

# Reproduce SALMONN

Speech **A**udio Language **M**usic **O**pen **N**eural **N**etwork

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- author trained SALMONN using A100-SXM-80GB
- in my case, RTX5090 32GB \* 4

## How to train a model

For SALMONN-13B v1, you need to use the following dependencies:

1. Our environment: The python version is 3.9.17, and other required packages can be installed with the following command: `pip install -r requirements.txt`.
2. Download [whisper large v2](#) to `whisper_path`.
3. Download [Fine-tuned BEATs\\_iter3+ \(AS2M\) \(cpt2\)](#) to `beats_path`.
4. Download [vicuna 13B v1.1](#) to `llama_path`.
5. Running with `python3 train.py --cfg-path configs/config.yaml` in A100-SXM-80GB.

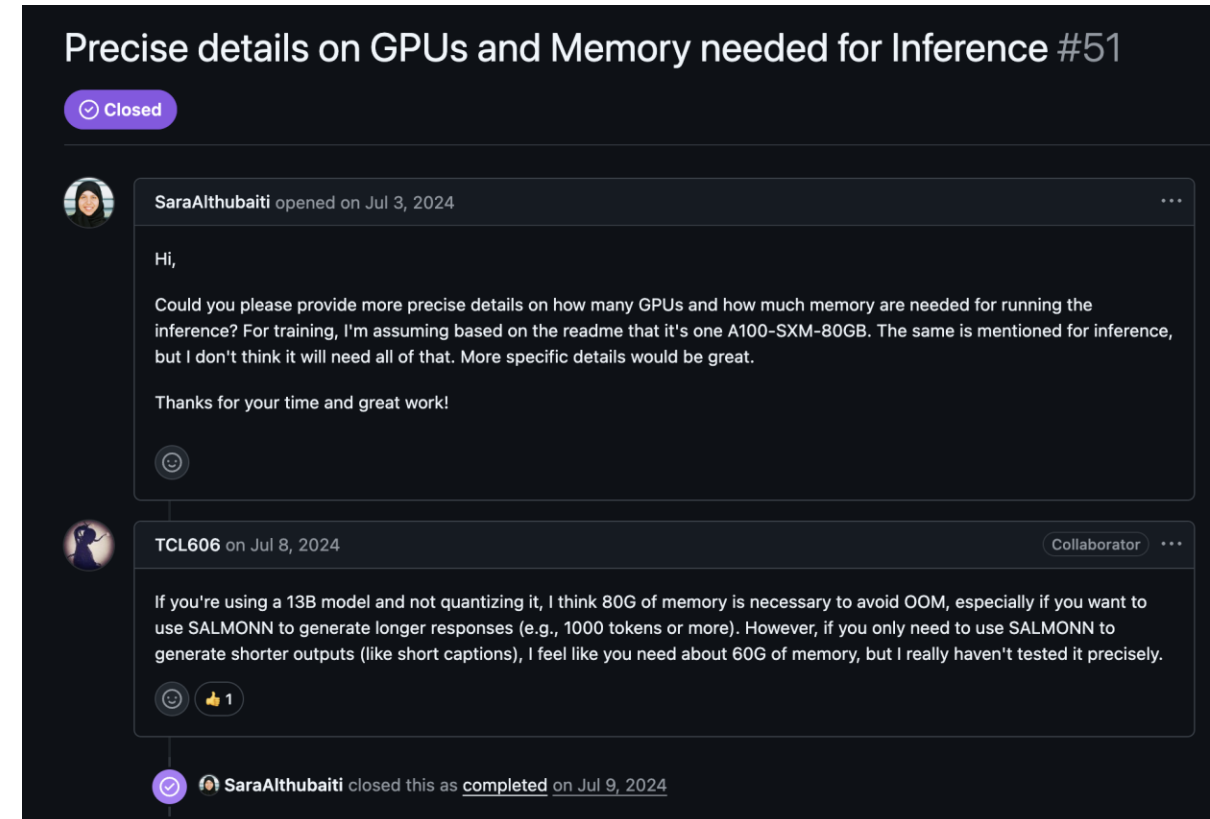
## How to inference in CLI

1. Same as **How to train a model: 1-4.**
2. Download [salmonn v1](#) to `ckpt`.
3. Running with `python3 cli_inference.py --cfg-path configs/decode_config.yaml` in A100-SXM-80GB.  
Now you can input `wav_path` and `prompt`. Enjoy yourself !

## How to launch a web demo

1. Same as **How to train a model: 1-4.**
2. Download [salmonn v1](#) to `ckpt`.
3. Running with `python3 web_demo.py --cfg-path configs/decode_config.yaml` in A100-SXM-80GB.

- no detailed environments are provided



## News

- [2024-05-28] 📦 We have released all the annotations (including 600k SQA/AQA data and 50k audio-based storytelling data) for the 3-stage training of SALMONN! Feel free to download them [here](#)!

```
335M Sep 15 16:28 salmonn_stage1_data.json
696M Sep 15 16:32 salmonn_stage2_data.json
 43M Sep 15 16:32 salmonn_stage3_data.json
```

JPONG LAB

- dataset not distinguished

```
1 # split into train, valid, test
2 ds_rate = {"train": 0.99, "valid": 0.005, "test": 0.005}
3 ds = {"train": [], "valid": [], "test": []}
4
5 for d in store_list:
6     rand = random.random()
7     if rand < ds_rate["train"]:
8         ds["train"].append(d)
9     elif rand < ds_rate["train"] + ds_rate["valid"]:
10         ds["valid"].append(d)
11     else:
12         ds["test"].append(d)
13
14 for dset in ds:
15     save_path = f"./ann/{train_set}_{dset}_ensured.json"
16     with open(save_path, "w") as f:
17         json.dump({"annotation": ds[dset]}, f)
```

- no response

AudioCaps Dataset Download Agreement

Soul National University (SNU) provides access to the AudioCaps Dataset (referred to as the Dataset) under the following conditions.

**Terms of Use**

By signing, the researcher agrees to the following terms of use:

1. SNU makes no warranties regarding the Dataset, including but not limited to being up-to-date, correct, or complete. SNU cannot be held liable for providing access to the Dataset or usage of the Dataset.
2. The Dataset should only be used for scientific or research purposes. Any other use is explicitly prohibited.
3. The Dataset must not be provided or shared in part or full with any third party.
4. The researcher takes full responsibility for usage of the Dataset at any time.
5. SNU reserves the right to terminate the researcher's access to the Dataset at any time.
6. If any part of this agreement is legally invalid, this shall not affect the remaining agreement.

Full Name: Jaewon Choi

Date: 2025. 09. 16

Signature: 지우

1

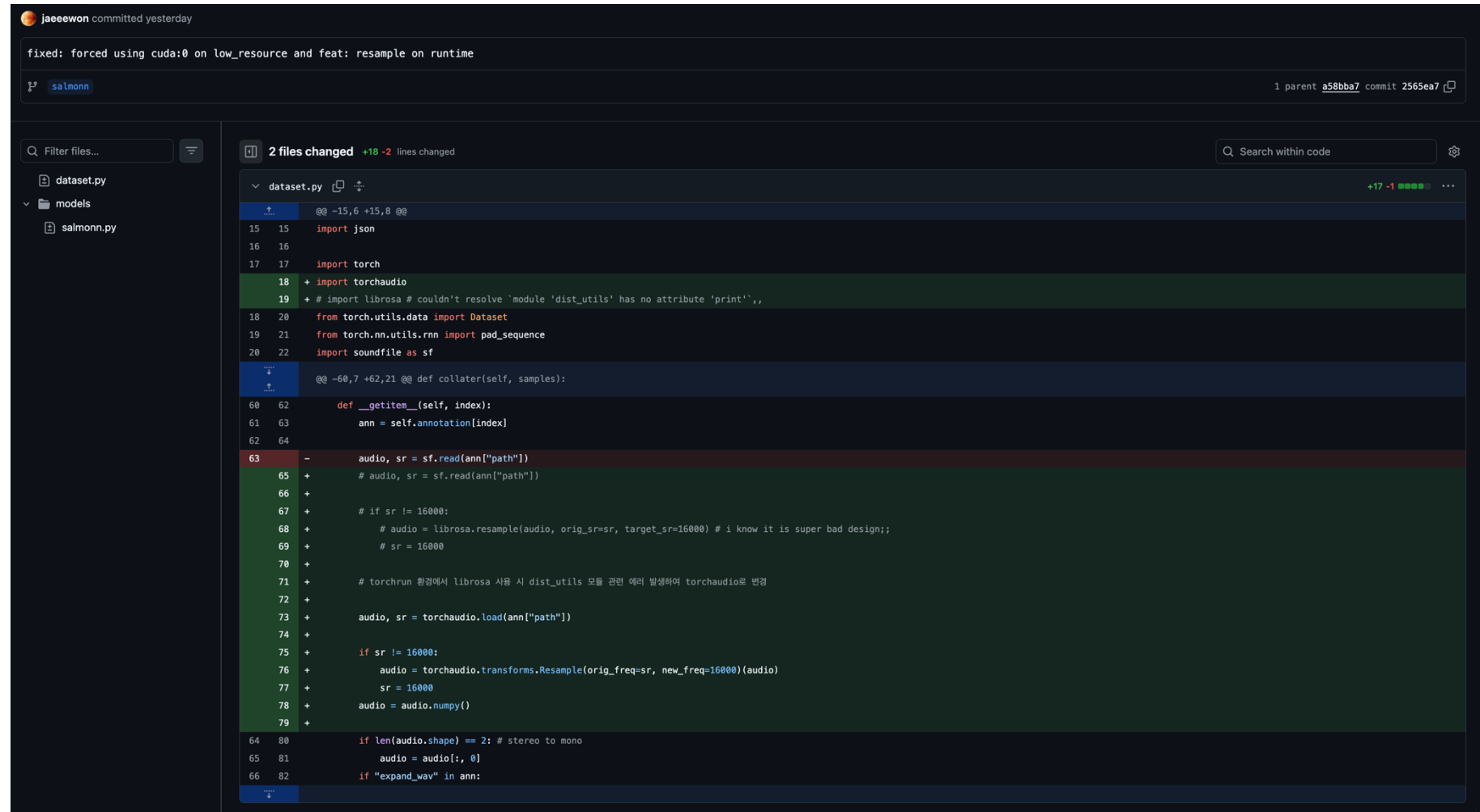


- whisper encoder force dataset to be sampled with 16k sr
- but the raw data differs between each set
- resampling is not affordable
  - resample on runtime!

```
(salmonn) jpong@hufs_5090_4ea:~/Workspace/jaeewon/repr_salmonn$ python util/anns_inspector.py
===== salmonn_stage1_data.json =====
inspecting ./ann/salmonn_stage1_data.json...: 0%|
===== [asr] LibriSpeech | /LibriSpeech/train-clean-100/103/1240/103-1240-0000.flac =====
file path: /LibriSpeech/train-clean-100/103/1240
sampling rate: 16000
channels: 1
format: FLAC (Free Lossless Audio Codec) (FLAC)
duration: 14.09s
inspecting ./ann/salmonn_stage1_data.json...: 17%|
===== [asr] GigaSpeech | /GigaSpeech/YOU0000000315_S0000660.wav =====
file path: /GigaSpeech
sampling rate: 16000
channels: 1
format: WAV (Microsoft) (WAV)
duration: 3.18s
inspecting ./ann/salmonn_stage1_data.json...: 73%|
===== [audiocaption] AudioCaps | /AudioCaps/train/r1nic0VtvkQ.wav =====
file path: /AudioCaps/train
sampling rate: 24000
channels: 1
format: WAV (Microsoft) (WAV)
duration: 6.67s
inspecting ./ann/salmonn_stage1_data.json...: 75%|
===== [audiocaption] WavCaps | /WavCaps/AudioSet_SL/YbJgb7tyh6Uk.flac =====
file path: /WavCaps/AudioSet_SL
sampling rate: 32000
channels: 1
format: FLAC (Free Lossless Audio Codec) (FLAC)
duration: 9.25s
inspecting ./ann/salmonn_stage1_data.json...: 98%|
===== [audiocaption] Clotho | /Clotho/train/Distorted AM Radio noise.wav =====
file path: /Clotho/train
sampling rate: 44100
channels: 1
format: WAV (Microsoft) (WAV)
duration: 26.16s
```

```
• (base) user@hufs:~$ df -h | grep nvme
/dev/nvme0n1p2 3.6T 3.1T 386G 90% /
```

- resample on runtime!

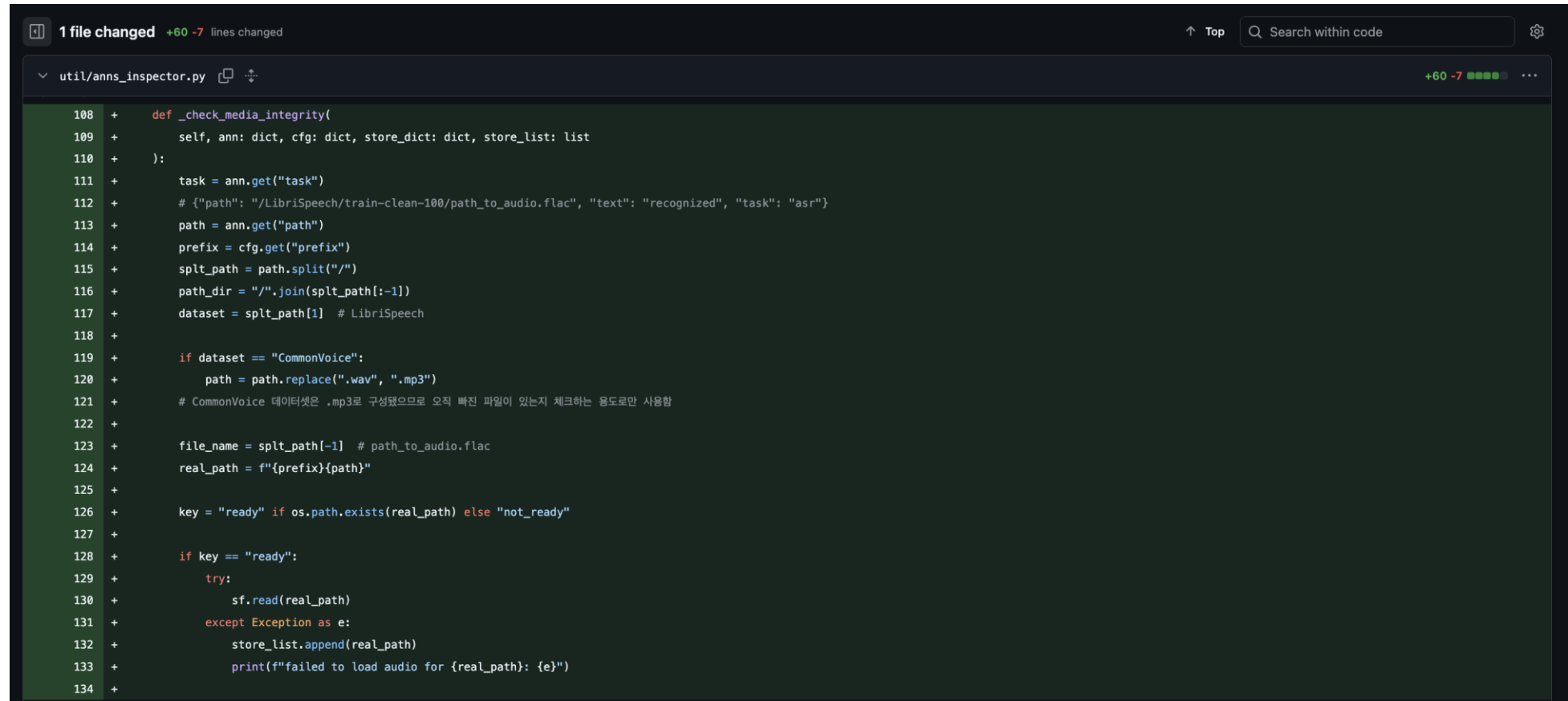


The screenshot shows a GitHub commit by user 'jaeeewon' with the message 'fixed: forced using cuda:0 on low\_resource and feat: resample on runtime'. The commit includes a file named 'dataset.py' with 2 files changed and 18 lines added, 2 lines removed. The code in 'dataset.py' shows imports for 'json', 'torch', 'torchaudio', 'librosa', 'Dataset', 'pad\_sequence', and 'soundfile'. The 'collater' method is defined, which reads audio files and resamples them to 16000 Hz using 'torchaudio.transforms.Resample' if the original sample rate is not 16000 Hz. The code also handles stereo to mono conversion and expands WAV files.

```
@@ -15,6 +15,8 @@
15 15 import json
16 16
17 17 import torch
18 + import torchaudio
19 + # import librosa # couldn't resolve 'module 'dist_utils' has no attribute 'print'.,
18 20 from torch.utils.data import Dataset
19 21 from torch.nn.utils.rnn import pad_sequence
20 22 import soundfile as sf

@@ -60,7 +62,21 @@ def collater(self, samples):
60 62     def __getitem__(self, index):
61 63         ann = self.annotation[index]
62 64
63 -         audio, sr = sf.read(ann["path"])
65 +         # audio, sr = sf.read(ann["path"])
66 +
67 +         # if sr != 16000:
68 +         #     audio = librosa.resample(audio, orig_sr=sr, target_sr=16000) # i know it is super bad design;;
69 +         #     sr = 16000
70 +
71 +         # torchrn 환경에서 librosa 사용 시 dist_utils 모듈 관련 에러 발생하여 torchaudio로 변경
72 +
73 +         audio, sr = torchaudio.load(ann["path"])
74 +
75 +         if sr != 16000:
76 +             audio = torchaudio.transforms.Resample(orig_freq=sr, new_freq=16000)(audio)
77 +             sr = 16000
78 +         audio = audio.numpy()
79 +
64 80         if len(audio.shape) == 2: # stereo to mono
65 81             audio = audio[:, 0]
66 82         if "expand_wav" in ann:
```

- invalid media crashed learning repeatedly
  - check media integrity!



```
108 + def _check_media_integrity(  
109 +     self, ann: dict, cfg: dict, store_dict: dict, store_list: list  
110 + ):  
111 +     task = ann.get("task")  
112 +     # {"path": "/LibriSpeech/train-clean-100/path_to_audio.flac", "text": "recognized", "task": "asr"}  
113 +     path = ann.get("path")  
114 +     prefix = cfg.get("prefix")  
115 +     splited_path = path.split("/")  
116 +     path_dir = "/".join(splited_path[:-1])  
117 +     dataset = splited_path[-1] # LibriSpeech  
118 +  
119 +     if dataset == "CommonVoice":  
120 +         path = path.replace(".wav", ".mp3")  
121 +         # CommonVoice 데이터셋은 .mp3로 구성됐으므로 오직 빠진 파일이 있는지 체크하는 용도로만 사용함  
122 +  
123 +     file_name = splited_path[-1] # path_to_audio.flac  
124 +     real_path = f"{prefix}{path}"  
125 +  
126 +     key = "ready" if os.path.exists(real_path) else "not_ready"  
127 +  
128 +     if key == "ready":  
129 +         try:  
130 +             sf.read(real_path)  
131 +         except Exception as e:  
132 +             store_list.append(real_path)  
133 +             print(f"failed to load audio for {real_path}: {e}")  
134 +
```

- run 4 GPU
  - deepspeed – 🤔
    - interferes autocast
  - torchrun – implemented
    - **D**istributed **D**ata **P**arallel


```
# model
self._model = model
self._model.to(self.device)
if self.use_distributed:
    self.model = DDP(
        self._model, device_ids=[self.config.config.run.gpu]
    )
else:
    self.model = self._model
```

cute typo 🤪

## experimental settings

- randomly split `salmonn_stage1_data.json` into train, validation and test set with 80:10:10 ratio
- use smaller speech model `whisper-large-v2` → `whisper-medium`
- use smaller llm `vicuna-13b-v1.1` → `vicuna-7b-v1.1`
- load llm in 8bit for low resource
- use torchrun for distributed learning

## train | 2nd epoch | crashed

```
Train: data epoch: [1] [ 285/3000] eta: 0:22:16 lr: 0.000030 loss: 2.7421 time: 0.4907 data: 0.0000 max mem: :   
[rank3]: Traceback (most recent call last):  
[rank3]:   File "/home/jpong/Workspace/jaeewon/repr_salmonn/salmonn/train.py", line 91, in <module>  
[rank3]:     main()  
[rank3]:   File "/home/jpong/Workspace/jaeewon/repr_salmonn/salmonn/train.py", line 87, in main  
[rank3]:     runner.train()  
[rank3]:   File "/home/jpong/Workspace/jaeewon/repr_salmonn/salmonn/runner.py", line 276, in train  
[rank3]:     train_stats = self.train_epoch(cur_epoch)  
[rank3]:   File "/home/jpong/Workspace/jaeewon/repr_salmonn/salmonn/runner.py", line 116, in train_epoch  
[rank3]:     samples = next(self.train_loader)  
[rank3]:   File "/home/jpong/Workspace/jaeewon/repr_salmonn/salmonn/utils.py", line 121, in __next__  
[rank3]:     data = next(self.iter_loader)  
[rank3]:   File "/home/jpong/miniconda3/envs/salmonn/lib/python3.9/site-packages/torch/utils/data/dataloader.py", line 111, in __next__  
[rank3]:     data = self._next_data()  
[rank3]:   File "/home/jpong/miniconda3/envs/salmonn/lib/python3.9/site-packages/torch/utils/data/dataloader.py", line 111, in _next_data()  
[rank3]:     return self._process_data(data, worker_id)  
[rank3]:   File "/home/jpong/miniconda3/envs/salmonn/lib/python3.9/site-packages/torch/utils/data/dataloader.py", line 111, in _process_data()  
[rank3]:     data.reraise()  
[rank3]:   File "/home/jpong/miniconda3/envs/salmonn/lib/python3.9/site-packages/torch/_utils.py", line 769, in reraise()  
[rank3]:     raise exception  
[rank3]: soundfile.LibsndfileError: <exception str() failed>
```

## result

### first epoch

train | 1st epoch | completed

```
{"train_lr": "0.000", "train_loss": "3.291"}
```

eval | 1st epoch | completed

```
{"valid_loss": 3.066974401473999, "valid_agg_metrics": 0.3930814266204834, "valid_best_epoch": 0}
```

- 
- [https://github.com/jaeeewon/repr\\_salmonn/tree/master/configs](https://github.com/jaeeewon/repr_salmonn/tree/master/configs)



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SALMONN

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Language & AI융합전공

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