

NOTE: This example is outdated and is not longer actively maintained. Please follow the new instructions of fine-tuning Wav2Vec2 [here](#)

Fine-tuning Wav2Vec2

The `run_asr.py` script allows one to fine-tune pretrained Wav2Vec2 models that can be found [here](#).

This finetuning script can also be run as a google colab [TODO: here](#).

Fine-Tuning with TIMIT

Let's take a look at the [script](#) used to fine-tune [wav2vec2-base](#) with the [TIMIT dataset](#):

```
#!/usr/bin/env bash
python run_asr.py \
--output_dir="./wav2vec2-base-timit-asr" \
--num_train_epochs="30" \
--per_device_train_batch_size="20" \
--per_device_eval_batch_size="20" \
--evaluation_strategy="steps" \
--save_steps="500" \
--eval_steps="100" \
--logging_steps="50" \
--learning_rate="5e-4" \
--warmup_steps="3000" \
--model_name_or_path="facebook/wav2vec2-base" \
--fp16 \
--dataset_name="timit_asr" \
--train_split_name="train" \
--validation_split_name="test" \
--orthography="timit" \
--preprocessing_num_workers="$(nproc)" \
--group_by_length \
--freeze_feature_extractor \
--verbose_logging \
```

The resulting model and inference examples can be found [here](#). Some of the arguments above may look unfamiliar, let's break down what's going on:

`--orthography="timit"` applies certain text preprocessing rules, for tokenization and normalization, to clean up the dataset. In this case, we use the following instance of `Orthography` :

```
Orthography(
    do_lower_case=True,
    # break compounds like "quarter-century-old" and replace pauses "--"
    translation_table=str.maketrans({"-": " "}),
)
```

The instance above is used as follows:

- creates a tokenizer with `do_lower_case=True` (ignores casing for input and lowercases output when decoding)
- replaces `"-"` with `" "` to break compounds like `"quarter-century-old"` and to clean up suspended hyphens
- cleans up consecutive whitespaces (replaces them with a single space: `" "`)
- removes characters not in vocabulary (lacking respective sound units)

`--verbose_logging` logs text preprocessing updates and when evaluating, using the validation split every `eval_steps`, logs references and predictions.

Fine-Tuning with Arabic Speech Corpus

Other datasets, like the [Arabic Speech Corpus dataset](#), require more work! Let's take a look at the [script](#) used to fine-tune [wav2vec2-large-xlsr-53](#):

```
#!/usr/bin/env bash
python run_asr.py \
--output_dir="./wav2vec2-large-xlsr-53-arabic-speech-corpus" \
--num_train_epochs="50" \
--per_device_train_batch_size="1" \
--per_device_eval_batch_size="1" \
--gradient_accumulation_steps="8" \
--evaluation_strategy="steps" \
--save_steps="500" \
--eval_steps="100" \
--logging_steps="50" \
--learning_rate="5e-4" \
--warmup_steps="3000" \
--model_name_or_path="elgeish/wav2vec2-large-xlsr-53-arabic" \
--fp16 \
--dataset_name="arabic_speech_corpus" \
--train_split_name="train" \
--validation_split_name="test" \
--max_duration_in_seconds="15" \
--orthography="buckwalter" \
--preprocessing_num_workers="$(nproc)" \
--group_by_length \
--freeze_feature_extractor \
--target_feature_extractor_sampling_rate \
--verbose_logging \
```

First, let's understand how this dataset represents Arabic text; it uses a format called [Buckwalter transliteration](#). We use the [lang-trans](#) package to convert back to Arabic when logging. The Buckwalter format only includes ASCII characters, some of which are non-alpha (e.g., `">"` maps to `"i"`).

`--orthography="buckwalter"` applies certain text preprocessing rules, for tokenization and normalization, to clean up the dataset. In this case, we use the following instance of `Orthography`:

```
Orthography(
    vocab_file=pathlib.Path(__file__).parent.joinpath("vocab/buckwalter.json"),
    word_delimiter_token="/", # "|" is Arabic letter alef with madda above
```

```

words_to_remove={"sil"}, # fixing "sil" in arabic_speech_corpus dataset
untransliterator=arabic.buckwalter.untransliterate,
translation_table=str.maketrans(translation_table = {
    "-": " ", # sometimes used to represent pauses
    "^": "v", # fixing "tha" in arabic_speech_corpus dataset
}),
)

```

The instance above is used as follows:

- creates a tokenizer with Buckwalter vocabulary and `word_delimiter_token="/"`
- replaces `"-"` with `" "` to clean up hyphens and fixes the orthography for `"ٓ"`
- removes words used as indicators (in this case, `"sil"` is used for silence)
- cleans up consecutive whitespaces (replaces them with a single space: `" "`)
- removes characters not in vocabulary (lacking respective sound units)

`--verbose_logging` logs text preprocessing updates and when evaluating, using the validation split every `eval_steps`, logs references and predictions. Using the Buckwalter format, text is also logged in Arabic abjad.

`--target_feature_extractor_sampling_rate` resamples audio to target feature extractor's sampling rate (16kHz).

`--max_duration_in_seconds="15"` filters out examples whose audio is longer than the specified limit, which helps with capping GPU memory usage.

DeepSpeed Integration

To learn how to deploy Deepspeed Integration please refer to [this guide](#).

But to get started quickly all you need is to install:

```
pip install deepspeed
```

and then use the default configuration files in this directory:

- `ds_config_wav2vec2_zero2.json`
- `ds_config_wav2vec2_zero3.json`

Here are examples of how you can use DeepSpeed:

(edit the value for `--num_gpus` to match the number of GPUs you have)

ZeRO-2:

```

PYTHONPATH=../.././src deepspeed --num_gpus 2 \
run_asr.py \
--output_dir=output_dir --num_train_epochs=2 --per_device_train_batch_size=2 \
--per_device_eval_batch_size=2 --evaluation_strategy=steps --save_steps=500 --
eval_steps=100 \
--logging_steps=5 --learning_rate=5e-4 --warmup_steps=3000 \
--model_name_or_path=patrickvonplaten/wav2vec2_tiny_random_robust \
--dataset_name=hf-internal-testing/librispeech_asr_dummy --dataset_config_name=clean \
--train_split_name=validation --validation_split_name=validation --orthography=timit \
--preprocessing_num_workers=1 --group_by_length --freeze_feature_extractor --

```

```
verbose_logging \  
--deepspeed ds_config_wav2vec2_zero2.json
```

For ZeRO-2 with more than 1 gpu you need to use (which is already in the example configuration file):

```
"zero_optimization": {  
    ...  
    "find_unused_parameters": true,  
    ...  
}
```

ZeRO-3:

```
PYTHONPATH=../../src deepspeed --num_gpus 2 \  
run_asr.py \  
--output_dir=output_dir --num_train_epochs=2 --per_device_train_batch_size=2 \  
--per_device_eval_batch_size=2 --evaluation_strategy=steps --save_steps=500 --  
eval_steps=100 \  
--logging_steps=5 --learning_rate=5e-4 --warmup_steps=3000 \  
--model_name_or_path=patrickvonplaten/wav2vec2_tiny_random_robust \  
--dataset_name=hf-internal-testing/librispeech_asr_dummy --dataset_config_name=clean \  
--train_split_name=validation --validation_split_name=validation --orthography=timit \  
--preprocessing_num_workers=1 --group_by_length --freeze_feature_extractor --  
verbose_logging \  
--deepspeed ds_config_wav2vec2_zero3.json
```

Pretraining Wav2Vec2

The `run_pretrain.py` script allows one to pretrain a Wav2Vec2 model from scratch using Wav2Vec2's contrastive loss objective (see official [paper](#) for more information). It is recommended to pre-train Wav2Vec2 with Trainer + DeepSpeed (please refer to [this guide](#) for more information).

Here is an example of how you can use DeepSpeed ZeRO-2 to pretrain a small Wav2Vec2 model:

```
PYTHONPATH=../../src deepspeed --num_gpus 4 run_pretrain.py \  
--output_dir="./wav2vec2-base-libri-100h" \  
--num_train_epochs="3" \  
--per_device_train_batch_size="32" \  
--per_device_eval_batch_size="32" \  
--gradient_accumulation_steps="2" \  
--save_total_limit="3" \  
--save_steps="500" \  
--logging_steps="10" \  
--learning_rate="5e-4" \  
--weight_decay="0.01" \  
--warmup_steps="3000" \  
--model_name_or_path="patrickvonplaten/wav2vec2-base-libri-100h" \  
--dataset_name="librispeech_asr" \  
--dataset_config_name="clean" \  
--train_split_name="train.100" \  
--preprocessing_num_workers="4" \  
--max_duration_in_seconds="10.0" \  

```

```
--group_by_length \  
--verbose_logging \  
--fp16 \  
--deepspeed ds_config_wav2vec2_zero2.json \  

```