



# Multilingual Speech Representations

*2025*

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# Contributors

Speech / Research teams in Google

## Recent Talks and Special Sessions at Interspeech 2024

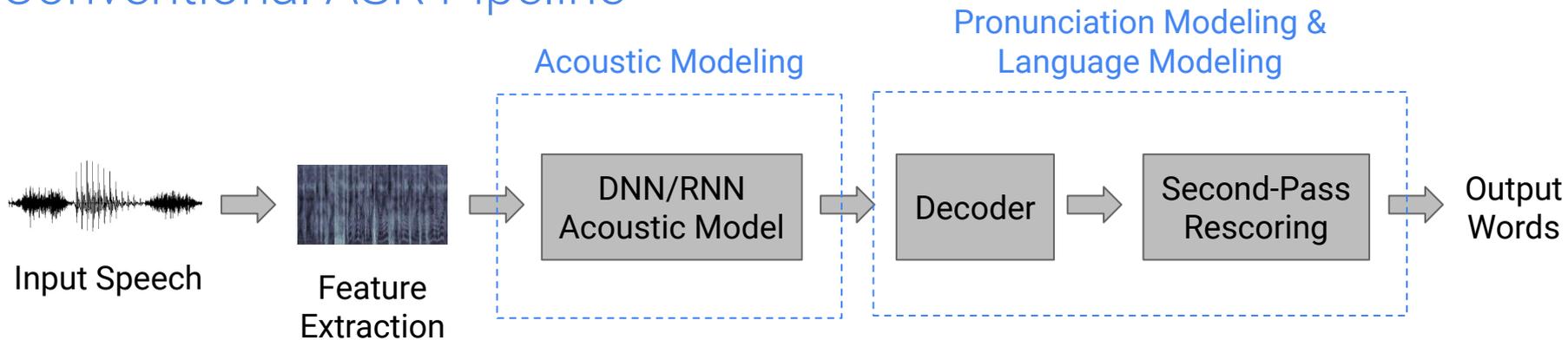
- Hung-yi Lee, “Development of Spoken Language Models“
  - David Harwath, “Visually grounded speech models “
  - **Rohit Prabhavalkar, “Novel architectures for ASR “**
  - Shinji Watanabe, “Toward speech and audio foundation models”
- 
- Speech Processing Using Discrete Speech Units (SS10)
  - Satellite workshop: SynData4GenAI 2024 (Aug. 31, 2024)

... and many more in the conference !

Adapted from Rohit's Interspech 2024 survey talk

Brief Introduction to ASR Architectures

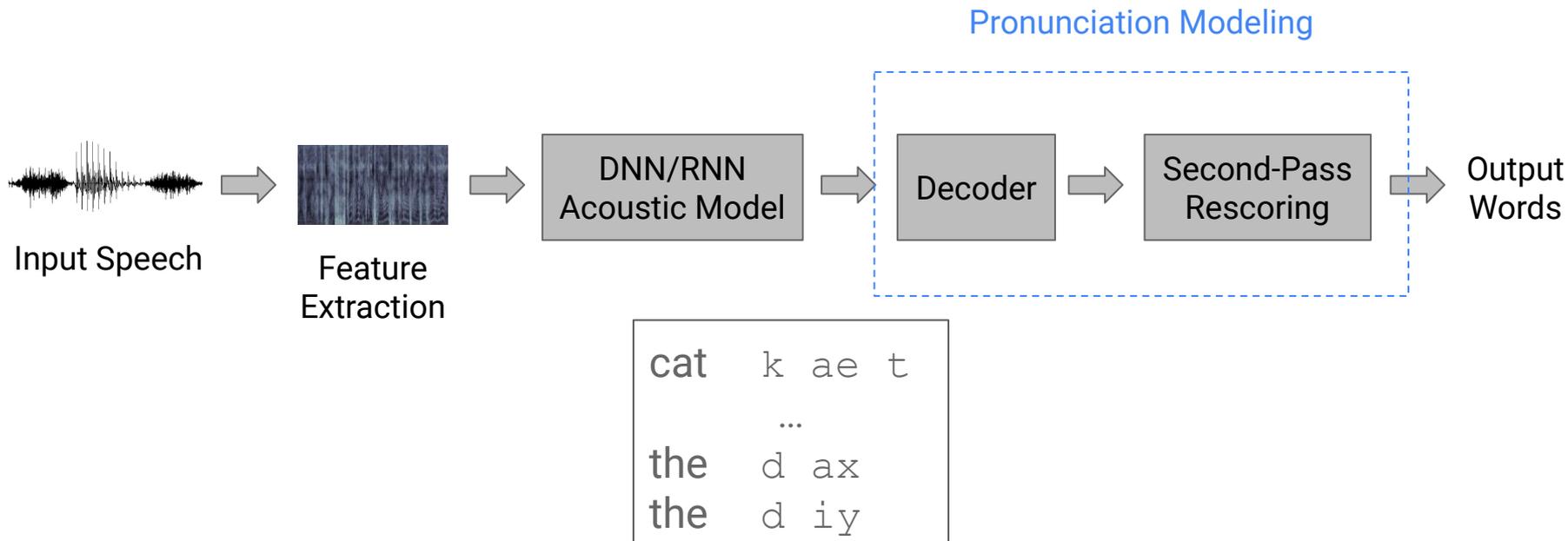
# Conventional ASR Pipeline



e.g., [Rabiner,89] [Mohri+,02]

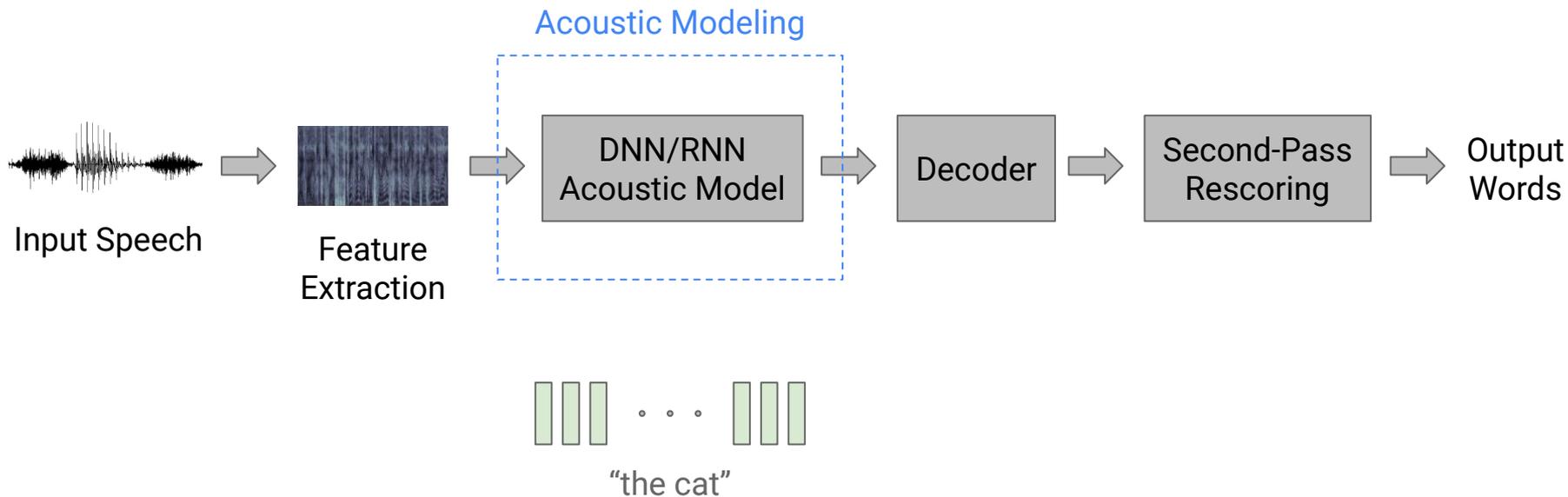
The “hybrid” ASR pipeline breaks the overall problem down into modular tasks: acoustic, pronunciation, and language modeling

# Conventional ASR Pipeline



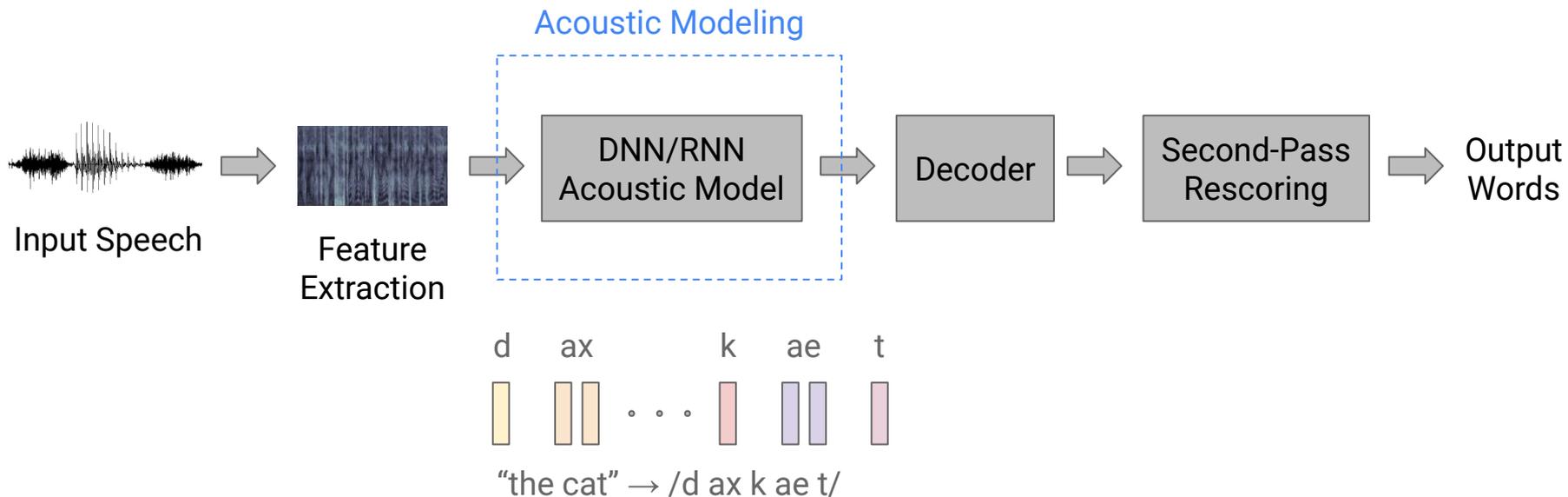
A pronunciation lexicon lists the pronunciation of words in terms of (phonemic) acoustic units; usually done through an expensive manual curation process

# Conventional ASR Pipeline



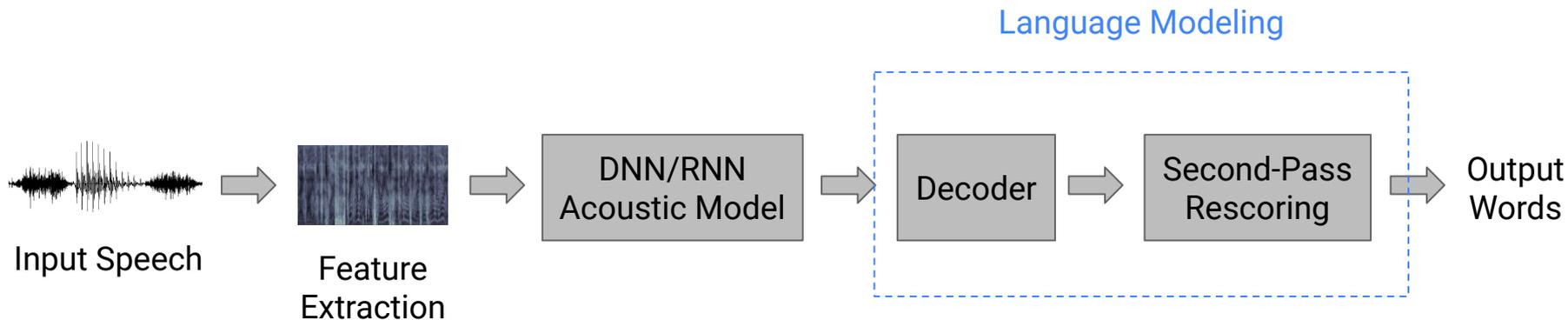
Acoustic modeling learns to associate acoustic feature vectors with the corresponding phonetic acoustic units

# Conventional ASR Pipeline



Acoustic modeling learns to associate acoustic feature vectors with the corresponding phonetic acoustic units

# Conventional ASR Pipeline



the cat sat

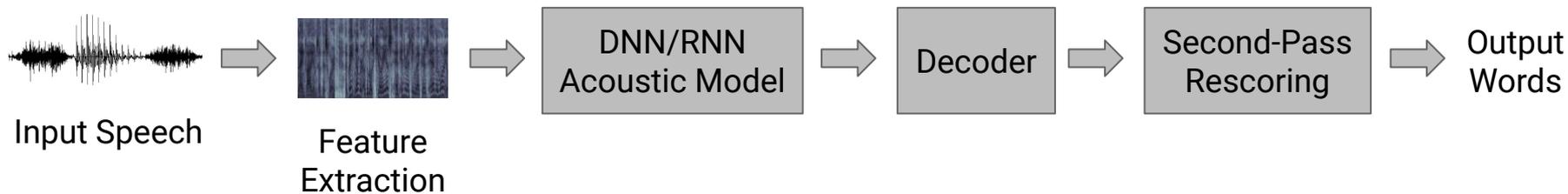


sat cat the



Language modeling assigns probabilities to word sequences, to model prior beliefs of the likelihoods of various sequences

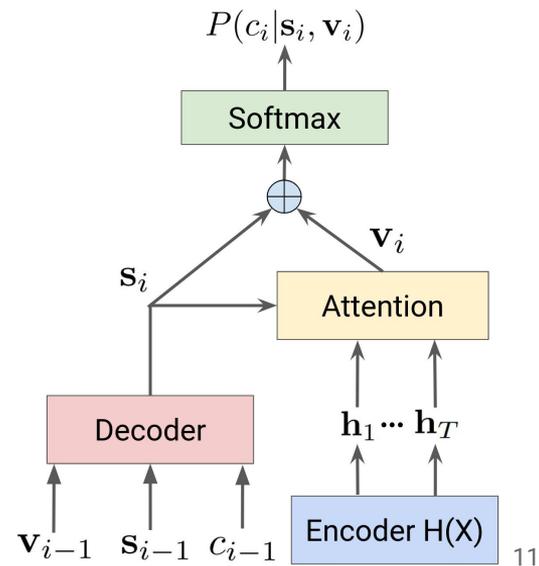
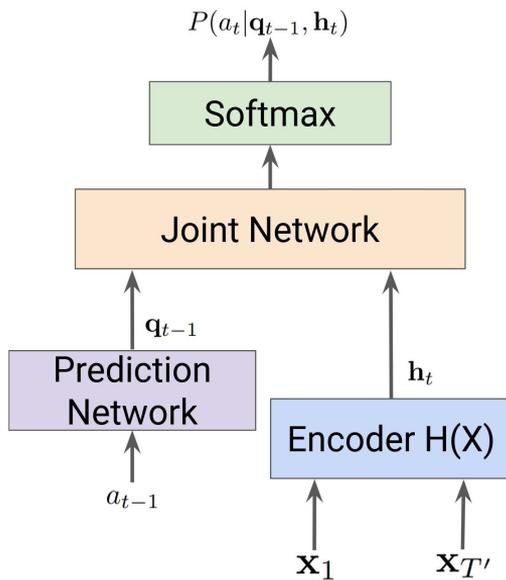
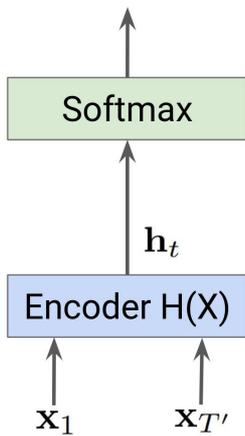
# Conventional ASR Pipeline



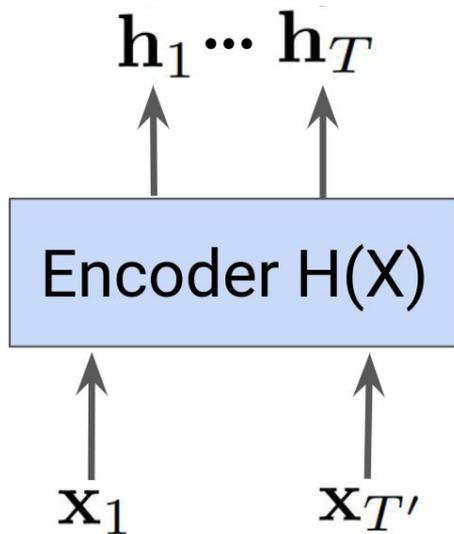
The pronunciation + language model can be represented efficiently in a finite-state transducer, with generic and efficient algorithms to model the search process.  
System is modular and is extensible (e.g., adaptation to new domains)

# E2E Approaches

$$P(a_t = c|X) = P(a_t|\mathbf{h}_t)$$



## Encoders



What is the role of the speech encoder?

Synthesizing context and learning representations

# Representation learning

*Is there a joint latent representation of multiple modalities that can help multilingual speech and language understanding?*

- Speech Understanding (Speech → Text with speaker/language/style annotations)
  - Scaling to many languages
  - Taking advantage of found data
  - Emotional, Medical diagnosis, atypical speech processing, perception
- Audio Generation (Text → Speech)
  - Can we share the same ideas from ASR?
  - Understanding shared representations of multiple modalities key?
- Language Understanding
  - Question Answering, Dialog / Conversations, Summarization, etc.

01

# Representation Learning

Self-supervised Learning allows for the efficient use of unlabeled data (audio, text, image and video) in models with the promise a single universal model that would benefit a wide variety of tasks and domains.

Historically, in speech and language processing, semi-supervised and unsupervised learning has been used extensively (BABEL program)

Cui, Jia, et al. "Multilingual representations for low resource speech recognition and keyword search." 2015 IEEE workshop on automatic speech recognition and understanding (ASRU). IEEE, 2015.

Mohamed, Abdelrahman, Hung-yi Lee, Lasse Borgholt, Jakob D. Havtorn, Joakim Edin, Christian Igel, Katrin Kirchhoff et al. "Self-supervised speech representation learning: A review." *IEEE Journal of Selected Topics in Signal Processing* 16, no. 6 (2022): 1179-1210.

Machine Learning has focussed on learning speech representations for a variety of tasks in an unsupervised fashion and combined with supervised training for maximum results

Diversity of the data used for unsupervised learning plays a key role

# Self Supervision

Model learns discriminating patterns in the data in contrast to providing the model with annotations and asking the model to be sensitive to those specific annotations.

- Cost function
- Adversarial Methods
- Supervised Fine tuning: tailoring learned representations to downstream tasks
- Architectures
- Robustness and transferability of speech representations with few examples
- Injecting other modalities

# Self Supervision: Cost function

- Contrastive Predictive Coding (CPC) [1], Autoregressive Predictive Coding (APC) [2], Triplet Loss based approaches[15], Time Contrastive Networks [16]
  - Encode underlying shared information in the high-dimensional signal
  - Maximize Mutual Information between encoded representations
  - Combine predicting future observations (predictive coding) with a probabilistic contrastive loss (NCE variants)

# Self Supervision: Cost function

- Selecting positive and negative samples
  - Sampling from joint versus marginalized distributions
  - Momentum Contrast (MoCo )in Vision[9] and Speech[17]
- Multiple tasks (views) for a more complete representation
  - Examples: Regressive, classification tasks
  - Multi task learning and self-supervision
    - Task Agnostic Speech Embeddings (PASE, PASE+) [7]

# Self Supervision: Cost function

- Unsupervised Latent Variable Model based data generation [4, 5]
  - Clustering latent representations, Unsupervised Unit Discovery
    - Codebook learning, VQVAE
- wav2vec/ wav2vec2 : Combine contrastive loss and discrete representations [10]
- HuBERT: Iterative learning of discretized representations (k-means clustering) and representation learning [21]
- W2V-BERT [22]: Combines contrastive loss on continuous signal with masked language modeling (MLM) loss on the discretized representations.

# Self Supervision: *Adversarial /Augmentation Methods*

- SimCLR [3]: input augmentation is coupled with a contrastive consistency loss to allow model to learn without any labels.
- Virtual adversarial training(VAT) is a form of model regularization that applies adversarial noise to the model input.
- Masking and Reconstruction
  - Augmentation methods (SpecAugment) to learn invariant representations [6]

# Speech-only Self-supervised Pretrain

[Wav2vec 2.0](#) has shown to be an effective encoder pre-training strategy, especially on [in-domain data](#).

State of the art results when combined with [Conformer encoders](#).

Leverages large amounts of untranscribed data without a good teacher model.

Nevertheless, Cannot take advantage of **unspoken text**.

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Baevski, Alexei, et al. "wav2vec 2.0: A framework for self-supervised learning of speech representations." arXiv preprint arXiv:2006.11477 (2020).

Hsu, Wei-Ning, et al. "Robust wav2vec 2.0: Analyzing Domain Shift in Self-Supervised Pre-Training." arXiv preprint arXiv:2104.01027 (2021).

Zhang, Yu, et al. "Pushing the limits of semi-supervised learning for automatic speech recognition." arXiv preprint arXiv:2010.10504 (2020).

# Self Supervision in Speech and Audio

*Text Injection[24-36]*

- Injecting Text In Self-Supervised Speech Pre-Training
  - Combining supervised and unsupervised loss and TTS
- Joint training of speech and text using text, speech and multimodal encoders

# Self Supervision: Supervised Fine Tuning

- Joint training of encoder and decoders (predictors) as proposed in PASE and PASE+ [7]
- Strategies to freeze different parts of the network during fine tuning on a small amount of labeled data
- Meta Learning: Self supervision and fine-tuning on few samples [18]
- BERT [8] and its variants used in Natural Language Processing tasks
- Foundation Models [23] to capture representations that can be fine-tuned for tasks such as, object recognition, image captioning, information retrieval, etc.

# Self Supervision: Architectures and Transferability

- Encoders: Transformers, VGGs, Conformers, SincNet [19], etc.
- Multilingual representations and bottleneck features from various architectures
- Are the losses used in speech and language transferable to vision or robotics and vice-versa [20] ?

02

# Multilinguality

# Multilingual work dates back to 90s and earlier....

- Multilinguality refers to the ability to handle several languages for different tasks (recognition, translation, synthesis, etc.)
- More recently, training multilingual representations [1, 2] and end-to-end models [3, 4] have demonstrated that the best performing models require conditioning on language information
  - [1] B. Ma, C. Guan, H. Li, and C.-H. Lee, "Multilingual speech recognition with language identification," in ICSLP, 2002.
  - [2] A. Cutler, Y. Zhang, E. Chuangsuwanich, and J.R. Glass, "Language ID-based training of multilingual stacked bottleneck features," in Interspeech, 2014.
  - [3] S. Watanabe, T. Hori, and J.R. Hershey, "Language independent end-to-end architecture for joint language identification and speech recognition," in ASRU, 2017.
  - [4] A. Kannan, A. Datta, T.N. Sainath, E. Weinstein, B. Ramabhadran, Y. Wu, A. Bapna, Z. Chen, and S. Lee, "Large-scale multilingual speech recognition with a streaming end-to-end model," arXiv preprint arXiv:1909.05330, 2019.

## Key requirement

- Need to track language switches within an utterance [5, 6], adjust language sampling ratios, or add additional parameters based on the data distribution [4]
  - [5] H. Seki, S. Watanabe, T. Hori, J. Le Roux, and J.R. Hershey, "An end-to-end language-tracking speech recognizer for mixed-language speech," in ICASSP, 2018.
  - [6] A. Waters, N. Gaur, P. Haghani, P. Moreno, and Z. Qu, "Leveraging language id in multilingual end-to-end speech recognition," in ASRU, 2019.

# What language clusters and why?

For example, many Indic languages can we cover with one model?

- Take advantage of overlap in acoustic and lexical content
  - due to either language family relations or the geographic and cultural proximity of the native speakers.
- However, their writing systems occupy different unicode blocks
- Can we combine languages from multiple languages families efficiently and produce “usable” models for users?
- Can the representations derived by these models help with building models for “unseen” languages?

...what challenge does this pose?

# Challenges: Code-Switching

- Code-switching is a commonly occurring phenomenon in many multilingual communities, wherein a speaker switches between languages within a single utterance (Hindi-English, Bengali-English, Arabic-English and Chinese-English, Spanish-English, etc.)
- Can occur at morphological, lexical, syntactic, semantic, pragmatic levels
- A good read on Bilingual Speech from a linguistic perspective:
  - Analysis of many language-pairs
  - Bilingual verbs: the phenomenon of verbal compounds combining elements from two languages
  - Impact of psycholinguistic and social factors : language dominance, duration of contact, bilingual proficiency, speaker type, age-group or generation and language attitudes.

Pieter Muysken, Bilingual speech: A typology of code-mixing. Cambridge: Cambridge University Press, 2000.

# Examples of code-switching

- Words with different language indices are inserted into a phrase structure
- Spanish-English
  - Cuando mi novio *tweetea* pero no contesta (When my boyfriend tweets but doesn't answer)
  - Agarrar *my Master's* (Get my Master's)
- Ambiguities in transcription
  - डिस्कवरी vs *discovery*
  - होम्योपथी में अर्थराइटिस *treatment* vs *Homeopathy में arthritis treatment*
- These *rendering* errors artificially inflate the **Word Error Rate** (WER)
- Harder to differentiate between ***modeling*** and ***rendering*** errors
  - fancy साड़ी दिखाइए vs fancy *Sadi dikhaiye*

# Handling Code-Switching

- Handled the problem of foreign word pronunciation using language dependent phonemes by creating linguistically motivated pairwise mappings for each language involved in code-switching.

White, Christopher M., Sanjeev Khudanpur, and James K. Baker. "An investigation of acoustic models for multilingual code-switching." *Ninth Annual Conference of the International Speech Communication Association*. 2008.

- In Mandarin-English use of combined subwords from both languages as modeling units along with an additional objective of training with language ID was found to be useful.

Luo, Ne, et al. "Towards end-to-end code-switching speech recognition." *arXiv preprint arXiv:1810.13091* (2018).

# Handling Code-Switching

- Separately train an E2E CTC model and a frame-level language identification (LID) model. Linearly adjust the posteriors of an E2E CTC model using the LID scores (Mandarin-English)

Li, Ke, et al. "Towards code-switching ASR for end-to-end CTC models." ICASSP 2019-2019 IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP). IEEE, 2019.

- Effectiveness of multilingual models on NLU tasks such as named entity recognition and part-of-speech tagging tasks (Hindi-English, Spanish-English, and Modern Standard Arabic-Egyptian)? Pretrained multilingual models not as effective as hierarchical embeddings to deal with code-switching

White, Christopher M., Sanjeev Khudanpur, and James K. Baker. "An investigation of acoustic models for multilingual code-switching." Ninth Annual Conference of the International Speech Communication Association. 2008.

# Handling Code-Switching

- In Frisian-Dutch merging phones of both languages provides the best recognition performance for code-switched words

Yilmaz, Emre, Henk van den Heuvel, and David Van Leeuwen. "Investigating bilingual deep neural networks for automatic recognition of code-switching frisian speech." *Procedia Computer Science* 81 (2016): 159-166.

- Data Augmentation by generating synthetic code-switched data with word translation or word insertion followed by audio splicing using text-to-speech

Du, Chenpeng, et al. "Data augmentation for end-to-end code-switching speech recognition." *2021 IEEE Spoken Language Technology Workshop (SLT)*. IEEE, 2021.

# Handling Code-Switching

Output token embeddings of two monolingual languages are differently distributed;  
Constrain with Jensen-Shannon divergence to force embeddings of monolingual languages to possess similar distributions

Khassanov, Yerbolat, et al. "Constrained output embeddings for end-to-end code-switching speech recognition with only monolingual data." *arXiv preprint arXiv:1904.03802* (2019).

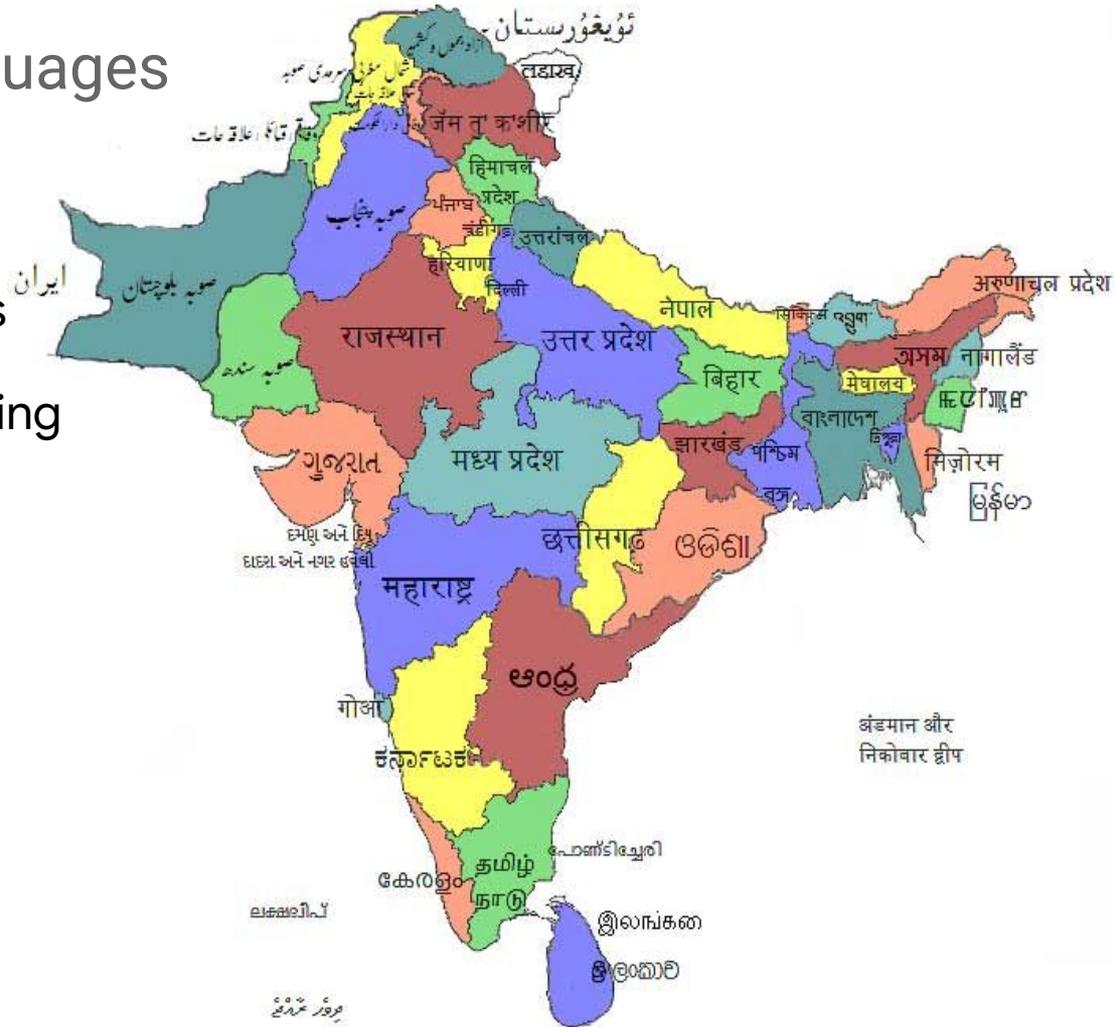
## What are these techniques forcing the model to learn?

Joint multilingual representations (embeddings) at multiple levels?

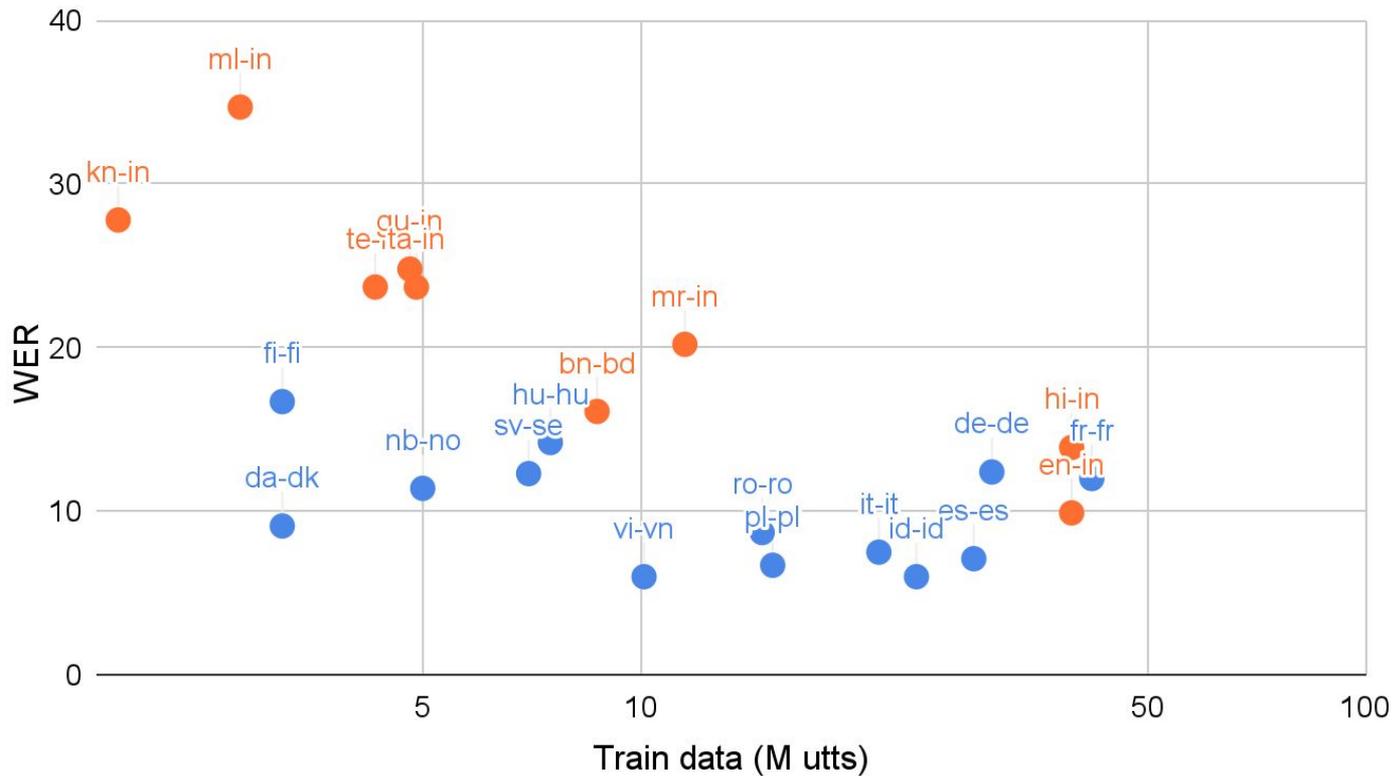
Learn implicit patterns in the data across languages?

# South Asia: A land of languages

- **Scripts:** Several writing systems
- **Code-switching:** language mixing
- **Several** languages and dialects!
- But **overlap** due to linguistic similarity, and/or geographic & cultural proximity of the native speakers.



# Data Distribution across languages



# Pooling Language Resources to learn representations

- State-of-the-art
  - Allow for joint training of data-rich and data-scarce languages in a single model
  - Require the encoding of language information which makes it less flexible
- Challenges in building a language-agnostic multilingual ASR system?
  - Can similar sounding acoustics across languages be mapped to a single, canonical target sequence of graphemes or sub-word units?

Add maestro paper

# Data Normalization: Challenge in multilingual transliteration

Attested romanizations of the English word “discovery”

Bengali	Hindi	Kannada	Tamil
ডিসকভারি	डिस्कवरी	ಡಿಸ್ಕವರಿ	டிஸ்கவரி
discoveri	discovery	discoverary	tiskavari
discovery		discovery	discovery
diskovary		discoveri	
diskovery		discowery	
diskoveri			

# Code-Switching Benchmark: For NLP research (<https://ritual.uh.edu/lince/>)

**LinCE** is a continuous effort, and we will expand it with more low-resource languages and tasks.

Language Pairs	LID	POS	NER	SA	MT
Spanish-English	✓	✓	✓	✓	
Hindi-English	✓	✓	✓		
Nepali-English	✓				
Modern Standard Arabic-Egyptian Arabic	✓		✓		
English-Hinglish					✓
Spanglish-English					✓
English-Spanglish					✓
(Modern Standard Arabic-Egyptian Arabic)-English					✓
English-(Modern Standard Arabic-Egyptian Arabic)					✓

# Text Representation Benchmark: (<https://huggingface.co/spaces/mteb/leaderboard>)

Massive Text Embedding Benchmark (MTEB) Leaderboard. To submit, refer to the [MTEB GitHub repository](#) 🗄️ Refer to the [MTEB paper](#) for details on metrics, tasks and models. Also check out [MTEB Arena](#) ✕

Search Bar (separate multiple queries with `;`)

Model types

Open  Proprietary  Sentence Transformers  Cross-Encoders

Bi-Encoders  Uses Instructions  No Instructions

Model sizes (in number of parameters)

<100M  100M to 250M  250M to 500M

500M to 1B  >1B

Overall Bitext Mining Classification Clustering Pair Classification Reranking Retrieval STS Summarization MultilabelClassification

Retrieval w/Instructions

English Chinese French Polish Russian

## Overall MTEB English leaderboard 🏆

- Metric: Various, refer to task tabs
- Languages: English

Rank	Model	Model Size (Million Parameters)	Memory Usage (GB, fp32)	Embedding Dimensions	Max Tokens	Average (56 datasets)	Classification Average (12 datasets)	Clustering Average (11 datasets)
1	<a href="#">NV-Embed-v2</a>	7851	29.25	4096	32768	72.31	90.37	58.46
2	<a href="#">bge-en-ic1</a>	7111	26.49	4096	32768	71.67	88.95	57.89
3	<a href="#">stella_en_1.5B_v5</a>	1543	5.75	8192	131072	71.19	87.63	57.69
4	<a href="#">SFR-Embedding-2_R</a>	7111	26.49	4096	32768	70.31	89.05	56.17
5	<a href="#">gte-Qwen2-7B-instruct</a>	7613	28.36	3584	131072	70.24	86.58	56.92
6	<a href="#">dunzhang-stella_en_400M_v5</a>	435	1.62	1024	8192	70.11	86.67	56.7
7	<a href="#">stella_en_400M_v5</a>	435	1.62	8192	8192	70.11	86.67	56.7
8	<a href="#">bge-multilingual-gemma2</a>	9242	34.43	3584	8192	69.88	88.08	54.65

Towards the future....

Can we have a similar code-switching only benchmark for speech across hundreds of languages ?

What tasks and associated metrics would help advance state-of-the-art?

Can these multilingual representations now be extended to do several tasks in one model? More than ASR?

03

# Multitask and multilingual representations

# State-of-the-art performance in ASR and ST tasks

- Efficient Pre-training
- Incorporating Untranscribed Speech, Unspoken Text, Paired Speech-Text
- Modality matching for in the Injection of unspoken text
- Language-ID
- Code-Switching

Bharadwaj, S., Ma, M., Vashishth, S., Bapna, A., Ganapathy, S., Axelrod, V., Dalmia, S., Han, W., Zhang, Y., van Esch, D. and Ritchie, S., 2023. Multimodal Modeling For Spoken Language Identification. arXiv preprint arXiv:2309.10567.

# Google Universal Speech Model for 100+ Languages [43]

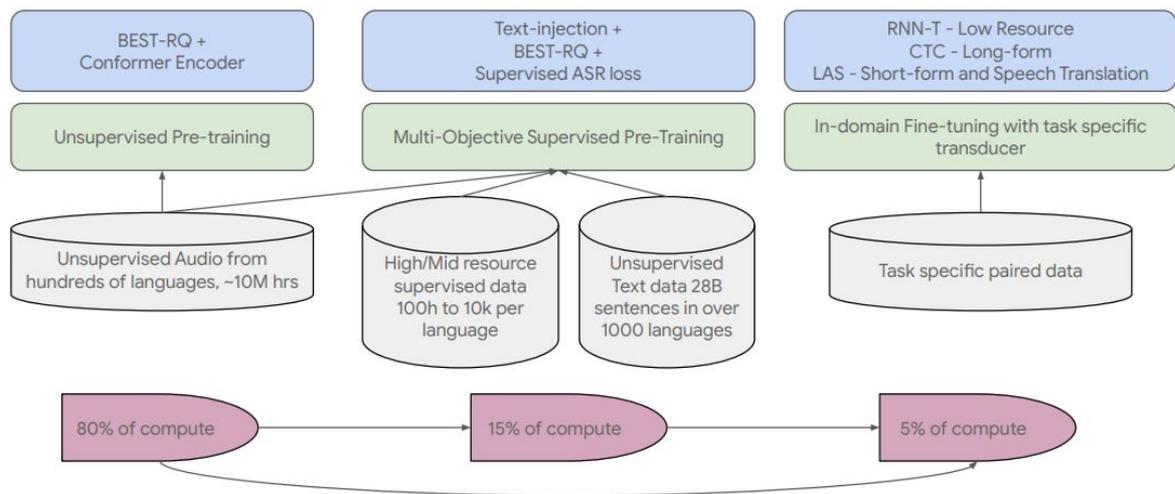


Figure 1: An overview of our approach. Training is split into three stages. (i) The first stage trains a conformer backbone on a large unlabeled speech dataset, optimizing for the BEST-RQ objective. (ii) We continue training this speech representation learning model while optimizing for multiple objectives, the BEST-RQ objective on unlabeled speech, the modality matching, supervised ASR and duration modeling losses on paired speech and transcript data and the text reconstruction objective with an RNN-T decoder on unlabeled text. (iii) The third stage fine-tunes this pre-trained encoder on the ASR or AST tasks.

# Representations learnt during pre-training

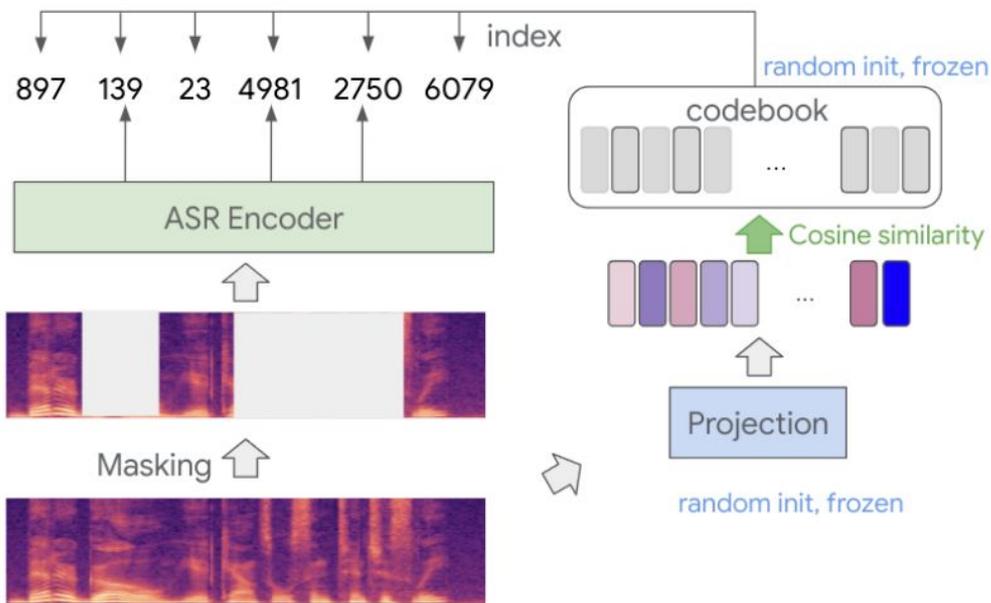


Figure 3: BEST-RQ based pre-training with conformer encoder.

BEST-RQ  
(BERT-based  
Speech pre-Training  
with Random  
projection  
Quantizer) is used  
to pre-train the  
encoder of the  
conformer model

# Text-Injection and modality matching

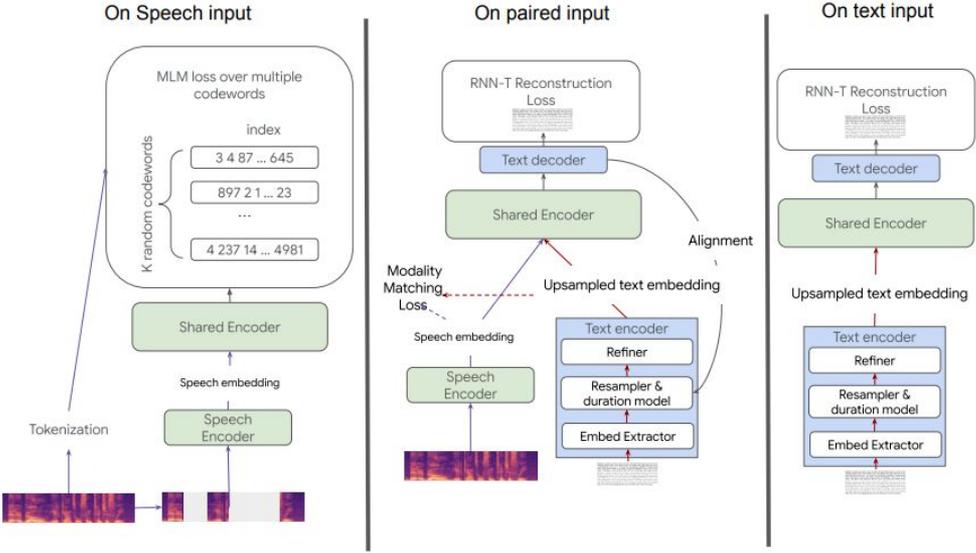


Figure 5: Overview of MOST text injection. The left-most panel depicts MOST training on unlabeled speech input; the center panel depicts training on paired speech and text input; the right-most panel depicts training on unlabeled text data.

# Key Findings

- BEST-RQ is a scalable speech representation learner: We find that BEST-RQ pre-training can effectively scale to the very large data regime with a 2B parameter Conformer-based backbone.
- MOST (BEST-RQ + text-injection) is a scalable speech and text representation learner: It is an effective method for utilizing large scale text data for improving quality on downstream speech tasks, as demonstrated by quality gains exhibited for the FLEURS and CoVoST 2 tasks.
- Representations from MOST (BEST-RQ + text-injection) can quickly adapt to new domains with light-weight residual adapters.
- SoTA results for downstream multilingual speech tasks:
  - SpeechStew (mono-lingual ASR)
  - CORAAL (African American Vernacular English (AAVE) ASR)
  - FLEURS (multi-lingual ASR) [16], YT (multilingual long-form ASR)
  - CoVoST (AST from English to multiple languages).

# Scalability: Language Expansion Results

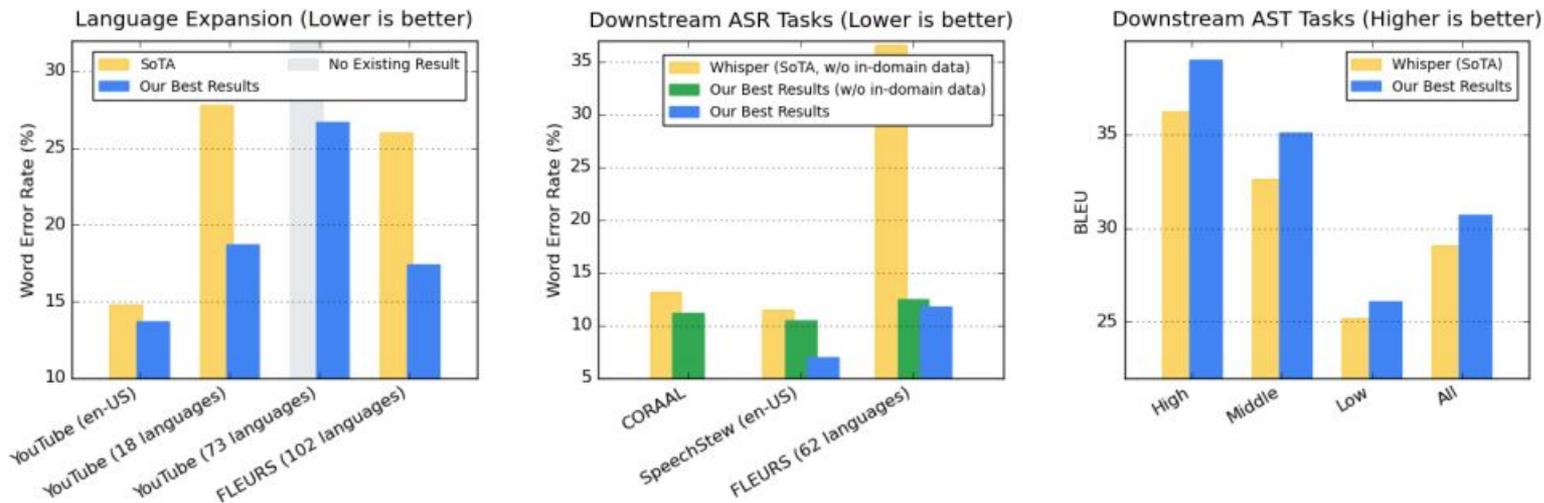


Figure 2: **(Left)**<sup>†</sup> WERs (%) Our language expansion effort to support more languages on YouTube (73 languages) and extending to 100+ languages on the public dataset (FLEURS). Lower is better. To the best of our knowledge, no published model can successfully decode all 73 languages from our YouTube set, thus we only list our results. **(Middle)**<sup>†</sup> Our results on ASR benchmarks, with or without in-domain data. Lower is better. **(Right)** SoTA results on public speech translation tasks. Results presented are presented as high/middle/low resources languages defined in [20]. Higher is better.

# USM Results across ASR and ST tasks

Task	Multilingual Long-form ASR			Multidomain en-US	Multilingual ASR		AST	
Dataset	YouTube		CORAAL	SpeechStew	FLEURS		CoVoST 2	
Languages	en-US	18	73	en-US	en-US	62	102	21
<b>Prior Work (single model)</b>								
Whisper-longform	17.7	27.8	-	23.9	12.8			
Whisper-shortform <sup>†</sup>	-	-	-	13.2 <sup>‡</sup>	11.5	36.6	-	29.1
<b>Our Work (single model)</b>								
USM-LAS	14.4	19.0	29.8	<b>11.2</b>	<b>10.5</b>	<b>12.5</b>	-	-
USM-CTC	<b>13.7</b>	<b>18.7</b>	<b>26.7</b>	12.1	10.8	15.5	-	-
<b>Prior Work (in-domain fine-tuning)</b>								
BigSSL [3]	14.8	-	-	-	7.5	-	-	-
Maestro [67]					7.2			25.2
Maestro-U [67]							26.0 (8.7)	
<b>Our Work (in-domain fine-tuning)</b>								
USM	13.2	-	-	-	7.4	13.5	19.2 (6.9)	28.7
USM-M	<b>12.5</b>	-	-	-	<b>7.0</b>	<b>11.8</b>	<b>17.4 (6.5)</b>	<b>30.7</b>
<b>Our Work (frozen encoder)</b>								
USM-M-adapt <sup>§</sup>	-	-	-	-	7.5	12.4	17.6 (6.7)	29.6

# Speech-text representation learning

- **Complementary information** contained in Text and Speech<sup>1</sup>
  - **text**: domain; **speech**: acoustic conditions, speakers, etc.
- **Unify** speech and text representations
  - Simplify learning from both modalities
  - Learn better linguistic context in (conformer) encoders
- **Data minimization** by incorporating **unspoken text**
  - Low-resource speech processing

# Cross-modality and Cross-lingual Knowledge Transfer

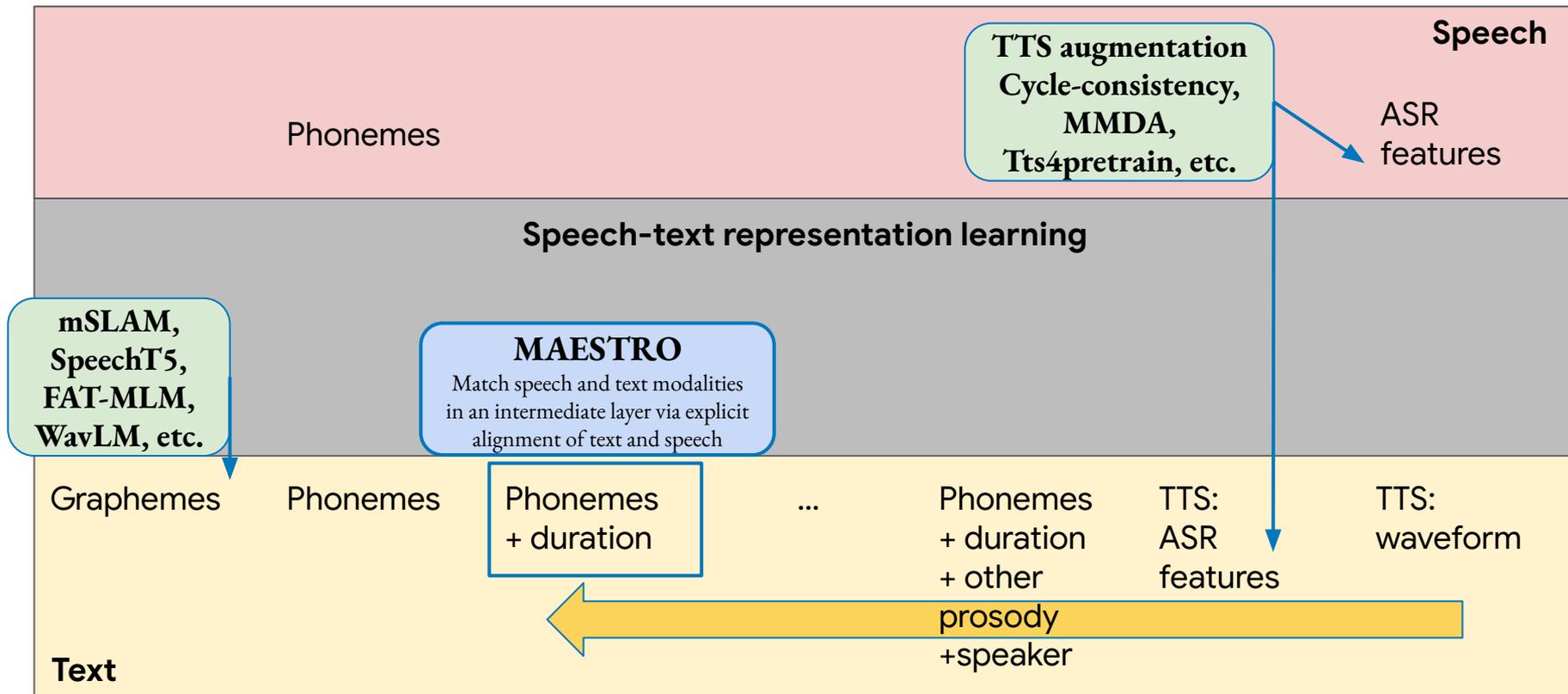
- **Maestro-U (ASR with zero-transcribed speech)**

- Modality specific encoders feed a shared encoder.
- Language specific adapters in the shared encoder.
- Labeled speech for some languages
- Only unpaired speech and unpaired text for some languages
- NO LEXICON or G2P - Unicode Byte inputs support performance even on unseen scripts

- **Virtuoso (TTS with zero-transcribed speech)**

- Similar approach but applied to TTS
- Speech decoder (feature to spectrogram) doesn't see any transcribed audio.
- NO LEXICON or G2P - graphemic to acoustic form can be learned directly without explicit intermediate phone labels

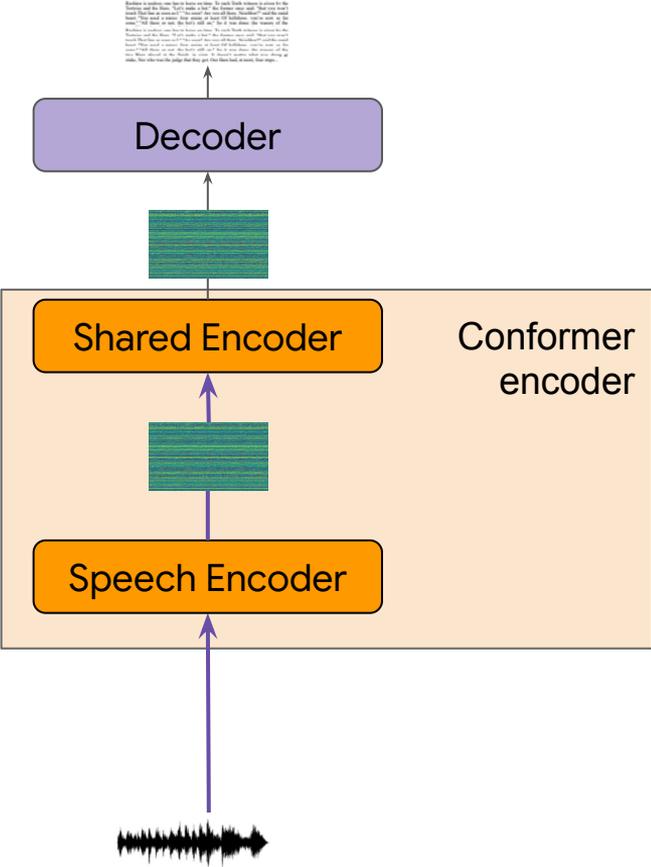
# Joint speech+text representations



More related works can be referred to the paper.

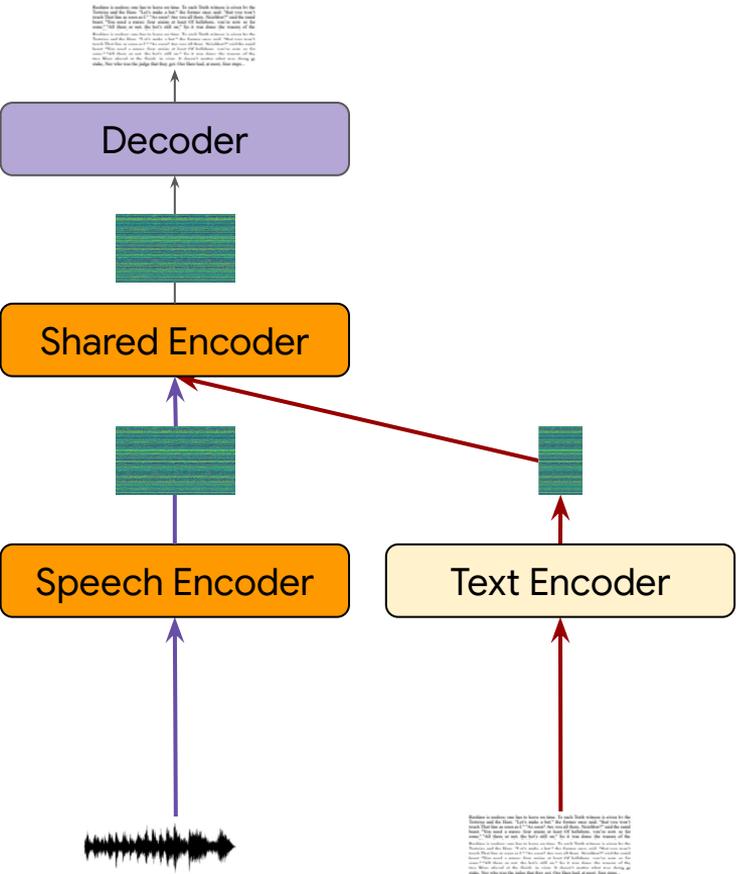
# Architecture

Split original Encoder into two



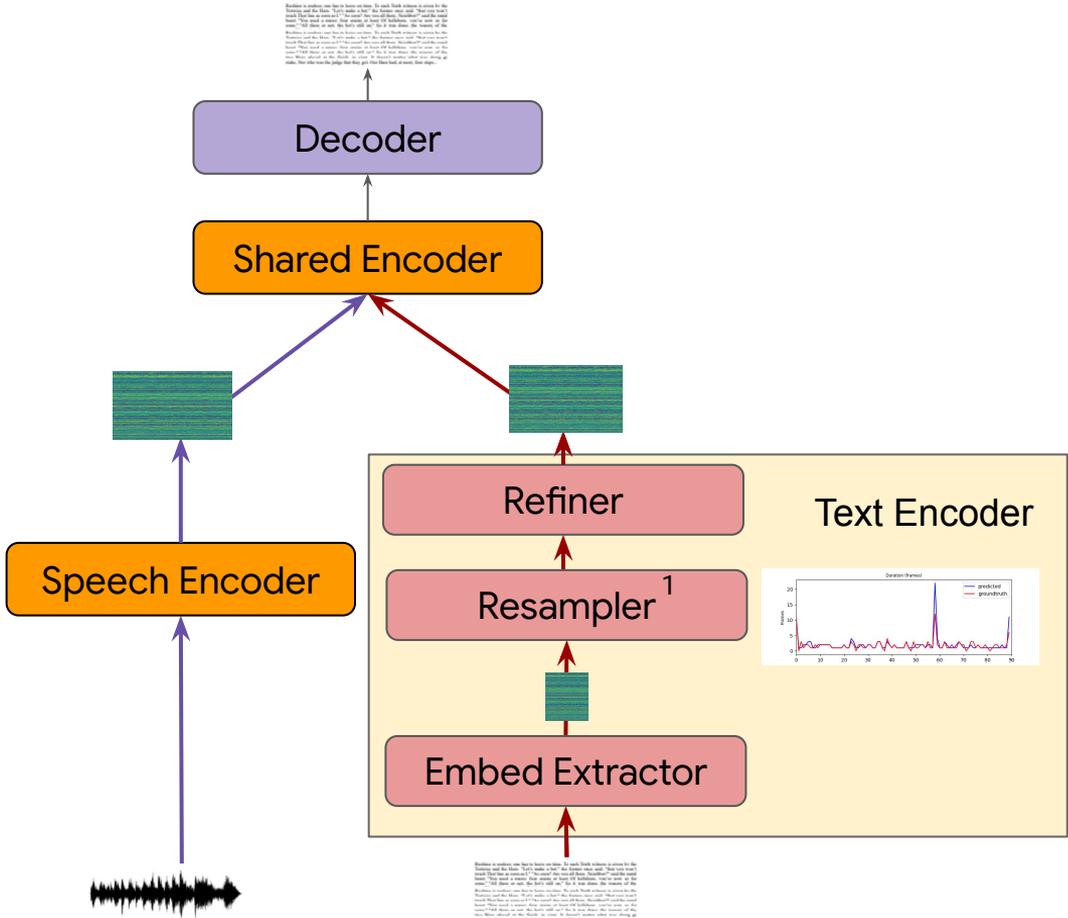
# Architecture

Inject text representations in the middle



# Architecture

## How to match the two modalities?

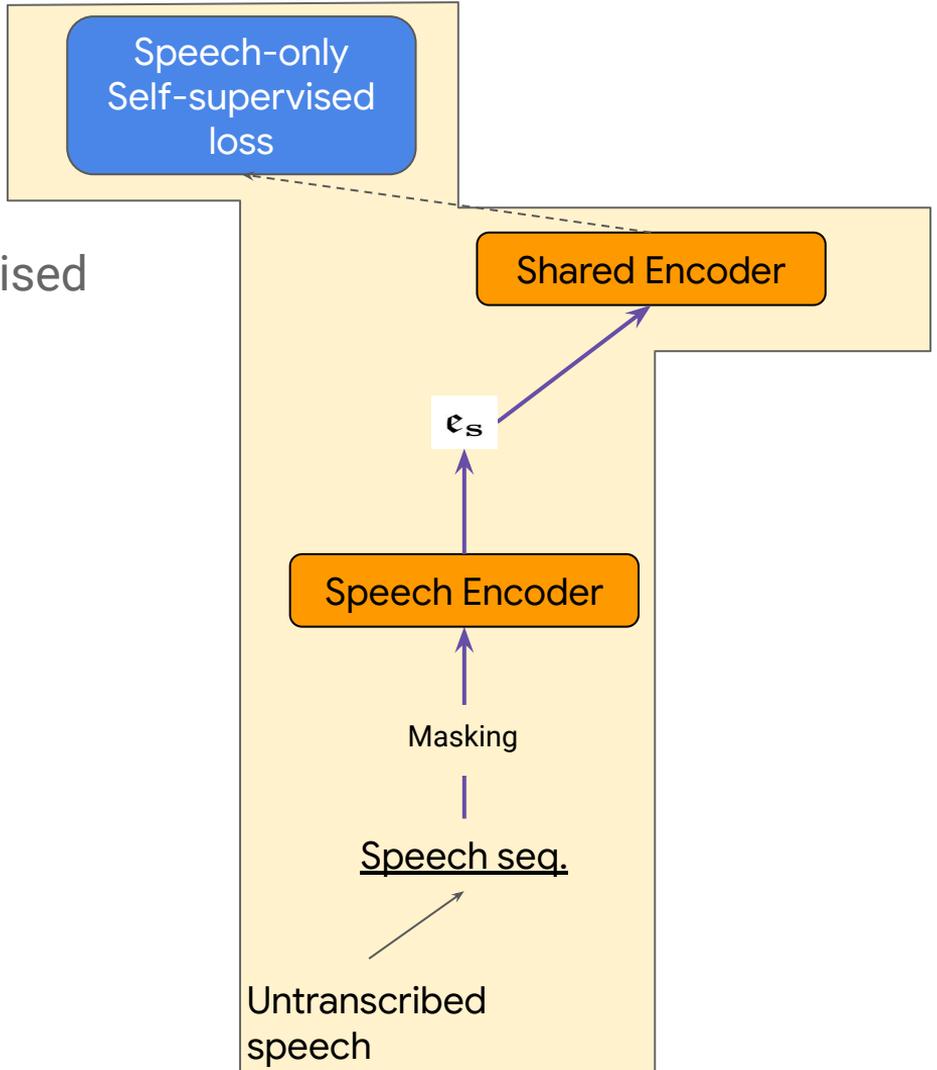


<sup>1</sup>Elias, Isaac, et al. "Parallel tacotron: Non-autoregressive and controllable tts." 2021.

# Loss breakdown: Speech-only

Reuse any self-supervised pretraining objective

- W2v-BERT
- Best-RQ
- w2v1



# Loss breakdown: Paired Speech

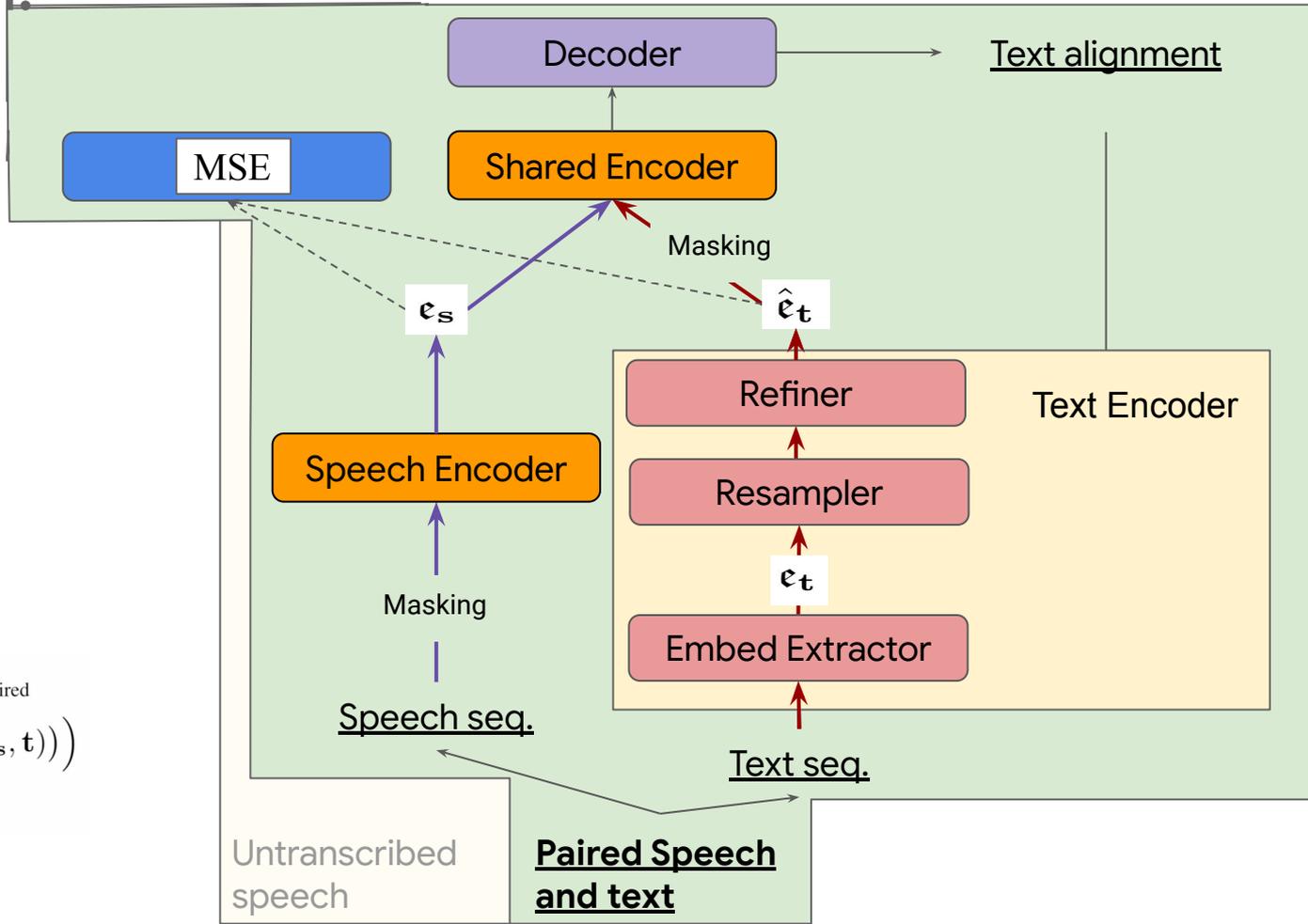
Train with  $\mathcal{L}_{MM}$  :

1. Align
2. Resample
3. Refine

$$\mathbf{e}_s = \theta_s(\mathbf{s}), \quad \mathbf{e}_t = \theta_t(\mathbf{t}), \quad (\mathbf{t}, \mathbf{s}) \in \mathcal{X}_{\text{paired}}$$

$$\hat{\mathbf{e}}_t = \theta_{\text{Refiner}}\left(\text{Resample}\left(\mathbf{e}_t, \text{Align}_{\text{Rnnt}}(\mathbf{e}_s, \mathbf{t})\right)\right)$$

$$\mathcal{L}_{MM} = \text{MSE}(\mathbf{e}_s, \hat{\mathbf{e}}_t) + \mathcal{L}_{\text{Rnnt}}(\mathbf{t} | \mathbf{e}_s)$$



# Loss breakdown: Text-only

Inference using  
**Text Encoder:**

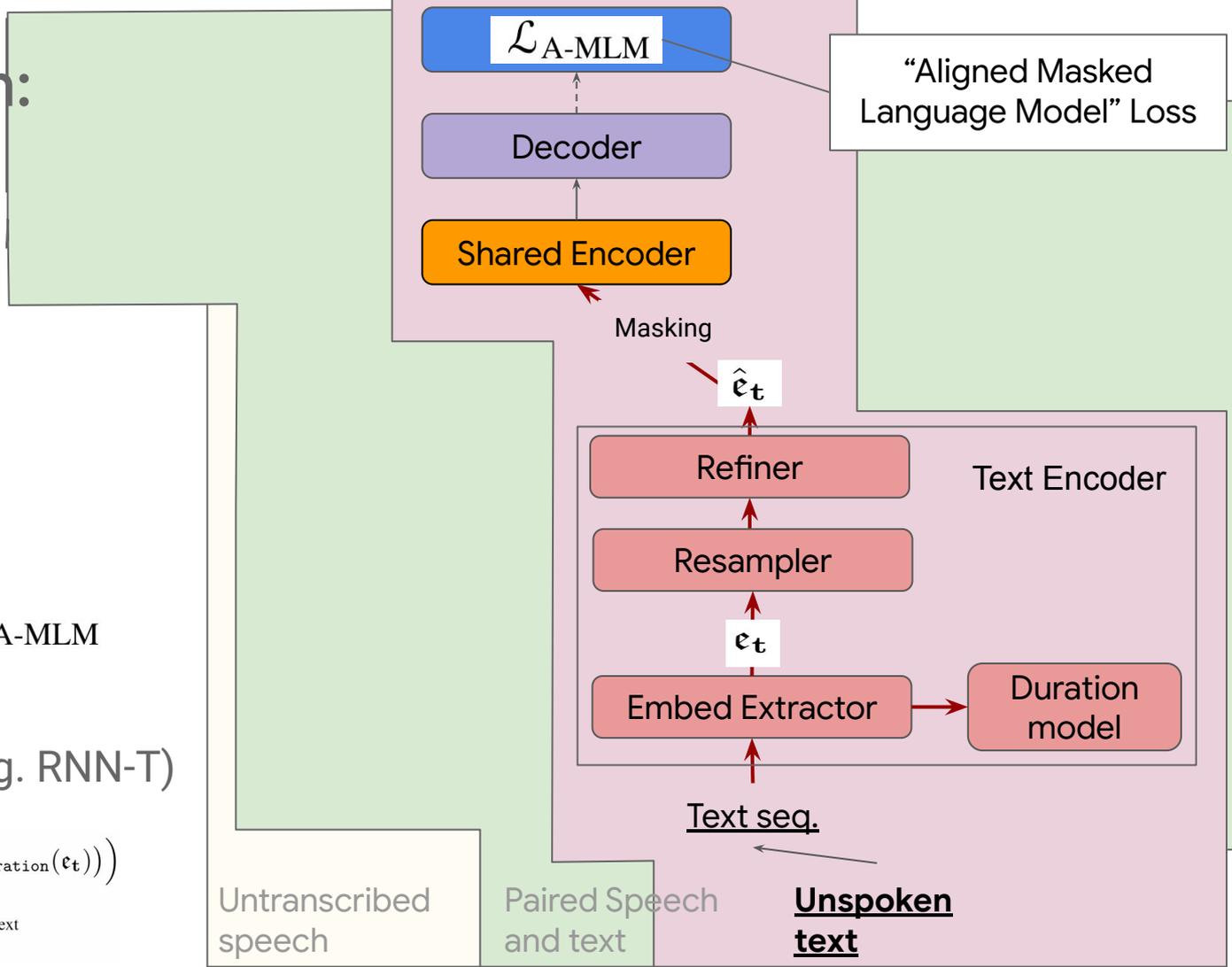
1. **Predict Duration**
2. Resample
3. Refine

Text learning with  $\mathcal{L}_{A-MLM}$

1. Mask
2. Decoder loss (e.g. RNN-T)

$$\mathbf{e}_t = \theta_t(\mathbf{t}), \hat{\mathbf{e}}_t = \theta_{\text{Refiner}}(\text{Resample}(\mathbf{e}_t, \theta_{\text{Duration}}(\mathbf{e}_t)))$$

$$\mathcal{L}_{A-MLM} = \mathcal{L}_{\text{Rnnt}}(\mathbf{t} \mid \text{Mask}(\hat{\mathbf{e}}_t)), \quad \mathbf{t} \in \mathcal{X}_{\text{text}}$$



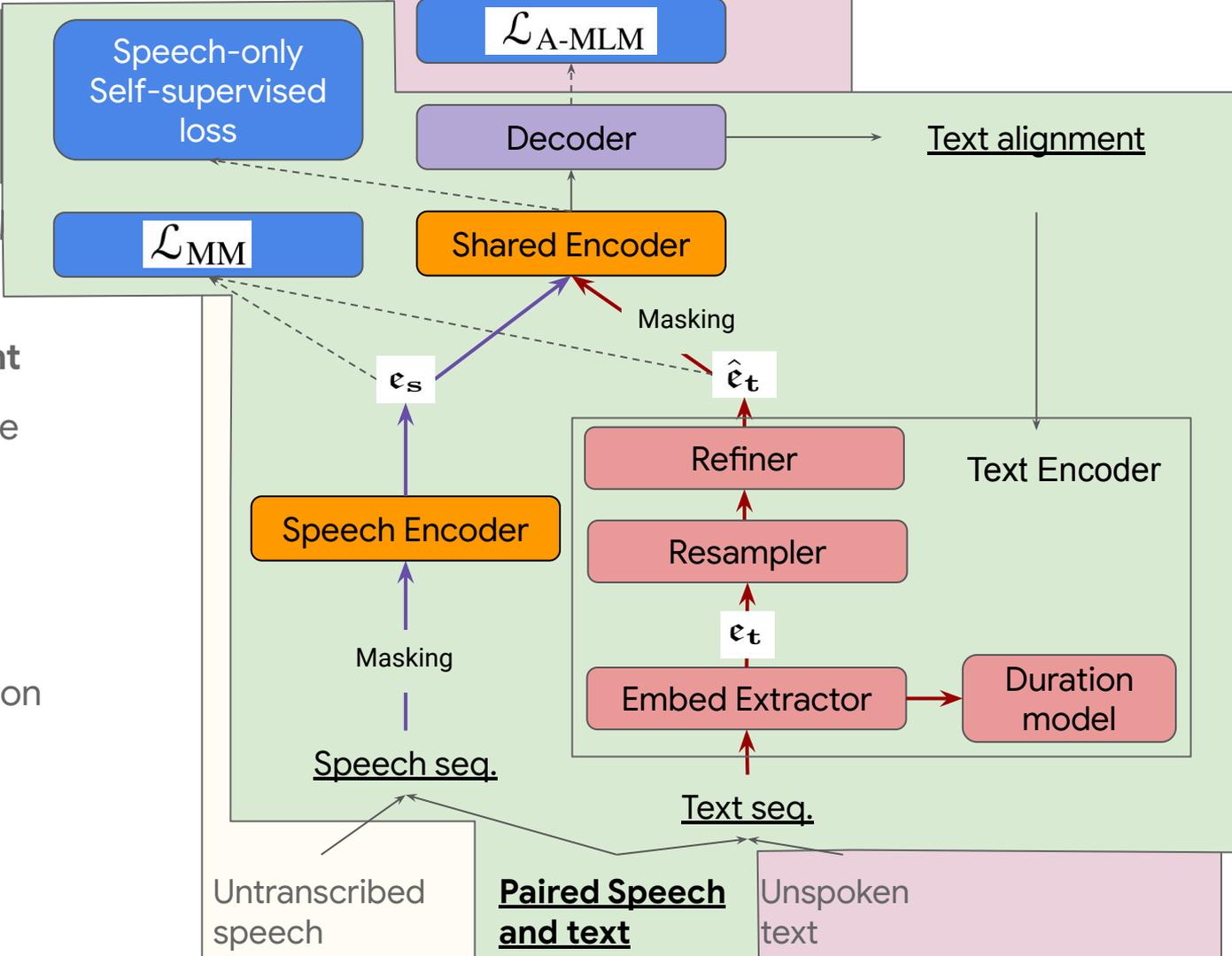
# Overview

Sequence **self-alignment**

**Modality matching** in the intermediate layer

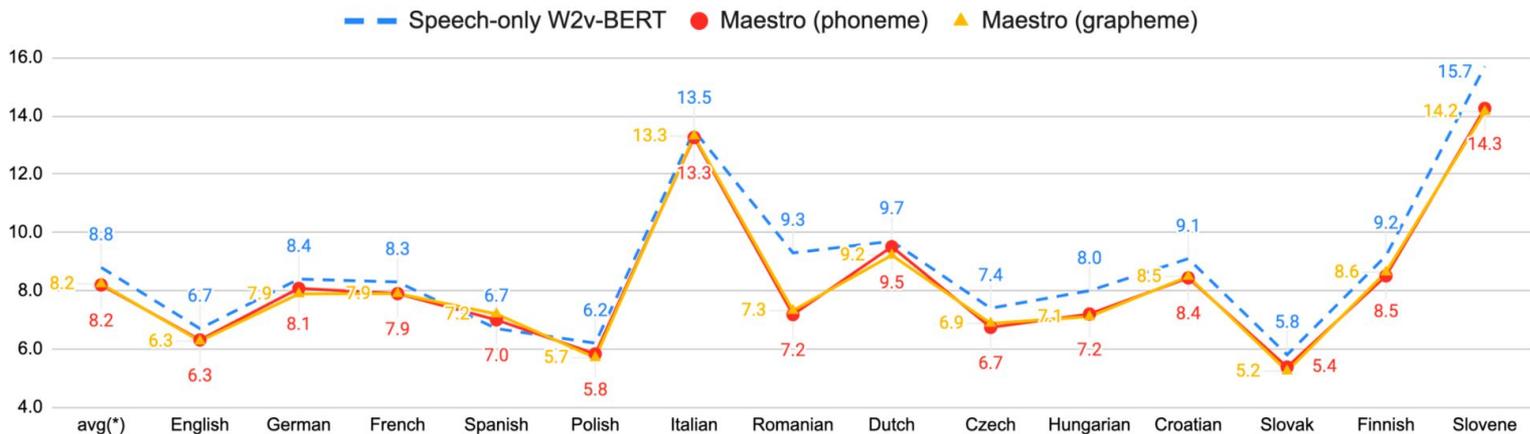
Reuse **duration** part of Parallel Tacotron

**Unified** framework for text-speech representation learning



# Multilingual ASR: Voxpopuli (14 languages)

Breakdown: Languages are sorted by the amount of paired data



Generalize to different amount of paired data  
No substantial difference from Phonemic and Graphemic modeling

Does this joint representation learning work on other tasks?

## Speech-to-text Translation (ST, 21 languages->en)

Method	Model size	Pretraining Data					Avg BLEU
		Speech	Text	ASR	ST	MT	
Finetune: ST-only; mBART decoder init							
XLS-R	1B	437k	-	-	✗	✗	19.3
XLS-R	2B	437k	-	-	✗	✗	22.1
Finetune: ST and Machine translation (MT) jointly							
w2v-bert	0.6B	429k	-	-	✗	✗	21.0
mSLAM	0.6B	429k	mC4	2.4k	✗	✗	22.4
mSLAM	2B	429k	mC4	2.4k	✗	✗	24.8
Maestro	0.6B	429k	VP-T + mC4	2.4k	✗	✗	24.3
<b>Maestro</b>	<b>0.6B</b>	<b>429k</b>	<b>VP-T + mC4</b>	<b>2.4k</b>	✓	✓	<b>25.2</b>

Numbers other than Maestro from "mSLAM: Massively multilingual joint pre-training for speech and text." [link](#).

Strong performance across  
ASR and Translation tasks

## Key Finding:

Learn unified **speech-text** representations simultaneously that can transfer to diverse tasks

## Solution: **Maestro**

- **Match speech and text modalities** in an intermediate layer via **explicit alignment of text and speech**

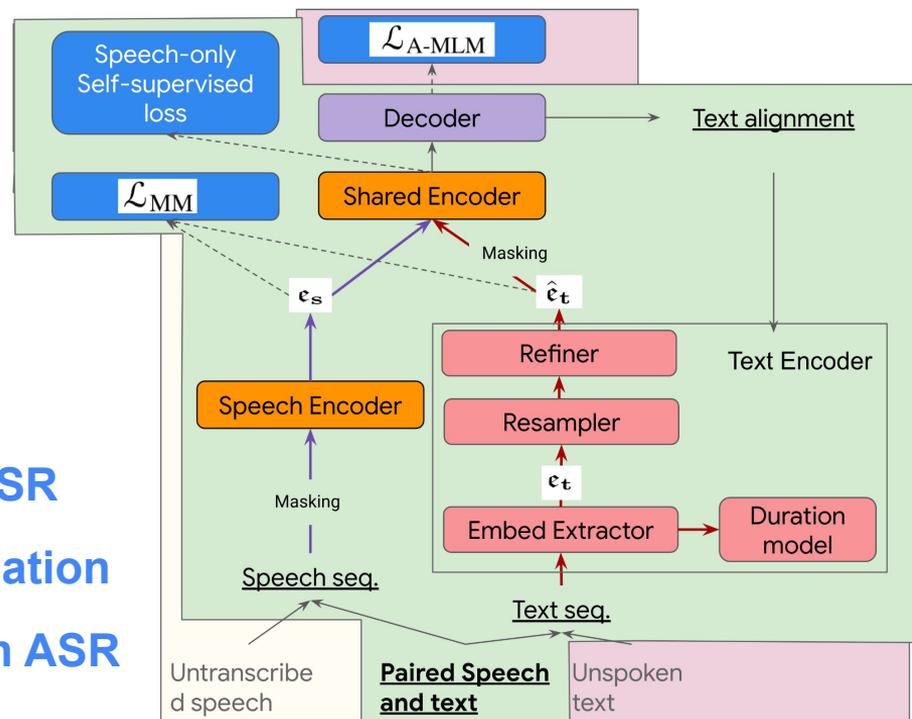
- Sequence alignment
- Matching modality embeddings
- Duration prediction
- Aligned masked-language model loss

**Result:** create new SOTAs

**8% WER reduction** on VoxPopuli **multilingual ASR**

**2.8 BLEU improve** on CoVoST 2 **Speech Translation**

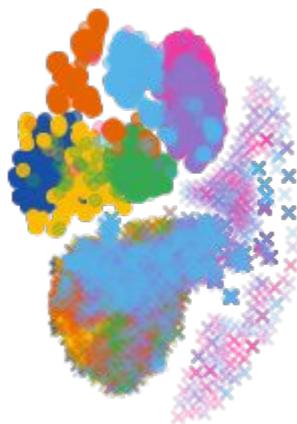
**4% WER reduction** on SpeechStew **multidomain ASR**



# Retrieval to measure Shared Representation (ICASSP 2023)

**Task:** Given a speech sample, find the matching text sample or vice versa

unimodal encoders



shared encoder



Librispeech retrieval performance  
test-clean: 20.5%  
test-other: 19.3%

CV retrieval performance: 7.4%

Librispeech retrieval performance  
test-clean: 83.5%  
test-other: 68.8%

CV retrieval performance: 28.8%



Chance: 0.1%      Other models at ~1-2%.  
LibriSpeech trained encoders

Inspired by <https://arxiv.org/abs/2209.15430> && <https://arxiv.org/abs/2210.01738>

## Goal:

Train ASR **without transcribed speech** and **G2P**

Enable **multilingual transfer** even with unseen writing systems

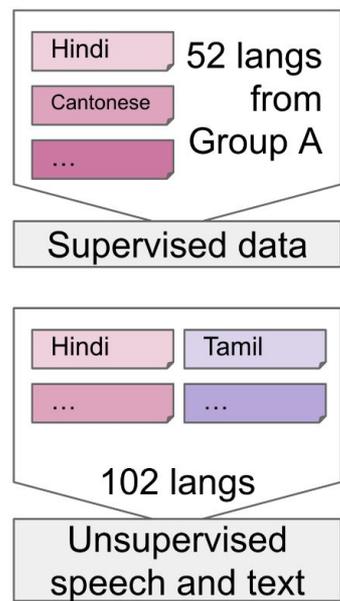
## Solution: **Maestro-U**

- Unsupervised speech and text learning with Maestro
- Promote multilingual knowledge transfer by Language ID and Residual Adapters
- Handling unseen writing systems by UTF-8 Bytes as text representation units

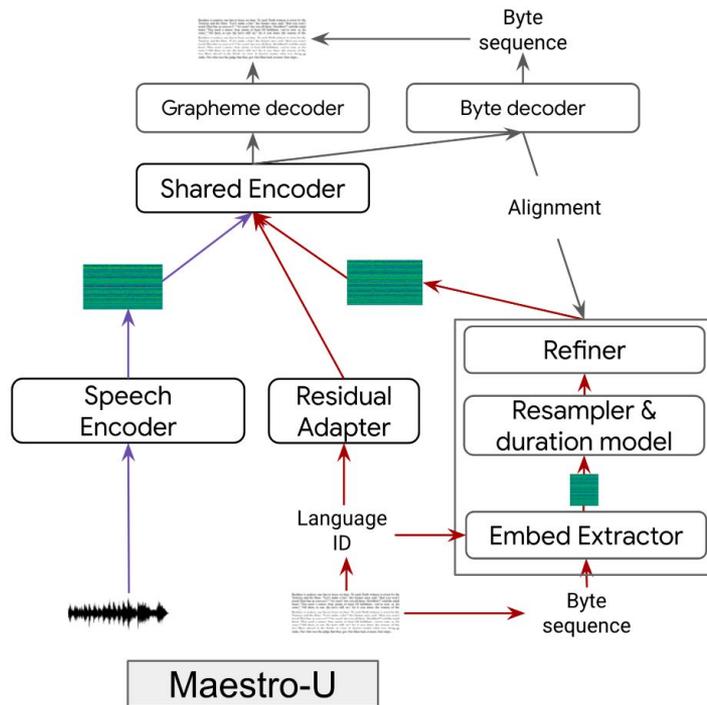
## Result:

- Train ASR models without transcribed speech on 50 unseen FLEURS languages.
- Reduce the CER on languages with no supervised speech from 64.8% to 30.8%.
- Close the gap to oracle performance by 68.5% relative and reduces the CER of 19 languages below 15%.

# Massive multilingual ASR language expansion with zero supervised speech

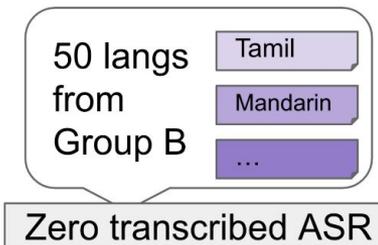


Training



Maestro-U

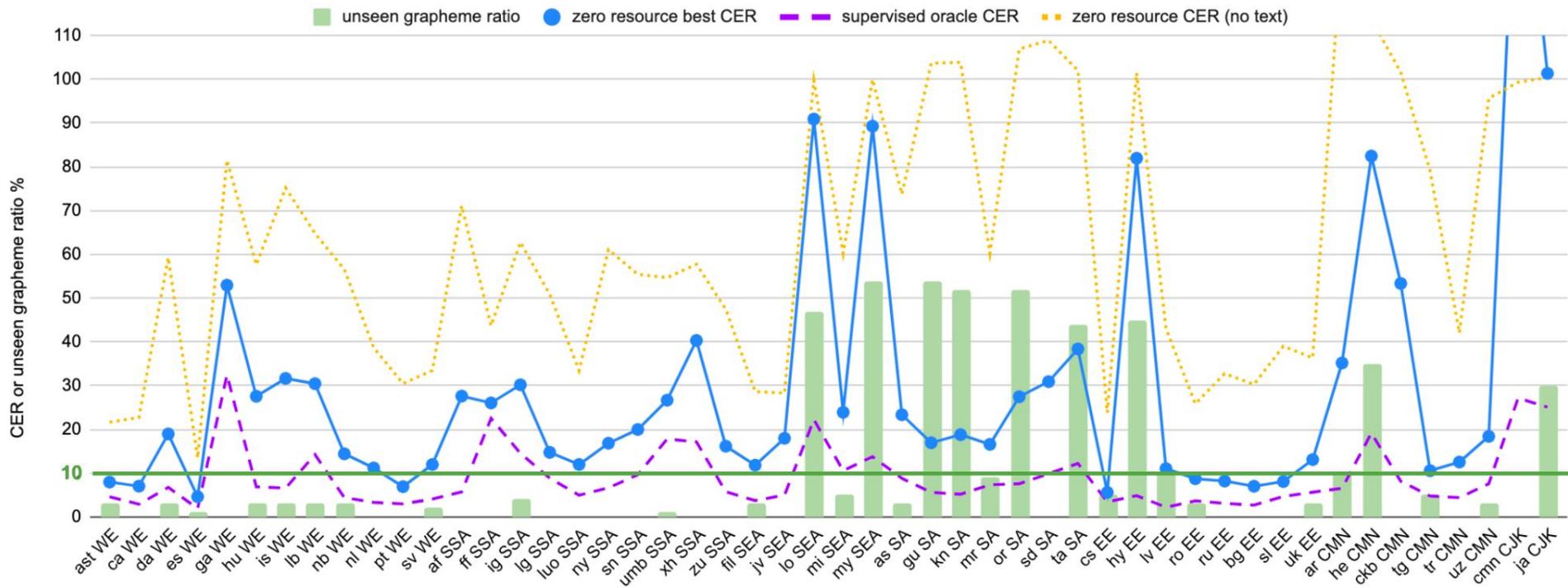
**Text encoder training:** learn to predict speech-like text representations on 52 supervised languages  
**Text encoder inference:** unspoken text learning on 102 languages



Zero transcribed ASR

Evaluation

# Results on 50 unseen languages (FLEURS)



Reduce the CER on languages with no supervised speech from 64.8% to 30.8%.  
Even on the langs with very different writing systems, e.g. South Asian langs

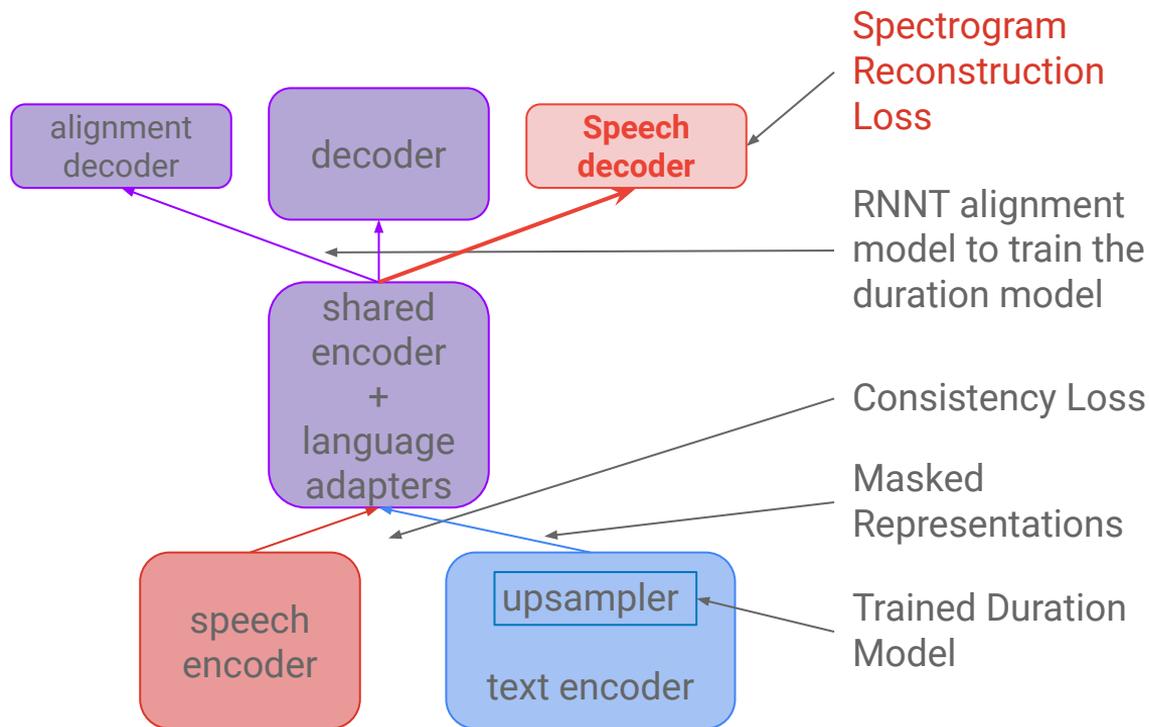
# Multilingual Text to Speech (TTS)

Current TTS covers around 100+ languages.

Around 7000 languages exist in the world.

→ **Can these representations do multilingual TTS??**

# Virtuoso = Maestro + speech decoder !!



Spectrogram  
Reconstruction  
Loss

RNNT alignment  
model to train the  
duration model

Consistency Loss

Masked  
Representations

Trained Duration  
Model

## Unpaired data $\Rightarrow$ Self-supervised learning

- Sp enc  $\rightarrow$  Sp dec  $\Rightarrow$  Masked AE
- Txt enc  $\rightarrow$  Txt dec  $\Rightarrow$  Masked LM

## Paired data $\Rightarrow$ Supervised learning

- Text enc  $\rightarrow$  Speech dec  $\Rightarrow$  TTS
- Speech enc  $\rightarrow$  Text dec  $\Rightarrow$  ASR

## Data

- Untranscribed speech
- Unspoken text
- Paired ASR data (in-the-wild)
- Paired TTS data (in-house)

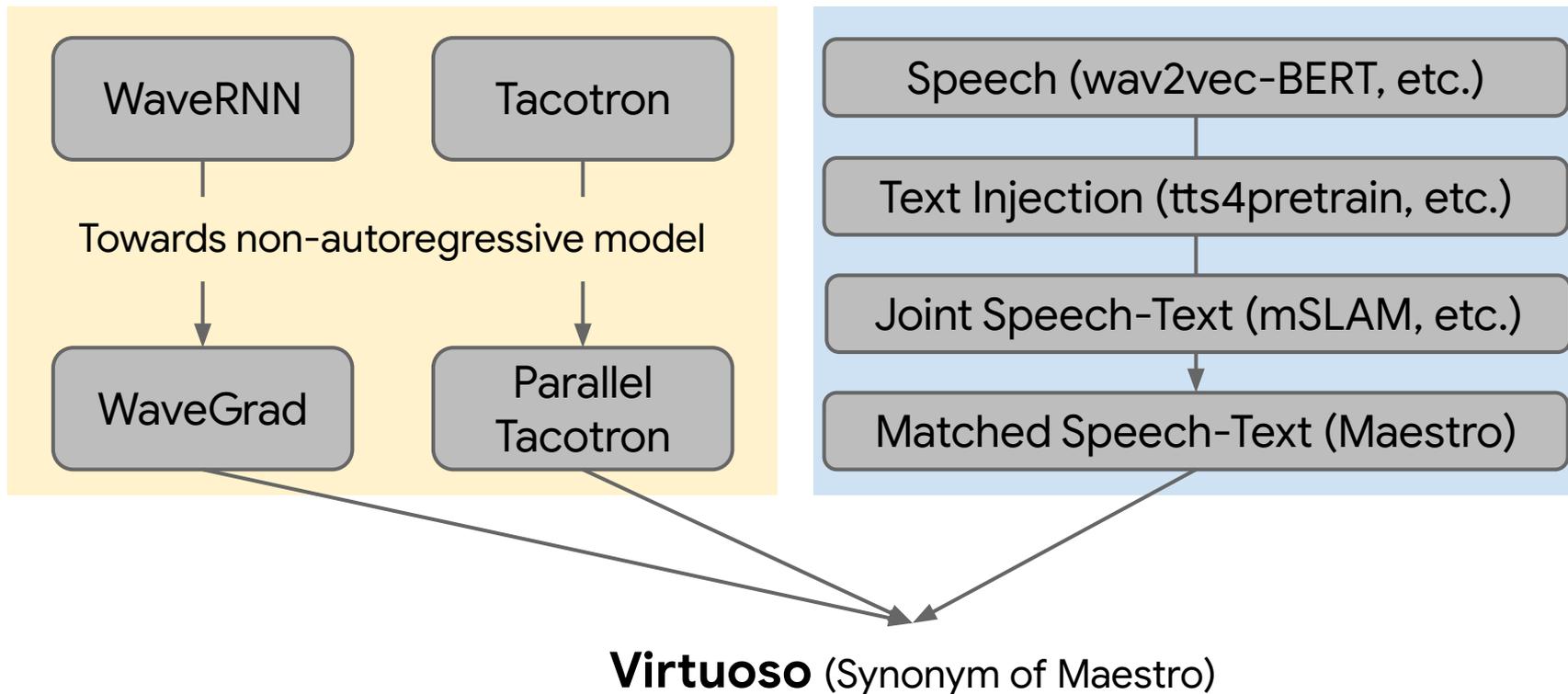
## Text representation

- Phonemes; Graphemes; Bytes

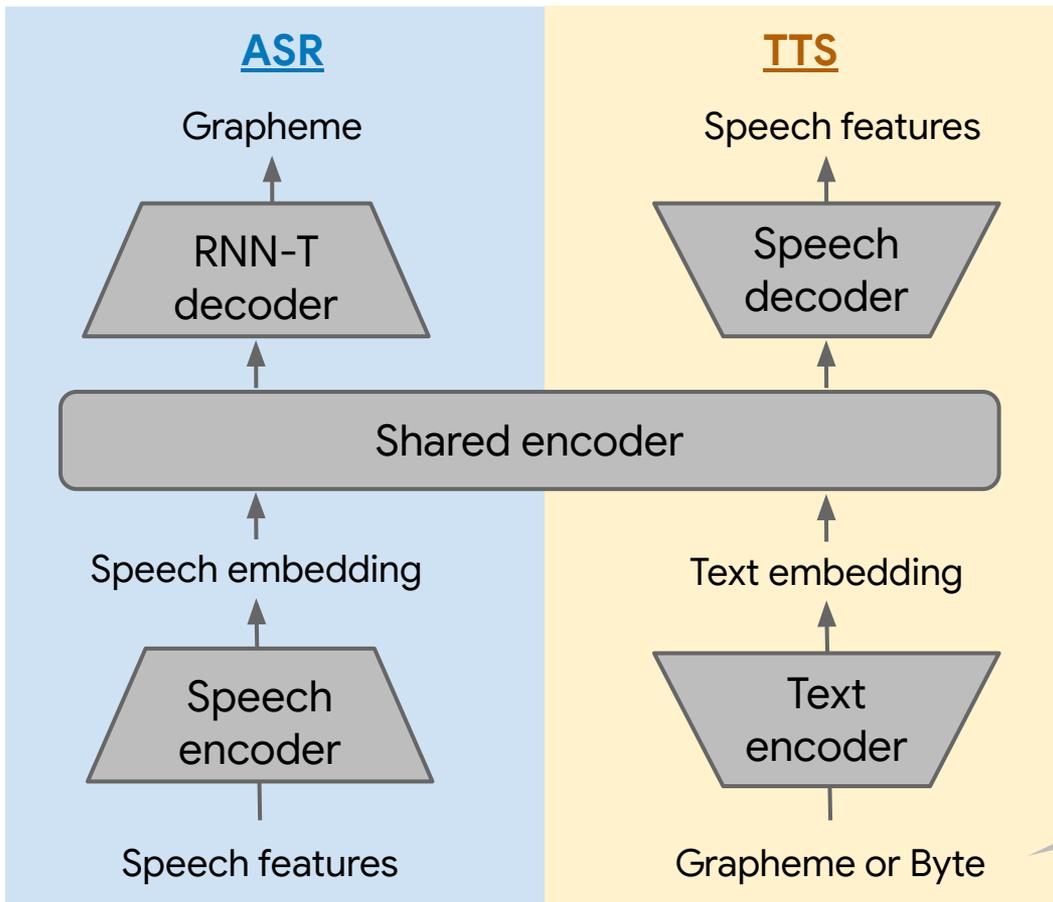
# Meeting of Conventional TTS and Newer methods in ASR

TTS

SpeechSSL



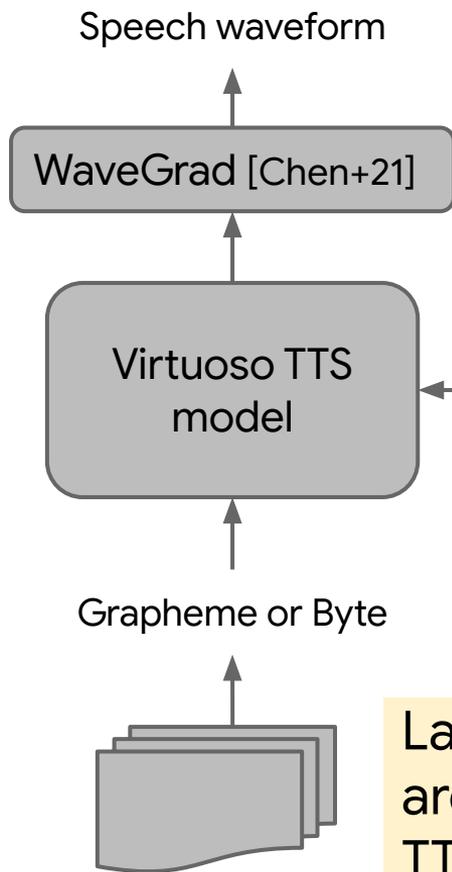
## Consisting of ASR part and TTS part



We can obtain full TTS model  
without fine-tuning

Grapheme or Byte-based TTS  
without any G2P modules

# Zero-Resource TTS



**Can massive multilingual knowledge obtained with ASR and SSL be transferred to TTS?**

Speaker embedding

Sampled from similar locales included in TTS data

Languages which are not included in TTS training data

## Multilingual TTS possible with the same ASR technology

- Virtuoso improved performance for **both major and low-resource locales**.
- Virtuoso performed well in **zero-resource settings**.
- **Byte-based model** achieved the highest linguistic accuracy.
- **Only using paired ASR+TTS** data was better in terms of naturalness.
- **Using unpaired data** was effective for zero-resource settings.

*Takaaki Saeki et al., , EXTENDING MULTILINGUAL SPEECH SYNTHESIS TO 100+ LANGUAGES WITHOUT TRANSCRIBED DATA, ICASSP 2024*

# Representation Learning: Summary

## Learning Within Modality

*Audio:* [Full-sum](#)/Sampling based  
Distillation, Sampling  
Guided-masking,  
Diffusion-based masking,  
Use of ephemeral sources  
(eg. Radio/Podcasts),  
SoundStream + AudioLM

*Text:* Large LMs integrated into  
e2e model

## Learning Across Modalities

Encourage unified  
representations

Share language adapters within  
language families

Acoustic Prompting

Additional modalities/signals  
(image, video, tonal language,  
etc.)

Intermediate representations  
help other downstream tasks  
(phone recognition, NLP?)

## Weak Supervision

Conditional adapters (on topic,  
contextual keywords)

Grounding around other  
information seen in the same  
context  
(text/audio/image/audio)

04

# Evaluating representations

# Evaluating Representations

- Benchmarks:
  - Pooneh Mousavi et al., “DASB - Discrete Audio and Speech Benchmark”: Discrete audio tokens across a wide range of discriminative tasks, including speech recognition, speaker identification and verification, emotion recognition, keyword spotting, and intent classification, as well as generative tasks such as speech enhancement, separation, and text-to-speech  
<http://arXiv:2406.14294>
  - Shikhar Vashishth et al., STAB: Speech Tokenizer Assessment Benchmark : Measurements across in variance, robustness, compressibility, coverage <http://arxiv.org/abs/2409.02384>
- Characterizing neural representations of context-dependent and dynamic patterns of neural activity: speech perception approach [46]

# Evaluating Representations

- Dialog tasks that capture speaking styles in their responses
  - StyleTalk (<https://github.com/DanielLin94144/StyleTalk>)
- Automatically detect domain shifts over time and adapt for best performance
  - *Vision*: stochastic model restoring (Wang et al., 2022), sample-efficiency entropy minimization (Niu et al., 2022a), sharpness-aware reliable entropy minimization (Niu et al., 2022b), and fixed frequency model reset (Press et al., 2024) are examples to address domain shifts.
  - *Speech*: Recent ACL paper, uses the loss function to detect domain shifts (Lin et al., EMNLP 2024)
- MTEB (Massive Text Embedding Benchmark) spans 8 embedding tasks covering a total of 58 datasets and 112 languages. Could we have a similar benchmark for joint representations?

# Promising lines of research

- Can we guide the masking during representation learning?
  - To prevent learning from erroneous samples in the vast amount of audio
  - Need to reduce amount of in-domain data for adapting pre-trained models trained on out-of-domain data [26,27]
  - Can we stabilize performance fluctuations that arise from changes in masking ratio?
  - Can we use a teacher to guide the samples to mask?
- Can we introduce more supervision into the pre-training process?
  - Curriculum training schedule
  - Consistency regularization
- What is a good tokenizer? How do you define 'good'?

# Concluding Remarks

- Code-switching is by no means a solved problem for ASR or other ST/TTS tasks
- Well-represented Data Resources are scarce- how can we grow these?
- Language Identification and Domain shifts are crucial and still remains a difficult problem for several code-switched languages - how do we make models robust to this?
- Joint speech-text representation learning is useful for ASR, ST, TTS. What about other tasks such as recognizing and responding to emphasis in speeches and dialogues? Spoken content retrieval?

*Machine Learning continues to produce large models that can scale and be prompted to solve these tasks. These fundamental challenges remain and more research in these areas will pave the way for usable, scalable, multilingual models.*

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