Automatic Speech Recognition

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Implementation

HuggingFace ASR Model for Turkish

Turkish has a small dataset on Common Voice so it's mostly used as a starting point to show how we can fine tune a model for a new language.

These are the steps that are followed:

The github repository is at

https://github.com/huggingface/transformers/tree/main/examples/pytorch/speech-recognition

Setup

mkdir asr cd asr vi run_speech_recognition_seq2seq.py vi requirements.txt sudo apt-get update sudo apt-get install software-properties-common sudo add-apt-repository ppa:deadsnakes/ppa sudo apt-get update sudo apt-get install python3.10 wget https://bootstrap.pypa.io/get-pip.py python3 get-pip.py pip --version pip install virtualenv virtualenv venv source venv/bin/activate pip install -r requirements.txt pip install git+https://github.com/huggingface/transformers git config --global credential.helper store huggingface-cli login sudo curl -s https://packagecloud.io/install/repositories/github/git-lfs/script.deb.sh | sudo bash sudo apt-get install git-lfs git Ifs install

Setup for Oracle

mkdir asr cd asr vi run_speech_recognition_seq2seq.py vi requirements.txt sudo yum update sudo yum install epel-release sudo yum install -y yum-utils sudo yum-config-manager --add-repo https://copr.fedorainfracloud.org/coprs/deadsnakes/ppa/repo/epel-7/deadsnakes-ppa-epel-7.repo sudo yum install -y gcc openssl-devel bzip2-devel libffi-devel zlib-devel wget https://www.python.org/ftp/python/3.10.0/Python-3.10.0.tgz tar -xf Python-3.10.0.tgz cd Python-3.10.0 ./configure --enable-optimizations make -j\$(nproc) sudo make altinstall sudo In -s \$(pwd)/python /usr/local/bin/python3.10 sudo In -sf /usr/local/bin/python3.10 /usr/bin/python3 which python3.10 python --version wget https://bootstrap.pvpa.io/get-pip.pv python get-pip.py pip --version pip install virtualenv virtualenv venv source venv/bin/activate pip install -r requirements.txt sudo yum install git git --version sudo yum install git-lfs pip install git+https://github.com/huggingface/transformers git config --global credential.helper store huggingface-cli login sudo curl -s https://packagecloud.io/install/repositories/github/git-lfs/script.rpm.sh | sudo bash sudo yum install git-lfs sudo yum install gcc make ncurses-devel wget https://ftp.gnu.org/gnu/screen/screen-4.8.0.tar.gz tar xvf screen-4.8.0.tar.gz cd screen-4.8.0 ./configure make sudo

Train CTC Model for Turkish

make install

```
python run_speech_recognition_ctc.py \
--dataset_name="common_voice" \
--model_name_or_path="facebook/wav2vec2-large-xlsr-53" \
--dataset_config_name="tr" \
--output_dir="./wav2vec2-common_voice-tr-demo" \
--overwrite_output_dir \
```

```
--num_train_epochs="15" \
--per_device_train_batch_size="16" \
--gradient_accumulation_steps="2" \
--learning_rate="3e-4" \
--warmup_steps="500" \
--evaluation_strategy="steps" \
--text_column_name="sentence" \
--length column name="input length" \
--save steps="400" \
--eval_steps="100" \
--layerdrop="0.0" \
--save total limit="3" \
--freeze feature encoder \
--gradient_checkpointing \
--chars_to_ignore , ? . ! - \; \: \" " % ' " � \
--fp16 \
--group_by_length \
--push_to_hub \
--do_train --do_eval
```

Final Output

```
{'train_runtime': 6409.9307, 'train_samples_per_second': 8.139, 'train_steps_per_second': 0.255, 'train_loss': 1.070046563688039, 'epoch': 15.0}
100%
```

| 1635/1635 [1:46:49<00:00, 3.92s/it]

Saving model checkpoint to ./wav2vec2-common_voice-tr-demo

Configuration saved in ./wav2vec2-common_voice-tr-demo/config.json

Model weights saved in ./wav2vec2-common_voice-tr-demo/pytorch_model.bin

Feature extractor saved in ./wav2vec2-common_voice-tr-demo/preprocessor_config.json

Saving model checkpoint to ./wav2vec2-common_voice-tr-demo

Configuration saved in ./wav2vec2-common_voice-tr-demo/config.json

 $Model\ weights\ saved\ in\ ./wav2vec2-common_voice-tr-demo/pytorch_model.bin$

Feature extractor saved in ./wav2vec2-common_voice-tr-demo/preprocessor_config.json

Several commits (2) will be pushed upstream.

 $04/07/2023\ 22:04:36\ -\ WARNING\ -\ hugging face_hub.repository\ -\ Several\ commits\ (2)\ will\ be\ pushed\ upstream.$

The progress bars may be unreliable.

04/07/2023 22:04:36 - WARNING - huggingface_hub.repository - The progress bars may be unreliable.

Upload file pytorch_model.bin: 1.18GB [04:52, 6.66MB/s]To

https://huggingface.co/tarunanand/wav2vec2-common_voice-tr-demo

29c21b3..7c2047e main -> main

04/07/2023 22:09:32 - WARNING - huggingface_hub.repository - To

https://huggingface.co/tarunanand/wav2vec2-common_voice-tr-demo

29c21b3..7c2047e main -> main

Upload file pytorch_model.bin:

100%

1.18G/1.18G [04:54<00:00, 4.29MB/s]

To https://huggingface.co/tarunanand/wav2vec2-common_voice-tr-demo

7c2047e..d28c03f main -> main

04/07/2023 22:09:37 - WARNING - huggingface_hub.repository - To

https://huggingface.co/tarunanand/wav2vec2-common voice-tr-demo

```
7c2047e..d28c03f main -> main
***** train metrics *****
 epoch
                       15.0
train_loss
                  = 1.07
                   = 1:46:49.93
train_runtime
train_samples
                   =
                         3478
train_samples_per_second = 8.139
train steps per second = 0.255
04/07/2023 22:09:40 - INFO - main - *** Evaluate ***
The following columns in the evaluation set don't have a corresponding argument in `Wav2Vec2ForCTC.forward` and have
been ignored: input length. If input length are not expected by 'Wav2Vec2ForCTC.forward', you can safely ignore this
message.
***** Running Evaluation *****
Num examples = 1647
Batch size = 8
100%
                                             | 206/206 [01:46<00:00, 1.93it/s]
**** eval metrics ****
epoch
                      15.0
eval_loss
                  = 0.376
eval_runtime
                  = 0:01:47.42
eval_samples
                        1647
eval_samples_per_second = 15.332
eval_steps_per_second =
                            1.918
eval_wer
               = 0.3521
Saving model checkpoint to ./wav2vec2-common_voice-tr-demo
Configuration saved in ./wav2vec2-common_voice-tr-demo/config.json
Model weights saved in ./wav2vec2-common_voice-tr-demo/pytorch_model.bin
Feature extractor saved in ./wav2vec2-common_voice-tr-demo/preprocessor_config.json
To https://huggingface.co/tarunanand/wav2vec2-common voice-tr-demo
 d28c03f..781e65f main -> main
04/07/2023 22:11:39 - WARNING - huggingface hub.repository - To
https://huggingface.co/tarunanand/wav2vec2-common voice-tr-demo
 d28c03f..781e65f main -> main
To https://huggingface.co/tarunanand/wav2vec2-common voice-tr-demo
 781e65f..e985695 main -> main
04/07/2023 22:11:43 - WARNING - huggingface_hub.repository - To
https://huggingface.co/tarunanand/wav2vec2-common voice-tr-demo
 781e65f..e985695 main -> main
```

Train CTC Model for Hindi

```
For Hindi use the following:

Single GPU

python run_speech_recognition_ctc.py \
    --dataset_name="common_voice" \
    --model_name_or_path="facebook/wav2vec2-large-xlsr-53" \
    --dataset_config_name="hi" \
    --output_dir="./wav2vec2-common_voice-hi-demo" \
    --overwrite_output_dir \
```

```
--num train epochs="15" \
--per_device_train_batch_size="16" \
--gradient_accumulation steps="2" \
--learning rate="3e-4" \
--warmup steps="500" \
--evaluation strategy="steps" \
--text column name="sentence" \
--length column name="input length" \
--save steps="400" \
--eval steps="100" \
--layerdrop="0.0" \
--save total limit="3" \
--freeze feature encoder \
--gradient_checkpointing \
--chars_to_ignore , ? . ! - \; \: \" " % ' " � \
--fp16 \
--group_by_length \
--push to hub \
--do_train --do_eval
```

Train Seq2Seq Model for Hindi

```
Multi GPU Training for Whisper Medium 2xV100-16-240GB Ubuntu22-14
screen -dmS asr torchrun --nproc per node 2 run speech recognition seg2seg.py
--model_name_or_path="openai/whisper-medium"
                                                   --dataset_name="mozilla-foundation/common_voice_11_0"
--dataset_config_name="hi" --language="hindi" --train_split_name="train+validation" --eval_split_name="test"
--max steps="5000"
                         --output_dir="./whisper-medium-hi"
                                                            --per_device_train_batch_size="16"
--per device eval batch size="16" --logging steps="25"
                                                            --learning rate="1e-5"
                                                                                       --warmup steps="500"
--evaluation strategy="steps"
                                  --eval steps="1000"
                                                            --save_strategy="steps"
                                                                                       --save steps="1000"
--generation_max_length="225"
                                  --preprocessing_num_workers="16" --length_column_name="input_length"
--max_duration_in_seconds="30"
                                  --text column name="sentence"
                                                                     --freeze_feature_encoder="False"
                                                   --overwrite output dir
                                                                              --do train
--gradient checkpointing
                         --group by length --fp16
                                                                                               --do eval
--predict_with_generate
                          --use_auth_token --push_to_hub
Multi GPU Training for Whisper Large V2 GDC-2xA100-32-230GB_Ubuntu22-175
screen -dmS asr torchrun --nproc per node 2 run speech recognition seg2seq.py
--model name or path="openai/whisper-large-v2"
                                                    --dataset name="mozilla-foundation/common voice 11 0"
--dataset config name="hi" --language="hindi" --train split name="train+validation" --eval split name="test"
--max steps="5000"
                          --output_dir="./whisper-large-v2-hi" --per_device_train_batch_size="10"
--per device eval batch size="10" --logging steps="25"
                                                            --learning rate="1e-5"
                                                                                       --warmup steps="500"
--evaluation strategy="steps"
                                  --eval steps="1000"
                                                            --save strategy="steps"
                                                                                       --save steps="1000"
--generation max length="225"
                                  --preprocessing num workers="16" --length column name="input length"
--max_duration_in_seconds="30"
                                  --text column name="sentence"
                                                                     --freeze_feature_encoder="False"
--gradient checkpointing
                          --group by length --fp16
                                                  --overwrite output dir
                                                                              --do train
                                                                                               --do eval
--predict with generate
                          --use auth token --push to hub
```

Train Seq2Seq Model for Kangri

Multi GPU Training for Whisper Large V2 GDC-2xA100-32-230GB_Ubuntu22-175 on Snow-Mountain Dataset

```
screen -dmS asr torchrun --nproc_per_node 2 run_speech_recognition_seq2seq.py
--model_name_or_path="vasista22/whisper-hindi-large-v2"
--dataset_name="bridgeconn/snow-mountain" --dataset_config_name="kangri"
--language="hindi" --train_split_name="train_500" --eval_split_name="test_common"
--max_steps="5000" --output_dir="./whisper-large-v2-kangri"
--per_device_train_batch_size="8" --per_device_eval_batch_size="8"
--logging_steps="25" --learning_rate="le-5" --warmup_steps="500"
--evaluation_strategy="steps" --eval_steps="1000" --save_strategy="steps"
--save_steps="1000" --generation_max_length="225"
--preprocessing_num_workers="16" --length_column_name="input_length"
--max_duration_in_seconds="30" --text_column_name="sentence"
--freeze_feature_encoder="False" --gradient_checkpointing --group_by_length
--fp16 --overwrite_output_dir --do_train --do_eval --predict_with_generate
--use_auth_token --push_to_hub
```

Troubleshooting

- ProcessGroupNCCL is only supported with GPUs, no GPUs found!
 This issue is usually due to the Cloud Provider not having proper drivers for CUDA. Contact IT/Support.
- pytorch_cuda_alloc_conf cuda out of memory Tried the following and the suggestions in the link below but ultimately had to go with a lower model. export CUDA_VISIBLE_DEVICES=0,1 export PYTORCH_CUDA_ALLOC_CONF=garbage_collection_threshold:0.6,max_split_ size_mb:128

https://medium.com/@snk.nitin/how-to-solve-cuda-out-of-memory-error-850bb24 7cfb2

3.

Command Line Arguments

```
(venv) root@localhost:~/asr# python3 run speech recognition seg2seg.py --help
usage: run_speech_recognition_seq2seq.py [-h] --model_name_or_path
MODEL NAME OR PATH [--config name CONFIG NAME] [--tokenizer name
TOKENIZER NAME] [--feature extractor name FEATURE EXTRACTOR NAME]
                      [--cache dir CACHE DIR] [--use fast tokenizer
[USE_FAST_TOKENIZER]] [--no_use_fast_tokenizer] [--model_revision
MODEL REVISION] [--use auth token [USE AUTH TOKEN]]
                      [--freeze feature encoder [FREEZE FEATURE ENCODER]]
[--no freeze feature encoder] [--freeze encoder [FREEZE ENCODER]]
                      [--forced decoder ids FORCED DECODER IDS
[FORCED DECODER IDS ...]] [--suppress tokens SUPPRESS TOKENS
[SUPPRESS_TOKENS ...]] [--apply_spec_augment [APPLY_SPEC_AUGMENT]]
                      [--dataset name DATASET NAME] [--dataset config name
DATASET_CONFIG_NAME] [--overwrite_cache [OVERWRITE_CACHE]]
[--preprocessing num workers PREPROCESSING NUM WORKERS]
                      [--max train samples MAX TRAIN SAMPLES]
[--max eval samples MAX EVAL SAMPLES] [--audio column name
AUDIO COLUMN NAME] [--text column name TEXT COLUMN NAME]
                      [--max_duration_in_seconds
MAX_DURATION_IN_SECONDS] [--min_duration_in_seconds
MIN_DURATION_IN_SECONDS] [--preprocessing_only [PREPROCESSING_ONLY]]
                      [--train split name TRAIN SPLIT NAME] [--eval split name
EVAL SPLIT NAME] [--do lower case [DO LOWER CASE]] [--no do lower case]
[--language LANGUAGE] [--task TASK] --output dir
                      OUTPUT DIR [--overwrite output dir
[OVERWRITE OUTPUT DIR]] [--do train [DO TRAIN]] [--do eval [DO EVAL]]
[--do_predict [DO_PREDICT]] [--evaluation_strategy {no,steps,epoch}]
                      [--prediction loss only [PREDICTION LOSS ONLY]]
[--per device train batch size PER DEVICE TRAIN BATCH SIZE]
[--per_device_eval_batch_size PER_DEVICE_EVAL_BATCH_SIZE]
                      [--per_gpu_train_batch_size
PER_GPU_TRAIN_BATCH_SIZE] [--per_gpu_eval_batch_size
PER_GPU_EVAL_BATCH_SIZE] [--gradient_accumulation_steps
GRADIENT ACCUMULATION STEPS
                      [--eval accumulation steps EVAL ACCUMULATION STEPS]
[--eval delay EVAL DELAY] [--learning rate LEARNING RATE] [--weight decay
WEIGHT_DECAY] [--adam_beta1 ADAM_BETA1]
```

```
[--adam beta2 ADAM BETA2] [--adam epsilon
ADAM EPSILON] [--max grad norm MAX GRAD NORM] [--num train epochs
NUM TRAIN EPOCHS] [--max steps MAX STEPS]
                      [--Ir scheduler type
{linear,cosine,cosine_with_restarts,polynomial,constant,constant_with_warmup,inverse_
sqrt}] [--warmup ratio WARMUP RATIO] [--warmup steps WARMUP STEPS]
                      [--log level {debug,info,warning,error,critical,passive}]
[--log level replica {debug,info,warning,error,critical,passive}] [--log on each node
[LOG ON EACH NODE]]
                      [--no log on each node] [--logging dir LOGGING DIR]
[--logging_strategy {no,steps,epoch}] [--logging_first_step [LOGGING_FIRST_STEP]]
[--logging steps LOGGING STEPS]
                      [--logging nan inf filter [LOGGING NAN INF FILTER]]
[--no logging nan inf filter] [--save strategy {no,steps,epoch}] [--save steps
SAVE STEPS]
                      [--save total limit SAVE TOTAL LIMIT] [--save safetensors
[SAVE_SAFETENSORS]] [--save_on_each_node [SAVE_ON_EACH_NODE]]
[--no_cuda [NO_CUDA]] [--use_mps_device [USE_MPS_DEVICE]]
                      [--seed SEED] [--data seed DATA SEED] [--jit mode eval
[JIT MODE EVAL]] [--use ipex [USE IPEX]] [--bf16 [BF16]] [--fp16 [FP16]]
[--fp16_opt_level FP16_OPT_LEVEL]
                      [--half precision backend {auto,cuda amp,apex,cpu amp}]
[--bf16_full_eval [BF16_FULL_EVAL]] [--fp16_full_eval [FP16_FULL_EVAL]] [--tf32
TF32] [--local_rank LOCAL_RANK]
                      [--xpu backend {mpi,ccl,gloo}] [--tpu num cores
TPU NUM CORES] [--tpu metrics debug [TPU METRICS DEBUG]] [--debug
DEBUG] [--dataloader drop last [DATALOADER DROP LAST]]
                      [--eval steps EVAL STEPS] [--dataloader num workers
DATALOADER_NUM_WORKERS] [--past_index PAST_INDEX] [--run_name
RUN NAME] [--disable tqdm DISABLE TQDM]
                      [--remove unused columns
[REMOVE UNUSED COLUMNS]] [--no remove unused columns] [--label names
LABEL NAMES [LABEL NAMES ...]] [--load best model at end
[LOAD BEST MODEL AT END]]
                      [--metric for best model METRIC FOR BEST MODEL]
[--greater is better GREATER IS BETTER] [--ignore data skip
[IGNORE DATA SKIP]] [--sharded ddp SHARDED DDP] [--fsdp FSDP]
                      [--fsdp min num params FSDP MIN NUM PARAMS]
[--fsdp config FSDP CONFIG] [--fsdp transformer layer cls to wrap
FSDP TRANSFORMER LAYER CLS TO WRAP] [--deepspeed DEEPSPEED]
```

```
[--label smoothing factor LABEL SMOOTHING FACTOR]
                     [--optim
{adamw hf,adamw torch,adamw torch fused,adamw torch xla,adamw apex fused,a
dafactor,adamw bnb 8bit,adamw anyprecision,sgd,adagrad}] [--optim args
OPTIM_ARGS]
                     [--adafactor [ADAFACTOR]] [--group by length
[GROUP BY LENGTH]] [--length column name LENGTH COLUMN NAME]
[--report to REPORT TO [REPORT TO ...]]
                     [--ddp find unused parameters
DDP FIND UNUSED PARAMETERS] [--ddp bucket cap mb
DDP BUCKET CAP MB] [--dataloader pin memory [DATALOADER PIN MEMORY]]
[--no dataloader pin memory]
                     [--skip memory metrics [SKIP MEMORY METRICS]]
[--no skip memory metrics] [--use legacy prediction loop
[USE_LEGACY_PREDICTION_LOOP]] [--push_to_hub [PUSH_TO_HUB]]
                     [--resume from checkpoint
RESUME FROM CHECKPOINT] [--hub model id HUB MODEL ID] [--hub strategy
{end,every save,checkpoint,all checkpoints}] [--hub token HUB TOKEN]
                     [--hub private repo [HUB PRIVATE REPO]]
[--gradient checkpointing [GRADIENT CHECKPOINTING]]
[--include inputs for metrics [INCLUDE INPUTS FOR METRICS]]
                     [--fp16 backend {auto,cuda amp,apex,cpu amp}]
[--push_to_hub_model_id PUSH_TO_HUB_MODEL_ID] [--push_to_hub_organization
PUSH_TO_HUB_ORGANIZATION]
                     [--push to hub token PUSH TO HUB TOKEN]
[--mp parameters MP PARAMETERS] [--auto find batch size
[AUTO FIND BATCH SIZE]] [--full determinism [FULL DETERMINISM]]
                     [--torchdynamo TORCHDYNAMO] [--ray scope
RAY SCOPE] [--ddp timeout DDP TIMEOUT] [--torch compile [TORCH COMPILE]]
[--torch compile backend TORCH COMPILE BACKEND]
                     [--torch compile mode TORCH COMPILE MODE]
[--sortish sampler [SORTISH SAMPLER]] [--predict with generate
[PREDICT WITH GENERATE]] [--generation max length
GENERATION MAX LENGTH]
                     [--generation num beams GENERATION NUM BEAMS]
[--generation config GENERATION CONFIG]
options:
-h, --help
               show this help message and exit
--model name or path MODEL NAME OR PATH
```

Path to pretrained model or model identifier from huggingface.co/models (default: None) --config name CONFIG NAME Pretrained config name or path if not the same as model name (default: None) --tokenizer name TOKENIZER NAME Pretrained tokenizer name or path if not the same as model name (default: None) --feature extractor name FEATURE EXTRACTOR NAME feature extractor name or path if not the same as model name (default: None) --cache dir CACHE DIR Where to store the pretrained models downloaded from huggingface.co (default: None) --use fast tokenizer [USE FAST TOKENIZER] Whether to use one of the fast tokenizer (backed by the tokenizers library) or not. (default: True) --no use fast tokenizer Whether to use one of the fast tokenizer (backed by the tokenizers library) or not. (default: False) --model revision MODEL REVISION The specific model version to use (can be a branch name, tag name or commit id). (default: main) --use auth token [USE AUTH TOKEN] Will use the token generated when running 'huggingface-cli login' (necessary to use this script with private models). (default: False) --freeze feature encoder [FREEZE FEATURE ENCODER] Whether to freeze the feature encoder layers of the model. (default: True) --no freeze feature encoder Whether to freeze the feature encoder layers of the model. (default: False) --freeze encoder [FREEZE ENCODER]

Whether to freeze the entire encoder of the seq2seq model. (default:

--forced_decoder_ids FORCED_DECODER_IDS [FORCED_DECODER_IDS ...]

A list of pairs of integers which indicates a mapping from generation indices to token indices that will be forced before sampling. For example, [[0, 123]] means the first generated token will

always be a token of index 123. (default: None)

False)

--suppress_tokens SUPPRESS_TOKENS [SUPPRESS_TOKENS ...]

A list of tokens that will be suppressed at generation. (default: None)

--apply_spec_augment [APPLY_SPEC_AUGMENT]

Whether to apply *SpecAugment* data augmentation to the input features. This is currently only relevant for Wav2Vec2, HuBERT, WavLM and Whisper models. (default: False)

--dataset name DATASET NAME

The name of the dataset to use (via the datasets library). (default: None)

--dataset_config_name DATASET_CONFIG_NAME

The configuration name of the dataset to use (via the datasets library). (default: None)

--overwrite_cache [OVERWRITE_CACHE]

Overwrite the cached training and evaluation sets (default: False)

--preprocessing num workers PREPROCESSING NUM WORKERS

The number of processes to use for the preprocessing. (default: None)

--max train samples MAX TRAIN SAMPLES

For debugging purposes or quicker training, truncate the number of training examples to this value if set. (default: None)

--max eval samples MAX EVAL SAMPLES

For debugging purposes or quicker training, truncate the number of evaluation examples to this value if set. (default: None)

--audio column name AUDIO COLUMN NAME

The name of the dataset column containing the audio data. Defaults to 'audio' (default: audio)

--text_column_name TEXT_COLUMN_NAME

The name of the dataset column containing the text data. Defaults to 'text' (default: text)

--max_duration_in_seconds MAX_DURATION_IN_SECONDS

Truncate audio files that are longer than `max_duration_in_seconds` seconds to 'max_duration_in_seconds` (default: 20.0)

--min_duration_in_seconds MIN_DURATION_IN_SECONDS

Filter audio files that are shorter than `min_duration_in_seconds` seconds (default: 0.0)

--preprocessing_only [PREPROCESSING_ONLY]

Whether to only do data preprocessing and skip training. This is especially useful when data preprocessing errors out in distributed training due to timeout. In this case, one should run the

preprocessing in a non-distributed setup with `preprocessing_only=True` so that the cached datasets can consequently be loaded in distributed training (default: False)

--train split name TRAIN SPLIT NAME

The name of the training data set split to use (via the datasets library).

Defaults to 'train' (default: train)

--eval_split_name EVAL_SPLIT_NAME

The name of the training data set split to use (via the datasets library).

Defaults to 'train' (default: test)

--do lower case [DO LOWER CASE]

Whether the target text should be lower cased. (default: True)

- --no_do_lower_case Whether the target text should be lower cased. (default: False)
- --language LANGUAGE Language for multilingual fine-tuning. This argument should be set for multilingual fine-tuning only. For English speech recognition, it should be set to `None`. (default: None)
- --task TASK Task, either `transcribe` for speech recognition or `translate` for speech translation. (default: transcribe)
 - --output_dir OUTPUT_DIR

The output directory where the model predictions and checkpoints will be written. (default: None)

--overwrite_output_dir [OVERWRITE_OUTPUT_DIR]

Overwrite the content of the output directory. Use this to continue training if output_dir points to a checkpoint directory. (default: False)

--do train [DO TRAIN]

Whether to run training. (default: False)

- --do eval [DO EVAL] Whether to run eval on the dev set. (default: False)
- --do_predict [DO_PREDICT]

Whether to run predictions on the test set. (default: False)

--evaluation_strategy {no,steps,epoch}

The evaluation strategy to use. (default: no)

--prediction_loss_only [PREDICTION_LOSS_ONLY]

When performing evaluation and predictions, only returns the loss.

(default: False)

--per_device_train_batch_size PER_DEVICE_TRAIN_BATCH_SIZE

Batch size per GPU/TPU core/CPU for training. (default: 8)

--per device eval batch size PER DEVICE EVAL BATCH SIZE

Batch size per GPU/TPU core/CPU for evaluation. (default: 8)

--per_gpu_train_batch_size PER_GPU_TRAIN_BATCH_SIZE

Deprecated, the use of `--per_device_train_batch_size` is preferred.

Batch size per GPU/TPU core/CPU for training. (default: None)

--per_gpu_eval_batch_size PER_GPU_EVAL_BATCH_SIZE

Deprecated, the use of `--per_device_eval_batch_size` is preferred.

Batch size per GPU/TPU core/CPU for evaluation. (default: None)

--gradient_accumulation_steps GRADIENT_ACCUMULATION_STEPS

Number of updates steps to accumulate before performing a backward/update pass. (default: 1)

--eval accumulation steps EVAL ACCUMULATION STEPS

Number of predictions steps to accumulate before moving the tensors to the CPU. (default: None)

--eval delay EVAL DELAY

Number of epochs or steps to wait for before the first evaluation can be performed, depending on the evaluation_strategy. (default: 0)

--learning_rate LEARNING_RATE

The initial learning rate for AdamW. (default: 5e-05)

--weight_decay WEIGHT_DECAY

Weight decay for AdamW if we apply some. (default: 0.0)

--adam beta1 ADAM BETA1

Beta1 for AdamW optimizer (default: 0.9)

--adam beta2 ADAM BETA2

Beta2 for AdamW optimizer (default: 0.999)

--adam epsilon ADAM EPSILON

Epsilon for AdamW optimizer. (default: 1e-08)

--max grad norm MAX GRAD NORM

Max gradient norm. (default: 1.0)

--num_train_epochs NUM_TRAIN_EPOCHS

Total number of training epochs to perform. (default: 3.0)

--max steps MAX STEPS

If > 0: set total number of training steps to perform. Override num_train_epochs. (default: -1)

-- Ir scheduler type

passive)

{linear,cosine,cosine_with_restarts,polynomial,constant,constant_with_warmup,inverse_sqrt}

The scheduler type to use. (default: linear)

--warmup_ratio WARMUP_RATIO

Linear warmup over warmup ratio fraction of total steps. (default: 0.0)

--warmup steps WARMUP STEPS

Linear warmup over warmup steps. (default: 0)

--log level {debug,info,warning,error,critical,passive}

Logger log level to use on the main node. Possible choices are the log levels as strings: 'debug', 'info', 'warning', 'error' and 'critical', plus a 'passive' level which doesn't set anything

and lets the application set the level. Defaults to 'passive'. (default:

--log level replica {debug,info,warning,error,critical,passive}

Logger log level to use on replica nodes. Same choices and defaults as ``log_level`` (default: warning)

--log_on_each_node [LOG_ON_EACH_NODE]

When doing a multinode distributed training, whether to log once per node or just once on the main node. (default: True)

--no log on each node

When doing a multinode distributed training, whether to log once per node or just once on the main node. (default: False)

--logging_dir LOGGING_DIR

Tensorboard log dir. (default: None)

--logging strategy {no,steps,epoch}

The logging strategy to use. (default: steps)

--logging first step [LOGGING FIRST STEP]

Log the first global_step (default: False)

--logging steps LOGGING STEPS

Log every X updates steps. (default: 500)

--logging_nan_inf_filter [LOGGING_NAN_INF_FILTER]

Filter nan and inf losses for logging. (default: True)

--no logging nan inf filter

Filter nan and inf losses for logging. (default: False)

--save strategy {no,steps,epoch}

The checkpoint save strategy to use. (default: steps)

--save steps SAVE STEPS

Save checkpoint every X updates steps. (default: 500)

--save total limit SAVE TOTAL LIMIT

Limit the total amount of checkpoints. Deletes the older checkpoints in the output dir. Default is unlimited checkpoints (default: None)

--save safetensors [SAVE SAFETENSORS]

Use safetensors saving and loading for state dicts instead of default torch.load and torch.save. (default: False)

--save_on_each_node [SAVE_ON_EACH_NODE]

When doing multi-node distributed training, whether to save models and checkpoints on each node, or only on the main one (default: False)

--no_cuda [NO_CUDA] Do not use CUDA even when it is available (default: False)

--use mps device [USE MPS DEVICE]

Whether to use Apple Silicon chip based 'mps' device. (default: False)

--seed SEED Random seed that will be set at the beginning of training. (default: 42)

--data seed DATA SEED

```
Random seed to be used with data samplers. (default: None)
 --jit mode eval [JIT MODE EVAL]
              Whether or not to use PvTorch jit trace for inference (default: False)
 --use ipex [USE IPEX]
              Use Intel extension for PyTorch when it is available, installation:
'https://github.com/intel/intel-extension-for-pytorch' (default: False)
 --bf16 [BF16]
                   Whether to use bf16 (mixed) precision instead of 32-bit. Requires
Ampere or higher NVIDIA architecture or using CPU (no cuda). This is an experimental
API and it may change. (default: False)
                   Whether to use fp16 (mixed) precision instead of 32-bit (default:
 --fp16 [FP16]
False)
 --fp16 opt level FP16 OPT LEVEL
              For fp16: Apex AMP optimization level selected in ['O0', 'O1', 'O2', and
'O3']. See details at https://nvidia.github.io/apex/amp.html (default: O1)
 --half precision backend (auto,cuda amp,apex,cpu amp)
              The backend to be used for half precision. (default: auto)
 --bf16 full eval [BF16 FULL EVAL]
              Whether to use full bfloat16 evaluation instead of 32-bit. This is an
experimental API and it may change. (default: False)
 --fp16 full eval [FP16 FULL EVAL]
              Whether to use full float16 evaluation instead of 32-bit (default: False)
 --tf32 TF32
                  Whether to enable tf32 mode, available in Ampere and newer GPU
architectures. This is an experimental API and it may change. (default: None)
 --local rank LOCAL RANK
              For distributed training: local rank (default: -1)
 --xpu backend {mpi,ccl,gloo}
              The backend to be used for distributed training on Intel XPU. (default:
None)
 --tpu num cores TPU NUM CORES
              TPU: Number of TPU cores (automatically passed by launcher script)
(default: None)
 --tpu metrics debug [TPU METRICS DEBUG]
              Deprecated, the use of `--debug tpu metrics debug` is preferred. TPU:
Whether to print debug metrics (default: False)
 --debug DEBUG
                      Whether or not to enable debug mode. Current options:
'underflow overflow' (Detect underflow and overflow in activations and weights),
'tpu metrics debug' (print debug metrics on TPU). (default:
 --dataloader drop last [DATALOADER DROP LAST]
```

Drop the last incomplete batch if it is not divisible by the batch size.

(default: False)

--eval_steps EVAL_STEPS

Run an evaluation every X steps. (default: None)

--dataloader_num_workers DATALOADER_NUM_WORKERS

Number of subprocesses to use for data loading (PyTorch only). 0 means that the data will be loaded in the main process. (default: 0)

--past index PAST INDEX

If >=0, uses the corresponding part of the output as the past state for next step. (default: -1)

--run_name RUN_NAME An optional descriptor for the run. Notably used for wandb logging. (default: None)

--disable_tqdm DISABLE_TQDM

Whether or not to disable the tqdm progress bars. (default: None)

--remove unused columns [REMOVE UNUSED COLUMNS]

Remove columns not required by the model when using an nlp.Dataset. (default: True)

--no_remove_unused_columns

Remove columns not required by the model when using an nlp.Dataset. (default: False)

--label names LABEL NAMES [LABEL NAMES ...]

The list of keys in your dictionary of inputs that correspond to the labels. (default: None)

--load_best_model_at_end [LOAD_BEST_MODEL_AT_END]

Whether or not to load the best model found during training at the end of training. (default: False)

--metric_for_best_model METRIC_FOR_BEST_MODEL

The metric to use to compare two different models. (default: None)

--greater_is_better GREATER_IS_BETTER

Whether the `metric_for_best_model` should be maximized or not. (default: None)

--ignore data skip [IGNORE DATA SKIP]

When resuming training, whether or not to skip the first epochs and batches to get to the same training data. (default: False)

--sharded_ddp SHARDED_DDP

Whether or not to use sharded DDP training (in distributed training only). The base option should be `simple`, `zero_dp_2` or `zero_dp_3` and you can add CPU-offload to `zero_dp_2` or

`zero_dp_3` like this: zero_dp_2 offload` or `zero_dp_3 offload`. You can add auto-wrap to `zero_dp_2` or `zero_dp_3` with the same syntax: zero_dp_2 auto_wrap` or `zero_dp_3 auto_wrap`.

(default:)

--fsdp FSDP Whether or not to use PyTorch Fully Sharded Data Parallel (FSDP) training (in distributed training only). The base option should be `full_shard`, `shard_grad_op` or `no_shard` and you can add

CPU-offload to `full_shard` or `shard_grad_op` like this: full_shard offload` or `shard_grad_op offload`. You can add auto-wrap to `full_shard` or `shard_grad_op` with the same syntax:

full_shard auto_wrap` or `shard_grad_op auto_wrap`. (default:) --fsdp min num params FSDP MIN NUM PARAMS

This parameter is deprecated. FSDP's minimum number of parameters for Default Auto Wrapping. (useful only when `fsdp` field is passed). (default: 0)
--fsdp config FSDP CONFIG

Config to be used with FSDP (Pytorch Fully Sharded Data Parallel). The value is either afsdp json config file (e.g., `fsdp_config.json`) or an already loaded json file as `dict`. (default: None)

--fsdp_transformer_layer_cls_to_wrap

FSDP_TRANSFORMER_LAYER_CLS_TO_WRAP

This parameter is deprecated. Transformer layer class name (case-sensitive) to wrap, e.g, `BertLayer`, `GPTJBlock`, `T5Block` (useful only when `fsdp` flag is passed). (default: None)

--deepspeed DEEPSPEED

Enable deepspeed and pass the path to deepspeed json config file (e.g. ds_config.json) or an already loaded json file as a dict (default: None)

--label_smoothing_factor LABEL_SMOOTHING_FACTOR

The label smoothing epsilon to apply (zero means no label smoothing).

(default: 0.0)

--optim

{adamw_hf,adamw_torch,adamw_torch_fused,adamw_torch_xla,adamw_apex_fused,adafactor,adamw_bnb_8bit,adamw_anyprecision,sgd,adagrad}

The optimizer to use. (default: adamw_hf)

--optim args OPTIM ARGS

Optional arguments to supply to optimizer. (default: None)

--adafactor [ADAFACTOR]

Whether or not to replace AdamW by Adafactor. (default: False)

--group_by_length [GROUP_BY_LENGTH]

Whether or not to group samples of roughly the same length together when batching. (default: False)

```
--length column name LENGTH COLUMN NAME
              Column name with precomputed lengths to use when grouping by
length. (default: length)
 --report to REPORT TO [REPORT TO ...]
              The list of integrations to report the results and logs to. (default: None)
 --ddp find unused parameters DDP FIND UNUSED PARAMETERS
              When using distributed training, the value of the flag
`find_unused_parameters` passed to `DistributedDataParallel`. (default: None)
 --ddp bucket cap mb DDP BUCKET CAP MB
              When using distributed training, the value of the flag 'bucket cap mb'
passed to `DistributedDataParallel`. (default: None)
 --dataloader pin memory [DATALOADER PIN MEMORY]
              Whether or not to pin memory for DataLoader. (default: True)
 --no dataloader pin memory
              Whether or not to pin memory for DataLoader. (default: False)
 --skip memory metrics [SKIP MEMORY METRICS]
              Whether or not to skip adding of memory profiler reports to metrics.
(default: True)
 --no skip memory metrics
              Whether or not to skip adding of memory profiler reports to metrics.
(default: False)
 --use legacy prediction loop [USE LEGACY PREDICTION LOOP]
              Whether or not to use the legacy prediction loop in the Trainer. (default:
False)
 --push to hub [PUSH TO HUB]
              Whether or not to upload the trained model to the model hub after
training. (default: False)
 --resume from checkpoint RESUME FROM CHECKPOINT
              The path to a folder with a valid checkpoint for your model. (default:
None)
 --hub model id HUB MODEL ID
              The name of the repository to keep in sync with the local 'output dir'.
(default: None)
 --hub strategy {end,every save,checkpoint,all checkpoints}
              The hub strategy to use when `--push to hub` is activated. (default:
every save)
 --hub token HUB TOKEN
              The token to use to push to the Model Hub. (default: None)
 --hub private repo [HUB PRIVATE REPO]
              Whether the model repository is private or not. (default: False)
```

--gradient checkpointing [GRADIENT CHECKPOINTING]

If True, use gradient checkpointing to save memory at the expense of slower backward pass. (default: False)

--include_inputs_for_metrics [INCLUDE_INPUTS_FOR_METRICS]

Whether or not the inputs will be passed to the `compute_metrics` function. (default: False)

--fp16 backend {auto,cuda amp,apex,cpu amp}

Deprecated. Use half_precision_backend instead (default: auto)

--push_to_hub_model_id PUSH_TO_HUB_MODEL_ID

The name of the repository to which push the `Trainer`. (default: None)

--push_to_hub_organization PUSH_TO_HUB_ORGANIZATION

The name of the organization in with to which push the `Trainer`.

(default: None)

--push to hub token PUSH TO HUB TOKEN

The token to use to push to the Model Hub. (default: None)

--mp parameters MP PARAMETERS

Used by the SageMaker launcher to send mp-specific args. Ignored in Trainer (default:)

--auto_find_batch_size [AUTO_FIND_BATCH_SIZE]

Whether to automatically decrease the batch size in half and rerun the training loop again each time a CUDA Out-of-Memory was reached (default: False)
--full determinism [FULL DETERMINISM]

Whether to call enable_full_determinism instead of set_seed for reproducibility in distributed training. Important: this will negatively impact the performance, so only use it for debugging.

(default: False)

--torchdynamo TORCHDYNAMO

This argument is deprecated, use `--torch_compile_backend` instead. (default: None)

--ray_scope RAY_SCOPE

The scope to use when doing hyperparameter search with Ray. By default, `"last"` will be used. Ray will then use the last checkpoint of all trials, compare those, and select the best one.

However, other options are also available. See the Ray documentation (https://docs.ray.io/en/latest/tune/api_docs/analysis.html#ray.tune.ExperimentAnalysis.g et_best_trial) for more options.

(default: last)

--ddp timeout DDP TIMEOUT

Overrides the default timeout for distributed training (value should be given in seconds). (default: 1800)

--torch_compile [TORCH_COMPILE]

If set to 'True', the model will be wrapped in 'torch.compile'. (default:

False)

--torch compile backend TORCH COMPILE BACKEND

Which backend to use with `torch.compile`, passing one will trigger a model compilation. (default: None)

--torch compile mode TORCH COMPILE MODE

Which mode to use with `torch.compile`, passing one will trigger a model compilation. (default: None)

--sortish_sampler [SORTISH_SAMPLER]

Whether to use SortishSampler or not. (default: False)

--predict_with_generate [PREDICT_WITH_GENERATE]

Whether to use generate to calculate generative metrics (ROUGE,

BLEU). (default: False)

--generation max length GENERATION MAX LENGTH

The 'max_length' to use on each evaluation loop when

`predict_with_generate=True`. Will default to the `max_length` value of the model configuration. (default: None)

--generation_num_beams GENERATION_NUM_BEAMS

The `num_beams` to use on each evaluation loop when

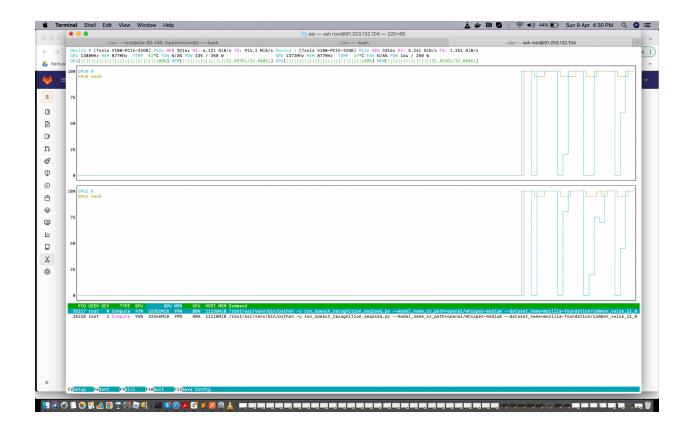
`predict_with_generate=True`. Will default to the `num_beams` value of the model configuration. (default: None)

--generation_config GENERATION_CONFIG

Model id, file path or url pointing to a GenerationConfig json file, to use during prediction. (default: None)

System Monitoring

Monitoring the GPU of Intel, nvidia - we can use a tool like htop apt install nvtop



Setup References

Installing Python, Pip and VirtualEnv Setup environment variables in Linux Install Transformers from source Create tokens for HuggingFace CLI

References

Kangri & Low Resource Language ASR

Automatic Recognition of Dialects of Himachal Pradesh Using MFCC &GMM https://ieeexplore.ieee.org/document/8988336/figures#figures

2022-01-13 - Automatic Speech Recognition for Low Resource Languages – Satwinder Singh https://www.youtube.com/watch?v=tHBtGBS60vA

Generic ASR

Speech Papers with Code for Google Fleurs

https://paperswithcode.com/dataset/fleurs

Speech-to-Text request construction

https://cloud.google.com/speech-to-text/docs/speech-to-text-requests

gcloud auth activate-service-account

https://cloud.google.com/sdk/gcloud/reference/auth/activate-service-account

Method: speech.longrunningrecognize

https://cloud.google.com/speech-to-text/docs/reference/rest/v1/speech/longrunningrecognize#Tr anscriptOutputConfig

Send a recognition request with model adaptation

https://cloud.google.com/speech-to-text/docs/adaptation

Method: projects.locations.phraseSets.create

https://cloud.google.com/speech-to-text/docs/reference/rest/v1/projects.locations.phraseSets/cr eate

Improve transcription results with model adaptation

https://cloud.google.com/speech-to-text/docs/adaptation-model#whats_next

oAuth2 Playground

https://developers.google.com/oauthplayground/?code=4/0ARtbsJp8pdfKKYHfdDD_nGOGb1G KQ1FkFCdNMHDIHtrNvYaiAkE5_XPZKkWRPKu88JIK2A&scope=https://www.googleapis.com/auth/cloud-platform

gcloud auth application-default print-access-token

https://cloud.google.com/sdk/qcloud/reference/auth/application-default/print-access-token

Authentication

https://googleapis.dev/python/google-api-core/latest/auth.html

My GCP Project

https://console.cloud.google.com/apis/credentials?authuser=4&project=warm-airline-366511&pli =1

Execute code samples

https://developers.google.com/explorer-help/code-samples

Install the Google Cloud CLI

https://cloud.google.com/sdk/docs/install-sdk

Transcribe long audio files into text

https://cloud.google.com/speech-to-text/docs/async-recognize#speech transcribe async gcs-p rotocol

M4A to WAV Converter

https://cloudconvert.com/m4a-to-wav

How Big of a Deal Is 'Whisper' for ASR and Multilingual Transcription? https://slator.com/how-big-a-deal-is-whisper-for-asr-multilingual-transcription/

How to create a speech dataset for ASR, TTS, and other speech tasks https://ogunlao.github.io/blog/2021/01/26/how-to-create-speech-dataset.html

Asr label data

https://www.google.com/search?q=asr+label+data&rlz=1C5CHFA_enIN855AE858&oq=asr+label+data&aqs=chrome..69i57j33i160l3.5844j0j7&sourceid=chrome&ie=UTF-8

Installing Whisper

https://colab.research.google.com/github/openai/whisper/blob/master/notebooks/LibriSpeech.ipynb#scrollTo=v5hvo8QWN-a9

https://usfoor.com/nvidia-riva-sets-new-bar-for-fully-customizable-speech-ai/

Benchmarking OpenAl Whisper on non-English datasets

https://blog.deepgram.com/benchmarking-openai-whisper-for-non-english-asr/

Fine Tuning Whisper

https://huggingface.co/docs/transformers/main/en/model_doc/whisper#transformers.WhisperFor ConditionalGeneration

https://huggingface.co/blog/fine-tune-whisper

https://huggingface.co/spaces/openai/whisper/discussions/6

Troubleshooting Whisper/HuggingFace Fine Tuning

https://discuss.huggingface.co/t/trainer-runtimeerror-the-size-of-tensor-a-462-must-match-the-size-of-tensor-b-448-at-non-singleton-dimension-1/26010

Hindi ASR

- https://kunal-dhawan.weebly.com/asr-system-for-hindi-language-from-scratch.html (Used Kaldi - which is in CPP)
- https://blog.deepgram.com/6-challenges-asr-hindi/ Challenges we might face
- 3. https://ohmvikrant.github.io/Hindi-ASR/, https://ohmvikrant/ASR-for-Hindi Kaldi ASR better version, 98% Accuracy
- https://www.researchgate.net/publication/260508111_Development_and_Suitability_of_I ndian_Languages_Speech_Database_for_Building_Watson_Based_ASR_System Using WATSON
- 5. https://docs.nvidia.com/deeplearning/riva/user-guide/docs/tutorials/New-language-adapt ation/Hindi/README.html
 - **NVIDIA Deep Learning**
- 6. https://sites.google.com/view/asr-challenge/leaderboard IIT Hyderabad 7% WER
- 7. https://huggingface.co/speechbrain/asr-whisper-large-v2-commonvoice-hi
- 8. https://ai4bharat.org/, https://ai4bharat.org/, https://ai4bharat.org/indicwav2vec ai4Bharat very cool stuff, lots of data
- 9. https://huggingface.co/skylord/wav2vec2-large-xlsr-hindi
- 10. https://www.cse.iitd.ac.in/~aseth/Gram Vaani ASR Challenge Interspeech.pdf

Punjabi ASR

- 11. https://ohmvikrant.github.io/Punjabi-ASR/ Kaldi ASR with very low WER
- 12. https://www.youtube.com/watch?v=tHBtGBS60vA&ab_channel=InstituteofDataScience% 28IDS%29%2CNUS

Datasets

Hindi

- 1. https://officechai.com/stories/indian-govt-releases-version-of-openais-whisper-model-which-turns-hindi-speech-into-text/
- 2. https://www.twine.net/blog/top-indian-language-datasets/
- 3. https://data.ldcil.org/hindi-raw-speech-corpus
- 4. https://ai4bharat.org/shrutilipi
- 5. GramVaani ASR Corpus https://www.openslr.org/118/
- ULCA ASR Corpus https://github.com/Open-Speech-EkStep/ULCA-asr-dataset-corpus/blob/main/LICENSE

7. Google/Fleurs https://huggingface.co/datasets/google/fleurs/viewer/hi_in/train https://huggingface.co/datasets/google/xtreme_s

Dataset References

- 1. https://huggingface.co/docs/datasets/audio_dataset
- 2. https://huggingface.co/docs/datasets/upload_dataset

Testing Data for Whisper Medium Small Fine Tuned on Hindi

- 1. Predictions https://drive.google.com/file/d/1vNGeifOx0pRpxylWiymYs0gh6NMgXZox/view?usp=shar ing
- 2. References https://drive.google.com/file/d/1regaxf53W64S7GI7uq UlpODMACR3tD3/view?usp=sha ring
- 3. WER 65% Zero shot.