Deep Learning

End-to-End Automatic Speech Recognition

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Outline

- 1. Introduction to ASR (Automatic Speech Recognition)
- 2. Speech-To-Text (STT) as a Seq2Seq task
- 3. Audio feature extraction
- 4. Training/Pre-training + Fine tuning
- 5. SOTA Architectures, results
- 6. Tools/Practice with NVIDIA NeMo

Introduction to ASR

What is ASR?

• Speech-To-Text (STT): acoustic pressure(time) signal) → text transcription



- Speaker recognition/diarization/verification
- Speech diagnostics
- Speech emotion recognition
- Etc.

Auxiliary information

How is ASR related to deep learning?

Early attempts: limited success

- 1952, Bell Lab, Audrey
- 1961, IBM, Shoebox



First practical ASR systems: statistics and machine learning (HMM, GMM)

1975 CMU, IBM, ... (Baker, Bahl, Jelinek) – 2010



Breakthrough: 2011, Florence Interspeech

Microsoft "Rosetta-stone"

Frank Seide et al

Deep Neural nets for acoustic modeling



Deep learning is ASR



End-to-end approach (/almost/ purely deep neural network)



Speech-To-Text (STT/ASR) as a Seq2Seq task

ASR with S2S

- Encoder-Decoder structures
 - Conv1D (NVIDIA Jasper/QuartzNet)
 - RNN (DeepSpeech)
 - Transformer (SOTA: Google Conformer, META wav2vec2, OpenAI Whisper)
- Special loss function: CTC
 - Why we need it: time alignment problem



Listen-Attend-Spell (LAS) model

• Alternative to CTC alignment is best practice: attention + CTC (Watanabe et al)



T structures

RNN-T (Alex Graves): RNN-Transducer



Conformer-T, Transformer-T, ...

Audio Feature Extraction

Mel-Spectrogram



Feature extraction with Conv1D

Wav2vec 2.0 Latent Feature Encoder



jonathanbgn.com

https://jonathanbgn.com/2021/09/30/illustrated-wav2vec-2.html



Why and how?

- Multitude of pre-trained models
- Just not what you really need ...
- Training from scratch?
 - At least 200 hours but 2k or 20k performs significantly better
 - Needs lot of GPU's (multi GPU, multi node PyTorch DDP at least...)
- Transfer learning?! Yes!



- Always use pre-trained model + fine-tuning!
 - Data efficient: fine-tuning works even for 1 hours of supervised data



(Pre-)/(post-) training

- Pre-training:
 - Supervised (audio + exact transcription)
 - Weakly supervised (audio + edited/simplified transcription)
 - Self-supervised (audio only!)
- "Post training": noisy student training on ASR pseudo labels
- "ASR" augmentations:
 - SpecAugment (2019)
 - Speed perturbation



State of The Art



Conformer

/as encoder with a light decoder/

Transformer, Self-attention + Convolution + FF

Google

Whisper: Encoder + decoder transformer

Weak supervision = not exact transcriptions

- Multilingual, multitask learning
- Language ID
- Punctualization

BUT

- Slowww...
- Non-streamable
- "Input audio is split into 30second chunks"→ latency
- Non-English Accuracy?





Case-study

Hungarian ASR on studio quality spontaneous and read/repeated speech

Experimental data (Hungarian)

Table 1: Main characteristics of data sets used in the experiments.

	HGC SPOK	train-114	dev-repet	BEA-Base dev-spont	eval-repet	eval-spont	CV test
Length [hours]	-	71.2	0.65	4.02	0.95	4.91	6.8
Num of speakers	-	114	10	10	16	16	220
Num of segments	-	76 881	568	4 893	858	5 693	4 871
Num of characters	516.84M	3.1M	28 467	154 994	43 448	197 738	250 709
Num of words	56.13M	0.56M	4 110	27 939	6 229	35 178	35 485
3-gram PPL	-	-	924	771	846	857	2 387
OOV rate [%]	-	-	1.6	1.9	1.4	1.7	3.1
	LM training		AM training	5		Evaluation	

Evaluation Metrics

Accuracy

English/Hungarian : Word Error Rate (WER) Mandarin: Character Error Rate (CER)

S: Substitution. D: Deletion I: Insertion C: Correct N: Total numbers of characters, N = S + D + C

• Real-time Factor (RTF)

 $WER(CER) = \frac{S+D+I}{N} * 100\%$

$$RTF = \frac{T_{process}}{T_{audio}}$$

If RTF < 1, indicating the system can transcribe faster than real-time.

N-gram Language Model (LM)

• Language Model (LM)

A statistical model used to predict the likelihood of a sequence of words.

• N-gram

An N-gram refers to a continuous sequence of N items.



Transcribe spoken words by predicting the most likely word sequences.



Training from scratch vs. cross-lingual transfer learning

Conformer – NVIDIA NeMo implementation, supervised pre-training

From scratch, supervised end-to-end ASR with convolutional/Conformer acoustic models

Structure/ LM CV num of eval-repet eval-spont parameters 26.70 QuartzNet 11.56 26.83 15x3 / 12.7M 3-gram 6.86 **Conformer-**12.73 25.31 49.8 Small / 13M 7.98 22.78 42.7 6-gram **Conformer-**10.98 24.93 49.8 Medium / 30M 5.65 21.01 42.9 6-gram

WER[%] results on BEA-Base (starting from scratch)



QuartzNet (baseline)

Supervised pre-training* (En) + fine-tuning (Hu)

QuartzNet (baseline) 15x5 (18M) based transfer learning WER[%] results on BEA-Base

Pre-training data size [hours]	LM	eval-repet	eval-spont	CV
	_	10.63	24.87	_
	3-gram	5.83	25.23	—

Conformer Small (13M) / Large (121M) transfer learning WER[%] results on BEA-Base

~10k	–	11.22	21.39	40.8
	6-gram	4.96	17.77	34.8
	–	5.2	17.24	34.8
	6-gram	3.66	16.25	30.8



End-to-end deep learning approach – weakly-supervised training

Whisper in zero-shot and fine-tuning setups

Whisper results: zero-shot vs. fine-tuning

- (Pre-)training data size: 680k hours
- Num of languages: 97
- Composition: 83% English ... 0.03% Hungarian

Whisper Medium/Large_v2 WER [%] results of BEA-Base

Model	Fine-tuning	Num of parameters	eval-repet	eval-spont	CV
Whisper-Medium	_	769M	22.33	38.67	27.6
Whisper-Large	_	1550M	18.04	32.76	20.4
Whisper-Medium	Decoder (456M)	769M	4.90	20.60	27.9
Whisper-Large	Decoder (906M)	1550M	4.37	18.69	23.7



End-to-end deep learning approach – self-supervised pre-training

Transcription-free SSL pre-training + wav2vec2 encoder + attentional decoder

Self-Supervised Pre-training based Transfer Learning

- SSL pre-training
- All pre-trained models are downloaded from HuggingFace
 - wav2vec2-large-lv60: LibriVox (English)
 - wav2vec2-large-xlsr-53: CommonVoice + BABEL + Multilingual LibriSpeech
 - wav2vec2-xls-r-300m: CV + BABEL + MLS + VoxPopuli + VoxLingua107 (0.04% Hungarian)
 - wav2vec2-mms-300: MMS-lab-U + VoxPopuli + ... (Massively multilingual incl. Hungarian)

....

• wav2vec2-uralic: VoxPopuli (3 languages, 41% Hungarian)

Context

representations

Quantized

Latent speech representations

representations ${\cal Q}$

raw waveform \mathcal{X}

С

 \mathcal{Z}



 y_i

- Loss: CTC + NLL
- Decoder: GRU
- Encoder: wav2vec2.0large, 300M
- LM: –/Transformer

Wav2vec2 (SSL)-based Transfer Learning Results

wav2vec2-large+GRU+CTC+Attention+BPE_600 based transfer learning WER[%] results on BEA-Base

Model	SSL Pre- training languages	Pre-training data size [hours]	LM	eval-repet	eval-spont	CV
wav2vec2-large-lv60	1*	53k	_	8.46	19.17	36.5
wav2vec2-large-xlsr-53 wav2vec2-xls-r-300m wav2vec2-mms-300	53 128 1406	56k 440k 491k		5.81 6.16 6.56	16.62 15.61 18.82	34.2 30.5 34.9
wav2vec2-uralic	3**	42k	_ Transformer	4.24 2.42	11.55 10.50	21.3 17.2

* = English

** = Estonian + Finnish + Hungarian

Wav2vec2 (SSL)-based Transfer Learning Results

wav2vec2-large+GRU+CTC+Attention+BPE_600 based transfer learning CER[%] results on BEA-Base

Model	SSL Pre- training languages	Pre-training data size [hours]	LM	eval-repet	eval-spont	CV
wav2vec2-large-lv60	1*	60k	_	2.6	5.9	11.2
wav2vec2-large-xlsr-53 wav2vec2-xls-r-300m wav2vec2-mms-300	53 128 1406	56k 440k 491k	 	2.1 2.4 2.2	5.5 5.1 5.8	10.5 8.6 9.1
wav2vec2-uralic	3**	42k	– Transformer	1.7 0.7	3.7 3.3	5.8 4.5

* = English

** = Estonian + Finnish + Hungarian

Final comparison – with respect to RTF

Best of Conformer vs. Wav2vec2 vs. Whisper

Best ASR results on spontaneous Hungarian (without LM) vs. inference times



Relative inference times on an RTX A6000 GPU



/Word Error Rate: the lower the better/

/Real-Time Factor: the lower the better/

Recommended end-to-end ASR tools

TTTESPnet

SpeechBrain

Whisper

- <u>https://github.com/espnet/</u>
- <u>https://github.com/facebookresearch/fairseq</u>
- <u>https://github.com/k2-fsa/k2</u>
- <u>https://github.com/lhotse-speech/lhotse</u>
- <u>https://speechbrain.github.io/</u>
- <u>https://github.com/openai/whisper</u>
- <u>https://github.com/NVIDIA/NeMo</u>
- <u>https://github.compenet-e2e</u> **DVIDIA**. NEMO



FAIRSEO

LHOTSE



References

- <u>Stanford Lecture on ASR</u>
- "An Intuitive Explanation of Connectionist Temporal Classification"
- Explanation of CTC with Prefix Beam Search
- Listen Attend and Spell Paper (seq2seq ASR model)
- Explanation of the mel spectrogram in more depth
- Jasper Paper
- QuartzNet paper
- <u>SpecAugment Paper</u>
- Explanation and visualization of SpecAugment
- <u>Cutout Paper</u>
- <u>Transfer Learning Blogpost</u>

Please, don't forget to send feedback: https://bit.ly/bme-dl



Thank you for your attention

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