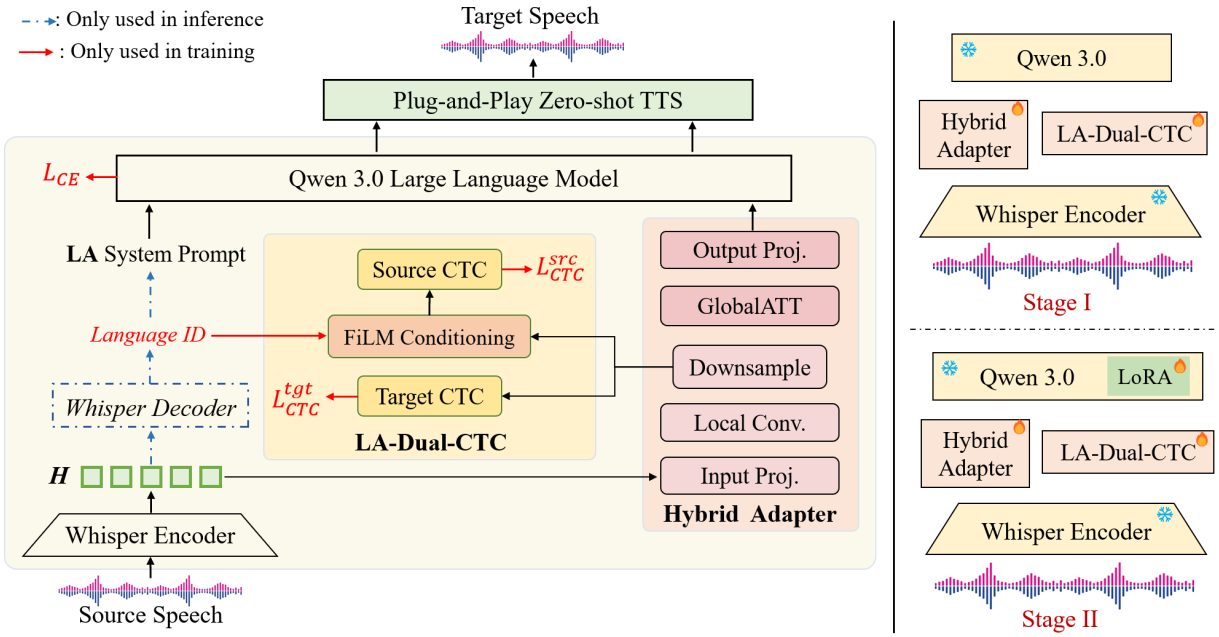

Figure 1: Overall architecture of the proposed S2ST-Omni framework. LA denotes language-aware.

rate translation, as different languages exhibit distinct syntactic structures (e.g., verb-final German clauses, post-nominal French adjectives), word-order patterns, and phonological systems. To capture these differences, we propose a *HLA augmentation* architecture that injects source-language information at two complementary levels: (i) LAP operates at the LLM input to guide high-level translation behavior; and (ii) LA-Dual-CTC operates on adapter features to modulate low-level acoustic representations. Together, they provide bottom-up and top-down language awareness.

3.3.1 LAP

Pretrained LLMs acquire rich cross-lingual knowledge from large-scale multilingual corpora, but effectively leveraging this knowledge requires explicit cues about the input language and task. LAP provides such cues by dynamically constructing language-aware prompts.

Given a source-language identifier ℓ (obtained from Whispers built-in language ID during inference, and from dataset labels during training), we instantiate a prompt $\mathcal{P}(\ell)$ using a simple template: *The following is* [LANG] *speech. Translate it accurately into English.*, where [LANG] is the full language name corresponding to ℓ (e.g., `fr` $\rightarrow$ *French*). We encode $\mathcal{P}(\ell)$ with the LLM tokenizer to obtain prompt embeddings $\mathbf{E}(\mathcal{P}(\ell))$, which are concatenated with the speech representation $\mathbf{Z}$ and the target text embeddings $\mathbf{E}(\mathbf{y})$ to form the LLM input sequence. By this means, LAP is able to explicitly declare the source language and steer the LLM towards the appropriate language-pair translation mode, without introducing additional cost.

3.3.2 LA-Dual-CTC

Although LAP operates at the linguistic level, we also aim to encode language-specific inductive biases directly into the speech representation. To this end, we design *LA-Dual-CTC*, which injects source-language information into the CTC heads via FiLM-style conditioning.

Detailed, we apply LA-Dual-CTC to the downsampled intermediate features $\mathbf{H}^{\text{down}}$ from the Hybrid Adapter (Section 3.2) rather than to its final output. The intuition behind is that this design ensures that $\mathbf{H}^{\text{down}}$ both preserves the monotonic alignment structure required by CTC and retains sufficient acoustic detail, while decoupling the CTC objective from the subsequent global-attention layers that are primarily responsible for high-level semantic modeling. Then, we obtain a language embedding $\mathbf{e}_\ell = \text{Embed}(\ell)$ and feed it into a small MLP to generate FiLM parameters $(\boldsymbol{\gamma}_\ell, \boldsymbol{\beta}_\ell) \in \mathbb{R}^{d_h} \times \mathbb{R}^{d_h}$, where ℓ denotes the source language. To control the overall strength of language conditioning, we introduce a learnable scalar gate g. Empirically, we initialize $g = 0.5$, which provides a moderate degree of modulation at the start of training and allows the model to adaptively increase or decrease its reliance on language-

Algorithm 1 Task-Specific PFT

Require: Dataset $\mathcal{D} = \{(x, y, \ell)\}$, encoder parameters θ_{enc}, LLM parameters θ_{llm}
Initialize Hybrid Adapter θ_{adpt}, LA-Dual-CTC heads θ_{ctc}, LoRA parameters θ_{lora}
Freeze θ_{enc} and base θ_{llm} in all stages
Stage I (alignment): optimize $\theta_{\text{adpt}}, \theta_{\text{ctc}}$
for $(x, y, \ell) \sim \mathcal{D}$ **do**
    Compute losses $\mathcal{L}_{\text{CE}}, \mathcal{L}_{\text{CTC}}^{\text{src}}, \mathcal{L}_{\text{CTC}}^{\text{tgt}}$
    Update $\theta_{\text{adpt}}, \theta_{\text{ctc}}$ w.r.t. $\mathcal{L}^{(1)}$
end for
Stage II (translation): enable θ_{lora}, reduce lr for $\theta_{\text{adpt}}, \theta_{\text{ctc}}$
for $(x, y, \ell) \sim \mathcal{D}$ **do**
    Compute losses $\mathcal{L}_{\text{CE}}, \mathcal{L}_{\text{CTC}}^{\text{src}}, \mathcal{L}_{\text{CTC}}^{\text{tgt}}$
    Update $\theta_{\text{adpt}}, \theta_{\text{ctc}}, \theta_{\text{lora}}$ w.r.t. $\mathcal{L}^{(2)}$
end for
Ensure: Trained S2TT frontend

specific conditioning as learning progresses. The gated FiLM transformation produces language-conditioned features

$$\mathbf{Z}^{\text{src}} = (1 + g\,\boldsymbol{\gamma}_\ell) \odot \mathbf{H}^{\text{down}} + g\,\boldsymbol{\beta}_\ell, \qquad (2)$$

on which the *source* CTC head is applied to produce logits $\mathbf{O}^{\text{src}}$. Because the target language is fixed to English, the *target* CTC head operates directly on $\mathbf{H}^{\text{down}}$ without language conditioning to produce logits $\mathbf{O}^{\text{tgt}}$.

3.4 Task-Specific PFT

Optimizing the S2TT frontend jointly over a frozen Whisper encoder, a large pretrained LLM, a newly initialized adapter, and auxiliary CTC heads is challenging due to heterogeneous initialization and optimization dynamics. As illustrated in Algorithm 1, we therefore adopt a task-specific two-stage *PFT* strategy: Stage I primarily builds a robust speech–text alignment under strong dual-CTC supervision, while Stage II injects translation capability into the LLM via parameter-efficient adaptation under weaker CTC regularization.

3.4.1 Stage I: Alignment Foundation

In Stage I, we freeze the Whisper encoder and the base Qwen3 parameters, and train only the Hybrid Adapter and LA-Dual-CTC. The LLM thus acts as a fixed conditional language model, while the adapter is optimized to produce features that are jointly aligned with source and target text.

Let $\mathcal{L}_{\text{CE}}$ denote the autoregressive cross-entropy loss of the LLM decoder, and $\mathcal{L}_{\text{CTC}}^{\text{src}}$ and $\mathcal{L}_{\text{CTC}}^{\text{tgt}}$ denote the CTC losses on the source and target branches of LA-Dual-CTC, respectively. The Stage I objective is

$$\mathcal{L}^{(1)} = \mathcal{L}_{\text{CE}} + \alpha_1\,\mathcal{L}_{\text{CTC}}^{\text{src}} + \beta_1\,\mathcal{L}_{\text{CTC}}^{\text{tgt}}, \qquad (3)$$

where we empirically set $\alpha_1 = 0.1$ and $\beta_1 = 0.2$ to emphasize reliable alignment.

3.4.2 Stage II: Translation Enhancement

In Stage II, we introduce parameter-efficient adaptation on the LLM while continuing to train the Hybrid Adapter and LA-Dual-CTC with reduced learning rates. Concretely, we apply LoRA to the query and value projections in the self-attention modules of Qwen3, with rank $r = 8$, scaling factor $\alpha = 32$, and dropout rate 0.1, and optimize these LoRA parameters with a higher learning rate than the adapter and CTC heads. To shift the focus towards end-to-end translation quality while retaining alignment signals, we down-weight the CTC losses and use

$$\mathcal{L}^{(2)} = \mathcal{L}_{\text{CE}} + \alpha_2\,\mathcal{L}_{\text{CTC}}^{\text{src}} + \beta_2\,\mathcal{L}_{\text{CTC}}^{\text{tgt}}, \qquad (4)$$

with $\alpha_2 = 0.01$ and $\beta_2 = 0.05$. This two-stage schedule first stabilizes speech–text alignment and then refines translation behavior on top of this converged foundation.

3.5 TTS Backend

S2ST-Omni supports plug-and-play integration with any SOTA speech synthesizer as the TTS backend. The intermediate English text produced by the S2TT frontend is passed directly to an external TTS model, without any task-specific coupling. In our experiments, we instantiate the backend with IndexTTS 2 (Zhou et al., 2025a), a zero-shot TTS system that can generate natural and fluent speech. This modular design allows practitioners to select or swap TTS systems according to application requirements without any retraining or modifying the S2TT frontend, reducing development and deployment overhead.

4 Experiments

4.1 Experimental Settings

4.1.1 Datasets

We use the CVSS-C benchmark (Jia et al., 2022b) for both model fine-tuning and evaluation. CVSS-C provides parallel triplets ⟨source speech, target

Table 1: Overall performance comparison among evaluated S2ST methods on CVSS-C.

Models	FR → EN		ES → EN		DE → EN		Avg.	
	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑
Ground Truth	-	84.52	-	88.54	-	75.53	-	-
Translatotron	-	16.96	-	8.72	-	1.97	-	9.22
Translatotron2	28.82	26.07	25.82	22.93	18.66	16.91	24.43	21.97
S2UT	-	22.23	-	18.53	-	2.99	-	14.58
UnitY	-	27.77	-	24.95	-	18.74	-	23.82
DASpeech	-	25.03	-	21.37	-	16.14	-	20.85
ComSpeech	30.72	28.15	26.51	24.80	19.41	18.16	25.55	23.70
StreamSpeech	32.60	28.45	30.35	27.25	23.36	20.93	28.77	25.54
Hibiki	-	30.50	-	-	-	-	-	-
Ours	**35.83**	**33.20**	**37.85**	**35.90**	**33.34**	**31.25**	**35.67**	**33.45**

text, target speech⟩ for speech-to-speech translation from 21 source languages into English. To ensure fair comparison with recent compositional S2ST systems (Fang et al., 2024; Zhang et al., 2024), we follow their protocol and consider three representative directions: French→English (Fr→En), German→English (De→En), and Spanish→English (Es→En). Specifically, the fine-tuning data consists of the CVSS-C supervised training sets for these three directions, including 264 hours for Fr→En, 184 hours for De→En, and 113 hours for Es→En, for a total of 561 hours. The reported results are obtained on the corresponding official test sets of the same three directions. During preprocessing, source speech is resampled to 16 kHz and target speech to 22.05 kHz.

4.1.2 Implementation Details

All experiments are implemented in PyTorch on two NVIDIA A6000 GPUs. In Stage I, we train only the Hybrid Adapter and LA-Dual-CTC for 150k steps using AdamW with a cosine schedule, learning rates of 1×10^{-5} (adapter) and 5×10^{-5} (LA-Dual-CTC), and 1k warmup steps. In Stage II, we attach LoRA modules and continue training for another 150k steps with learning rates 5×10^{-6} (adapter), 1×10^{-6} (LA-Dual-CTC), and 5×10^{-5} (LoRA). We use a batch size of 4 per GPU with gradient accumulation of 6. For LA-Dual-CTC, we employ 8k and 4k SentencePiece vocabularies for the multilingual source and English target branches, respectively, both trained on the source and target text of our training set.

4.1.3 Evaluation

We compare S2ST-Omni against eight strong S2ST baselines: Translatotron and Translatotron 2 (Jia et al., 2019, 2022a), S2UT (Lee et al., 2022a), DASpeech (Fang et al., 2023), UnitY (Inaguma et al., 2023), ComSpeech (Fang et al., 2024), StreamSpeech (Zhang et al., 2024), and Hibiki (Labiausse et al., 2025).

We use BLEU and ASR-BLEU as the primary automatic evaluation metrics. Regarding ASR-BLEU, the generated speech is first transcribed by a pretrained wav2vec 2.0 ASR model[2], and BLEU is then computed using SacreBLEU[3] with a fixed configuration [4]. Furthermore, to complement semantic-level evaluation, we additionally report COMET scores on representative competitive systems, including ComSpeech, StreamSpeech, and S2ST-Omni.

4.2 Main Results

Table 1 summarizes the overall results on CVSS-C. *S2ST-Omni* achieves the best performance across all three directions among the evaluated systems. Compared with the strongest compositional baseline StreamSpeech, our model improves the average BLEU score from 28.77 to 35.67 and the average ASR-BLEU from 25.54 to 33.45, corresponding to relative gains of approximately $+24\%$ BLEU and $+31\%$ ASR-BLEU. These results indicate that S2ST-Omni produces not only more accurate textual translations, but also synthesized speech whose ASR transcriptions are substantially closer to the ground-truth references.

To further assess semantic translation quality beyond surface-form overlap, we additionally report COMET results in Table 3. *S2ST-Omni* again achieves the best performance on all three direc-

[2] https://dl.fbaipublicfiles.com/fairseq/wav2vec/wav2vec_vox_960h_pl.pt

[3] https://github.com/mjpost/sacrebleu

[4] SacreBLEU signature: nrefs:1|case:mixed|eff:no|tok:13a|smooth:exp|version:2.6.1.dev1+gf615c7286

Table 2: Effect of the proposed hybrid speech adapter against Conformer-based and MLP-based adapters.

Model	Fr → En		Es → En		De → En		Avg.	
	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑
Ours	**35.83**	**33.20**	**37.85**	**35.90**	**33.34**	**31.25**	**35.67**	**33.45**
w/ Conformer	34.60	31.49	36.78	34.64	32.69	30.34	34.69	32.16
w/ MLP	32.30	28.96	34.70	32.47	30.09	27.54	32.37	29.66

Table 3: COMET results on CVSS-C.

Models	FR→EN	ES→EN	DE→EN
ComSpeech	70.13	66.96	57.96
StreamSpeech	76.66	74.80	65.51
S2ST-Omni	**81.94**	**83.39**	**80.73**

tions, reaching 81.94, 83.39, and 80.73 on Fr→En, Es→En, and De→En, respectively. Compared with StreamSpeech, this corresponds to absolute gains of +5.28, +8.59, and +15.22 COMET, confirming that the improvements of *S2ST-Omni* are not limited to n-gram overlap metrics, but also consistently translate into stronger semantic fidelity.

Breaking down the results by language pair, *S2ST-Omni* achieves consistent improvements. On the relatively more challenging **De→En** track, our system reaches 33.34 BLEU and 31.25 ASR-BLEU, compared with 23.36 and 20.93 for StreamSpeech, yielding large relative gains of about $+43\%$ BLEU and $+49\%$ ASR-BLEU. For **Es→En**, *S2ST-Omni* reaches 37.85 BLEU and 35.90 ASR-BLEU, improving over StreamSpeech (30.35 / 27.25) by approximately $+25\%$ and $+32\%$, respectively. On **Fr→En**, our model obtains 35.83 BLEU and 33.20 ASR-BLEU, which corresponds to relative gains of about $+10\%$ BLEU and $+17\%$ ASR-BLEU over StreamSpeech, and it also surpasses the recently proposed Hibiki that is specifically optimized for ASR-BLEU on this language pair.

Overall, we attribute these consistent gains to the combination of the LA-Dual-CTC–augmented hybrid adapter and the HLA architecture. The hybrid adapter provides a stronger speech-to-LLM interface by first consolidating local acoustic structure and then modeling long-range dependencies on a downsampled sequence, while LA-Dual-CTC and LAP inject complementary acoustic- and linguistic-level language cues. Together with the progressive fine-tuning schedule, these components enable *S2ST-Omni* to more effectively exploit Whisper and Qwen3 for multilingual S2TT and S2ST, yielding best performance on CVSS-C.

4.3 Ablation Study

4.3.1 Effect of Hybrid Speech Adapter

Table 2 compares our hybrid adapter with Conformer- and MLP-based variants. Replacing the hybrid adapter with the Conformer design results in a consistent drop across all directions: the average BLEU score decreases from 35.67 to 34.69 (about 2.8% relative), and the average ASR-BLEU from 33.45 to 32.16 (about 3.9%), with similar trends for each language pair. In contrast, the MLP adapter causes a substantially larger degradation, reducing the average BLEU/ASR-BLEU to 32.37/29.66, corresponding to roughly 9% and 11% relative drops, respectively. Overall, these results suggest that MLP-style adaptation is insufficient to bridge Whisper and Qwen3 for S2ST, and that densely interleaving convolution and attention (Conformer-style) is also suboptimal, highlighting the effectiveness of the proposed hybrid adapter.

4.3.2 Effect of HLA Architecture

Table 4 examines the contribution of our HLA architecture. Removing either LAP or LA-Dual-CTC leads to small but consistent performance drops: removing LAP reduces the average BLEU/ASR-BLEU from 35.67/33.45 to 35.25/32.89, while removing LA-Dual-CTC yields 35.16/32.72. In contrast, removing both components (w/o HLA) causes a much larger degradation, bringing the averages down to 33.68 BLEU and 30.95 ASR-BLEU, which corresponds to roughly 5.6% and 7.5% relative drops. These results indicate that linguistic-level prompting (LAP) and acoustic-level FiLM conditioning (LA-Dual-CTC) are complementary: each provides modest gains on its own, while jointly encoding source-language information at both acoustic and linguistic levels yields a stronger inductive bias for multilingual S2ST.

4.3.3 Effect of Task-Specific PFT

Table 5 evaluates the effect of the proposed two-stage PFT strategy. The Stage-I-only variant (w/o PFT-II), which omits LoRA-based LLM adapta-

Table 4: Effect of the proposed HLA architecture.

Model	Fr → En		Es → En		De → En		Avg.	
	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑
Ours	**35.83**	**33.20**	**37.85**	**35.90**	**33.34**	**31.25**	**35.67**	**33.45**
w/o LAP	35.25	32.36	37.37	35.30	33.13	31.01	35.25	32.89
w/o LA-Dual-CTC	35.16	32.11	37.22	35.09	33.10	30.97	35.16	32.72
w/o HLA	33.49	30.26	35.96	33.77	31.57	28.82	33.68	30.95

Table 5: Effect of the task-specific PFT strategy. "w/o PFT-II" denotes removing Stage II LoRA-based LLM adaptation; "w/o PFT" represents training in a single stage without the proposed two-stage schedule.

Model	Fr → En		Es → En		De → En		Avg.	
	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑	BLEU↑	ASR-BLEU↑
Ours	**35.83**	**33.20**	**37.85**	**35.90**	**33.34**	**31.25**	**35.67**	**33.45**
w/o PFT-II	33.17	29.78	35.58	33.36	31.23	28.74	33.33	30.63
w/o PFT	34.24	31.12	36.41	34.24	32.46	30.08	34.37	31.81

Table 6: ASR-BLEU results of different TTS backends.

Models	Fr→En	Es→En	De→En	Avg.
IndexTTS2	33.20	35.90	31.25	33.45
FireRedTTS2	31.55	35.18	30.94	32.56
CosyVoice3	31.56	35.15	30.36	32.36
ZipVoice	31.48	35.08	30.46	32.34
VoxCPM1.5	31.35	34.82	30.20	32.09

tion, exhibits the largest degradation: the average BLEU/ASR-BLEU drops from 35.67/33.45 to 33.33/30.63, with a particularly pronounced decline on De→En (33.34/31.25 → 31.23/28.74). Collapsing the two phases into a single-stage training (w/o PFT) mitigates the drop but still underperforms the full schedule, reducing the average BLEU/ASR-BLEU to 34.37/31.81 (about 3.6% and 4.9% relative drops). These results suggest that optimizing alignment and translation simultaneously can introduce harmful gradient interference, whereas the proposed two-stage schedule: first emphasizing dual-CTC alignment, then refining translation with LoRA-based LLM adaptation, provides a more stable and effective optimization path for the S2TT frontend.

4.4 TTS Backend Analysis

Table 6 examines how the choice of TTS backend affects synthesis quality when the S2TT frontend of *S2ST-Omni* is fixed and only the TTS module is swapped. Overall, the ASR-BLEU scores are relatively stable across five SOTA TTS systems: the average gap between the best configuration (IndexTTS2, 33.45) and the weakest (VoxCPM1.5, 32.09) is within roughly 1.4 points (about 4% relative), and all backends fall into a narrow band on each language pair. Notably, even the weakest backend (VoxCPM1.5) achieves an average ASR-BLEU of 32.09, which remains higher than StreamSpeech under the same evaluation setting.

These results indicate that, given a strong SpeechLLM-based S2TT frontend, the synthesis performance is largely insensitive to the specific choice of modern TTS backend. In practice, *S2ST-Omni* can thus be paired with diverse off-the-shelf TTS systems. Practitioners can choose the backend based on latency, speaker style, or deployment constraints, without redesigning or retraining the translation module.

4.5 Latency Analysis

Table 7: Average latency (ms) of the S2TT frontend of *S2ST-Omni* on CVSS-C (Fr/Es/De) test sets.

Module	Whisper Enc.	LID Dec.	Adapter	LLM	Total
Latency	118.4	22.8	6.8	555.8	703.8

As shown in Table 7, we report the latency of the S2TT frontend of *S2ST-Omni* on the CVSS-C test sets. For each utterance, we measure the latency of (i) Whisper encoding, (ii) Whisper decoding for language identification, (iii) the hybrid adapter, and (iv) LLM prefill and generation with batch size 1 on a single RTX A5000 GPU.

On average, the S2TT frontend requires about 704 ms per utterance across Fr/Es/De test sets. LLM decoding is the dominant cost (roughly 79% of the total time), Whisper encoding is the second

largest contributor (about 17%). Language-ID decoding and the hybrid adapter together contribute only about 4.2% to the overall latency. Additionally, since the TTS backend in *S2ST-Omni* is fully modular, practitioners can select a synthesizer that best matches their latency and quality requirements without modifying or retraining the S2TT frontend.

5 Conclusion

We presented *S2ST-Omni*, an accurate and flexible hierarchical language-aware framework for multilingual speech-to-speech translation. By coupling a SpeechLLM-based S2TT frontend with a plug-and-play TTS backend, S2ST-Omni decouples translation from synthesis, enabling independent optimization of each module while preserving an interpretable text intermediate. At the core of the frontend, a hybrid "local-then-global" speech adapter, augmented with LA-Dual-CTC and LAP, provides a strong speechLLM interface that jointly models local acoustic patterns, long-range semantics, and explicit source-language cues. A task-specific two-stage PFT strategy further stabilizes training by first establishing robust speech-text alignment and then improving translation capability via LoRA-based LLM adaptation. Experiments on CVSS-C demonstrate that S2ST-Omni outperforms eight strong E2E and compositional S2ST baselines on Fr→En, Es→En, and De→En, achieving the best BLEU and ASR-BLEU scores. Ablation studies further validate the effectiveness and contribution of each proposed component.

6 Limitations

While S2ST-Omni achieves strong performance on CVSS-C, several limitations remain. First, our empirical study is limited to three high-resource source languages (French, Spanish, German) with English as the only target language, and to a single benchmark. This leaves open questions about the robustness of the proposed hybrid adapter, HLA architecture, and PFT strategy in low-resource settings, many-to-many S2ST, or more diverse domains and speaking styles. Second, the current instantiation relies on large backbone models (Whisper Large-v3 and Qwen3-4B), which require nontrivial computational resources for both training and inference. While the hybrid adapter and PFT are designed to be parameter-efficient, deploying S2ST-Omni on edge devices or under strict latency/energy constraints may remain challenging and may require model compression or distillation. Third, our hierarchical language-aware design assumes access to a reliable source-language identifier. At inference, we reuse Whispers encoder and decoder for language prediction, which adds negligible latency and introduces a small amount of noise. In our experiments, language ID accuracy is 99.76% (De), 99.67% (Fr), and 99.80% (Es). Finally, our evaluation focuses on automatic metrics (BLEU and ASR-BLEU) and does not include human judgments of translation adequacy, naturalness, or speaker similarity, nor a detailed analysis of end-to-end latency and computational cost across deployment settings. A more comprehensive evaluation along these dimensions is left for future work.

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