# Towards Efficient Simultaneous Speech Translation: CUNI-KIT System for Simultaneous Track at IWSLT 2023

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## Abstract

In this paper, we describe our submission to the Simultaneous Track at IWSLT 2023. This year, we continue with the successful setup from the last year, however, we adopt the latest methods that further improve the translation quality. Additionally, we propose a novel online policy for attentional encoder-decoder models. The policy prevents the model to generate translation beyond the current speech input by using an auxiliary CTC output layer. We show that the proposed simultaneous policy can be applied to both streaming blockwise models and offline encoder-decoder models. We observe significant improvements in quality (up to 1.1 BLEU) and the computational footprint (up to 45 % relative RTF).

# 1 Introduction

Simultaneous speech translation (SST) is the task of translating speech into text in a different language before the utterance is finished. The goal of SST is to produce a high-quality translation in real-time while maintaining low latency. However, these two objectives are conflicting. If we decrease the latency, the translation quality also drops. Last year's IWSLT evaluation campaign (Anastasopoulos et al., 2022) showed that current methods for simultaneous speech translation can approach the translation quality of human interpreters (Polák et al., 2022). The disadvantage is a higher computation footprint that might make a widespread application prohibitive.

This paper describes the CUNI-KIT submission to the Simultaneous translation track at IWSLT 2023 (Agarwal et al., 2023). Following our last year's submission (Polák et al., 2022), we continue in our effort to onlinize the robust offline speech translation models. However, the main goal of this submission is to improve the computational footprint. To this end, we propose a novel online policy based on CTC. As we experimentally document, the online CTC policy can be used to onlinize the offline models achieving a 45 % improvement in real time factor (RTF) as well as to improve the quality of the streaming blockwise models (Tsunoo et al., 2021). Aside from improving the online policy, we also adopt the novel improved streaming beam search (Polák et al., 2023) that further improves the translation quality.

Our contributions are as follows:

- We adopt the latest online decoding algorithm that improves the translation quality of robust offline models in the simultaneous regime,
- We propose a novel online policy that significantly
  - lowers the computational complexity of the online decoding with robust offline models while maintaining the same or only slightly worse translation quality,
  - improves the translation quality of the streaming blockwise models while maintaining the same latency,
- We demonstrate that our systems can run on hardware accessible to a wide audience.

## 2 Methods

In our submission, we use two different model architectures — a traditional offline ST architecture and a blockwise simultaneous ST architecture (Tsunoo et al., 2021). In this section, we describe the methods applied to achieve simultaneous ST using these architectures.

## 2.1 Incremental Blockwise Beam Search with Controllable Quality-Latency Tradeoff

To use the traditional offline ST model in a simultaneous regime, Liu et al. (2020) proposed chunking, i.e., splitting the audio source utterance into small constant-length chunks that are then incrementally fed into the model. As translation quality tends to diminish toward the end of the unfinished source, an online policy is employed to control the latencyquality tradeoff in the generated output. Popular online policies include wait-k (Ma et al., 2019), shared prefix (Nguyen et al., 2020), hold-n and local agreement (Liu et al., 2020). In Polák et al. (2022), we showed that the tradeoff could be controlled by varying the chunk length.

To generate the translation, a standard beam search is typically applied (Sutskever et al., 2014). While this decoding algorithm enables the model to generate a complete translation for the current input, it also suffers from overgeneration (i.e., hallucinating tokens beyond sounds present in the input segment) and low-quality translations towards the end of the source context (Dong et al., 2020; Polák et al., 2022).

To tackle this issue, we adopt an improved incremental blockwise beam search (Polák et al., 2023). We outline the algorithm in Algorithm 1 and highlight the main differences from the original approach used in Polák et al. (2022) with red.

Algorithm 1:	Incremental	blockwise
streaming beam	search algorith	m for incre-

	0				
streaming beam search algorithm for incre-					
me	ental ST				
In	<b>put</b> : A list of blocks, an ST model				
0	utput: A set of hypotheses and scores				
1 Se	$een \leftarrow \emptyset;$				
2 fo	r each block do				
3	Encode block using the ST model;				
4	Stopped $\leftarrow \emptyset$ ;				
5	$minScore \leftarrow -\infty;$				
6	while #active beams $> 0$ and not max. length do				
7	Extend beams and compute scores;				
8	for each active beam b do				
9	if b ends with $< eos > or (score \le minScore$				
	and $b \notin Seen$ ) then				
10	$minScore \leftarrow max(minScore, score);$				
11	$Stopped \leftarrow Stopped \cup b;$				
12	Remove b from the beam search;				
13	end				
14	end				
15	end				
16	Seen $\leftarrow$ Seen $\cup$ Stopped;				
17	Sort Stopped by length-normalized score;				

Sort Stopped by length-normalized score;
Set the best hypothesis from Stopped as active beam;
Apply the incremental policy;
Remove the last two tokens from the active beam;
and

In Algorithm 1, the overgeneration problem is addressed by stopping unreliable beams (see Line 9). The unreliable beam is defined as a beam ending with  $< e \circ s >$  token or having a score lower or equal to any other unreliable beam detected so far. This means, that we stop any beam that has a score lower than any beam ending with  $< e \circ s >$  token. Since there might be a hypothesis that would always score lower than some hypothesis ending with the  $< e \circ s >$  token, the algorithm allows generating a hypothesis with a score lower than the unreliable score if it was seen during the decoding of previous blocks.

Finally, the algorithm removes two instead of one token in the current beam (see Line 20). Removing the last two tokens mitigates the issue of low-quality translation toward the end of the context.<sup>1</sup>

## 2.2 Rethinking Online Policies for Attention-based ST Models

While the improved incremental blockwise beam search improves the performance, it still requires a strong online policy such as hold-n or local agreement (Liu et al., 2020). A common property of these online policies is that they require multiple re-generations of the output translation. For example, the local agreement policy must generate each token at least twice to show it to the user, as each token must be independently generated by two consecutive contexts to be considered stable. Depending on the model architecture, the generation might be the most expensive operation. Additionally, the sequence-to-sequence models tend to suffer from exposure bias (i.e., the model is not exposed to its own errors during the training) (Ranzato et al., 2015; Wiseman and Rush, 2016). The exposure bias then causes a lower translation quality, and sometimes leads to hallucinations (i.e., generation of coherent output not present in the source) (Lee et al., 2018; Müller et al., 2019; Dong et al., 2020). Finally, attentional encoder-decoder models are suspected to suffer from label bias (Hannun, 2020).

A good candidate to address these problems is CTC (Graves et al., 2006). For each input frame, CTC predicts either a blank token (i.e., no output) or one output token independently from its previous predictions, which better matches the streaming translation and reduces the risk of hallucinations. Because the CTC's predictions for each frame are conditionally independent, CTC does not suffer from the label bias problem (Hannun, 2020). Although, the direct use of CTC in either machine or speech translation is possible, yet, its quality lags behind autoregressive attentional modeling (Libovický and Helcl, 2018; Chuang et al., 2021).

<sup>&</sup>lt;sup>1</sup>Initial experiments showed that removing more than two tokens leads to higher latency without any quality improvement.

Another way, how to utilize the CTC is joint decoding (Watanabe et al., 2017; Deng et al., 2022). In the joint decoding setup, the model has two decoders: the non-autoregressive CTC (usually a single linear layer after the encoder) and the attentional autoregressive decoder. The joint decoding is typically guided by the attentional decoder, while the CTC output is used for re-scoring. Since the CTC predicts hard alignment, the rescoring is not straightforward. To this end, Watanabe et al. (2017) proposed to use the CTC prefix probability (Graves, 2008) defined as a cumulative probability of all label sequences that have the current hypothesis h as their prefix:

$$p_{\rm ctc}(h,\ldots) = \sum_{\nu \in \mathcal{V}^+} p_{\rm ctc}(h \oplus \nu | X), \qquad (1)$$

where  $\mathcal{V}$  is output vocabulary (including the  $< e \circ s >$  symbol),  $\oplus$  is string concatenation, and X is the input speech. To calculate this probability effectively, Watanabe et al. (2017) introduce variables  $\gamma_t^{(b)}(h)$  and  $\gamma_t^{(n)}(h)$  that represent forward probabilities of h at time t, where the superscript denotes whether the CTC paths end with a blank or non-blank CTC symbol. If the hypothesis h is a complete hypothesis (i.e., ends with the  $< e \circ s >$  token), then the CTC probability of  $h = g \oplus < e \circ s >$  is:

$$p_{\text{ctc}}(h|X) = \gamma_T^{(b)}(g) + \gamma_T^{(n)}(g),$$
 (2)

where T is the final time stamp.

If  $h = g \oplus c$  is not final, i.e.,  $c \neq \langle e \circ s \rangle$ , then the probability is:

$$p_{\mathsf{ctc}}(h|X) = \sum_{t=1}^{T} \Phi_t(g) \cdot p(z_t = c|X), \quad (3)$$

where

$$\Phi_t(g) = \gamma_{t-1}^{(b)}(g) + \begin{cases} 0 & \text{last}(g) = c \\ \gamma_{t-1}^{(n)}(g) & otherwise. \end{cases}$$

#### 2.3 CTC Online Policy

Based on the the definition of  $p_{\text{ctc}}(h|X)$  in Equations (2) and (3), we can define the odds of g being at the end of context T:

$$\operatorname{Odds}_{\operatorname{end}}(g) = \frac{p_{\operatorname{ctc}}(g \oplus \langle \operatorname{eos} \rangle | X)}{\sum_{c \in \mathcal{V} / \{\langle \operatorname{eos} \rangle \}} p_{\operatorname{ctc}}(g \oplus c | X)}.$$
 (4)

The disadvantage of this definition is that  $p_{\text{ctc}}(\ldots | X)$  must be computed for every vocabulary entry separately and one evaluation costs  $\mathcal{O}(T)$ , i.e.,  $\mathcal{O}(|\mathcal{V}| \cdot T)$  in total. Contemporary ST systems use vocabularies in orders of thousands items making this definition prohibitively expensive. Since the CTC is used together with the label-synchronous decoder, we can approximate the denominator with a single vocabulary entry  $c_{\text{att}}$  predicted by the attentional decoder  $p_{\text{att}}$ :

$$\mathrm{Odds}_{\mathrm{end}}(g) \approx \frac{p_{\mathrm{ctc}}(g \oplus < \mathrm{eos} > |X)}{p_{\mathrm{ctc}}(g \oplus c_{\mathrm{att}}|X)}, \quad (5)$$

where  $c_{\text{att}} = \operatorname{argmax}_{c \in \mathcal{V}/\{\le \cos >\}} p_{\text{att}}(g \oplus c|X)$ . Now the evaluation of  $\operatorname{Odds}_{\operatorname{end}}(g)$  is  $\mathcal{O}(T)$ . If we consider that the baseline model already uses CTC rescoring, then evaluating  $\operatorname{Odds}_{\operatorname{end}}(g)$  amounts to a constant number of extra operations to evaluate  $p_{\operatorname{ctc}}(g \oplus < \operatorname{eos} > |X)$ .

Finally, to control the latency of the online decoding, we compare the logarithm of  $Odds_{end}(g)$  with a tunable constant  $C_{end}$ . If  $\log Odds_{end}(g) > C_{end}$ , we stop the beam search and discard the last token from g. We found values of  $C_{end}$  between -2 and 2 to work well across all models and language pairs.

#### **3** Experiments and Results

#### 3.1 Models

Our offline multilingual ST models are based on attentional encoder-decoder architecture. Specifically, the encoder is based on WavLM (Chen et al., 2022), and the decoder is based on multilingual BART (Lewis et al., 2019) or mBART for short. The model is implemented in the NMTGMinor library.<sup>2</sup> For details on the offline model see KIT submission to IWSLT 2023 Multilingual track (Liu et al., 2023).

The small simultaneous speech translation models for English-to-German and English-to-Chinese language pairs follow the blockwise streaming Transformer architecture (Tsunoo et al., 2021) implemented in ESPnet-ST-v2 (Yan et al., 2023). Specifically, the encoder is a blockwise Conformer (Gulati et al., 2020) with a block size of 40 and look-ahead of 16, with 18 layers, and a hidden dimension of 256. The decoder is a 6-layer Transformer decoder (Vaswani et al., 2017). To improve the training speed, we initialize the encoder with

<sup>&</sup>lt;sup>2</sup>https://github.com/quanpn90/NMTGMinor

weights pretrained on the ASR task. Further, we employ ST CTC (Deng et al., 2022; Yan et al., 2022) after the encoder with weight 0.3 during the training. During the decoding, we use 0.3 for English to German, and 0.4 for English to Chinese. We preprocess the audio with 80-dimensional filter banks. As output vocabulary, we use unigram models (Kudo, 2018) of size 4000 for English to German, and 8000 for English to Chinese.

#### 3.2 Evaluation

In all our experiments with the offline models, we use beam search of size 8 except for the CTC policy experiments where we use greedy search. For experiments with the blockwise models, we use the beam search of 6. For experiments with the improved blockwise beam search, we follow Polák et al. (2023) and remove the repetition detection in the underlying offline models, while we keep the repetition detection on for all experiments with the blockwise models.

For evaluation, we use Simuleval (Ma et al., 2020) toolkit and tst-COMMON test set of MuST-C (Cattoni et al., 2021). To estimate translation quality, we report detokenized case-sensitive BLEU (Post, 2018), and for latency, we report average lagging (Ma et al., 2019). To realistically assess the inference speed, we run all our experiments on a computer with Intel i7-10700 CPU and NVIDIA GeForce GTX 1080 with 8 GB graphic memory.

### 3.3 Incremental Blockwise Beam Search with Controllable Quality-Latency Tradeoff

In Table 1, we compare the performance of the onlinized version of the baseline blockwise beam search (BWBS) with the improved blockwise beam search (IBWBS; Polák et al., 2023). As we can see in the table, the improved beam search achieves higher or equal BLEU scores than the baseline beam search across all language pairs. We can observe the highest improvement in English-to-German (1.1 BLEU), while we see an advantage of 0.1 BLEU for English-to-Japanese. and no improvement in English-to-Chinese.

In Table 1, we also report the real-time factor (RTF), and the computation-aware average lagging (AL<sub>CA</sub>). Interestingly, we observe a higher computational footprint of the IBWBS compared to the baseline beam search by 13, 28, and 17 % on En $\rightarrow$ {De, Ja, Zh}, resp., when measured with RTF. This might be due to the fact that we recom-

Lang	Decoding	AL↓	$AL_{CA}\downarrow$	RTF↓	BLEU↑
En-De	BWBS IBWBS	1922 1977	<b>3121</b> 3277	<b>0.46</b> 0.52	30.6 <b>31.7</b>
En-Ja	BWBS IBWBS	1992 1935	<b>3076</b> 3264	<b>0.50</b> 0.64	15.5 <b>15.6</b>
En-Zh	BWBS IBWBS	1948 1945	<b>2855</b> 3031	<b>0.41</b> 0.48	26.5 26.5

Table 1: Incremental SST with the original BWBS and IBWBS. Better scores in bold.

pute the decoder states after each source increment. Since the IBWBS sometimes waits for more source chunks to output more tokens, the unnecessary decoder state recomputations might increase the computational complexity.

#### 3.4 CTC Online Policy

In Figure 1, we compare the improved blockwise beam search (IBWBS) with the proposed CTC policy using the blockwise streaming models. The tradeoff curves for English-to-German (see Figure 1a) and English-to-Chinese (see Figure 1b) show that the proposed CTC policy improves the quality (up to 1.1 BLEU for En $\rightarrow$ De, and 0.8 BLEU for En $\rightarrow$ Zh), while it is able to achieve the same latencies.

# 3.5 CTC Online Policy for Large Offline Models

We were also interested in whether the CTC policy can be applied to large offline models. Unfortunately, due to limited resources, we were not able to train a large offline model with the CTC output. Hence, we decided to utilize the CTC outputs of the online blockwise models and used them to guide the large offline model. Since the models have very different vocabularies,<sup>3</sup> we decided to execute the CTC policy after a whole word is generated by the offline model (rather than after every sub-word token). For the very same reason, we do not use CTC for rescoring.

We report the results in Table 2. Unlike in the blockwise models (see Section 3.4), the CTC policy does not improve the quality in  $En \rightarrow De$ , and has a slightly worse quality (by 0.7 BLEU) in  $En \rightarrow Zh$ . This is most probably due to the delayed CTC-attention synchronization that is not present for the blockwise models (as both decoders there share the

<sup>&</sup>lt;sup>3</sup>The blockwise models have a vocabulary size of 4000 for En $\rightarrow$ De and 8000 for En $\rightarrow$ Zh, and the offline model has 250k.



Figure 1: Comparison of the improved blockwise beam search (IBWBS) and the proposed CTC policy using blockwise streaming models.

same vocabulary and the models compute the CTC policy after each token rather than word). However, we still observe a significant reduction in computational latency, namely by 45 and 34 % relative RTF for En $\rightarrow$ De and En $\rightarrow$ Zh, respectively.

Lang	Decoding	AL↓	$AL_{CA}{\downarrow}$	RTF↓	BLEU↑
En-De	BWBS	1922	3121	0.46	30.6
	IBWBS	1977	3277	0.52	<b>31.7</b>
	CTC	1946	<b>2518</b>	<b>0.21</b>	30.6
En-Zh	BWBS	1948	2855	0.41	<b>26.5</b>
	IBWBS	1945	3031	0.48	<b>26.5</b>
	CTC	1981	<b>2515</b>	<b>0.28</b>	25.8

Table 2: Comparison of onlinization of the large offline model using chunking with the local agreement policy (LA-2) and with the proposed CTC policy.

## 4 Submission

In this section, we summarize our submission to the Simultaneous track at IWSLT 2023. In total, we submit 10 systems for all three language pairs.

#### 4.1 Onlinized Offline Models

Following our last year's submission, we onlinize two large offline models (our models for IWSLT 2022 Offline ST track and IWSLT 2023 Multilingual track). This year, however, we utilize the improved blockwise beam search to yield higher BLEU scores. We submit systems for all language pairs based on the last year's model, and our new model. We summarize the submitted models and their performance in Table 3. As we can observe in Table 3, the 2023 model appears to perform worse. However, we learned during the writing of this paper that there was some overlap between the training and test data for the 2022 model<sup>4</sup>, making

the BLEU scores for the 2022 model unreliable.

Lang	Model	AL↓	$AL_{CA} \downarrow$	BLEU↑
En-De	2022	1991	3138	31.8
	2023	1955	3072	31.4
En-Ja	2022	1906	3000	15.5
	2023	1982	3489	15.3
En-Zh	2022	1984	3289	26.8
	2023	1987	3508	26.6

Table 3: Submitted onlinized large offline models.

We also submit the system based on the large model onlinized using the CTC policy. The systems are summarized in Table 4. Unfortunately, we were not aware of the training and test data overlap during the evaluation period, so we decided to use our 2022 model also this year.

Lang	Model	AL↓	$AL_{CA}{\downarrow}$	BLEU↑
En-De	2022	1959	2721	31.4
En-Zh	2022	1990	2466	26.3

Table 4: Submitted large offline models onlinized using the proposed CTC policy.

#### 4.2 Blockwise Online Models

Finally, we submit small blockwise models. Their advantage is that they are able to run on a CPU faster than real time (more than  $5 \times$  faster). We report their performance in Table 5.

Lang	AL↓	$AL_{CA}{\downarrow}$	RTF↓	BLEU↑
En-De	1986	2425	0.19	25.4
En-Zh	1999	2386	0.19	23.8

Table 5: Submitted small blockwise models using the proposed CTC online policy.

<sup>&</sup>lt;sup>4</sup>(Zhang and Ao, 2022) found an overlap between ST-TED training corpus and tst-COMMON set of MuST-C dataset.

#### **5** Conclusion and Future Work

In this paper, we present the CUNI-KIT submission to the Simultaneous track at IWSLT 2023. We experimented with the latest decoding methods and proposed a novel CTC online policy. We experimentally showed that the proposed CTC online policy significantly improves the translation quality of the blockwise streaming models. Additionally, the proposed CTC policy significantly lowers the computational footprint of the onlinized large offline models. Unaware of a data overlap issue in 2022, we eventually chose to use our last years' models in the official evaluation also this year.

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