HW3P2 Bootcamp

Utterance to Phoneme Mapping using Sequence Models
Spring 2022

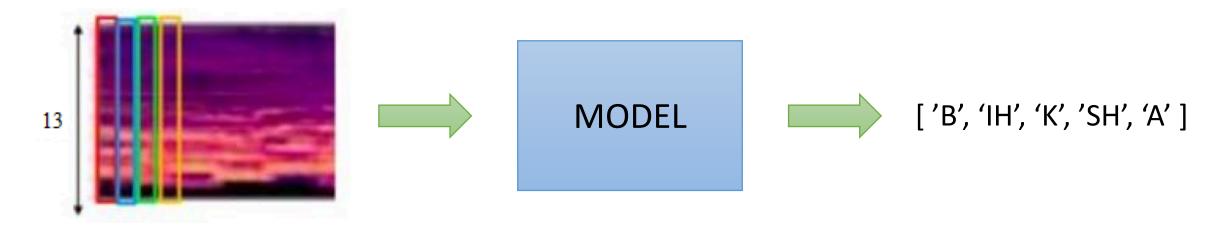
Aparajith Srinivasan

Logistics

- Early submission is due Saturday March 26th, 11:59PM ET
 - Kaggle submission a with Lev. Dist <= 30
 - Canvas MCQ
- On time submission deadline: April 7th, 11:59PM ET
- This part may not take time as much as HW2P2 for training but the high cut-off will be significantly harder
- Constrains:
 - No attention

Problem at hand

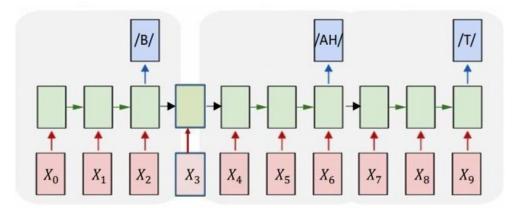
Input Utterance MFCC



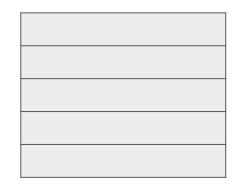
Sequence of Phonemes

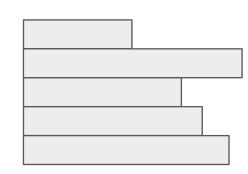
Data and Task

- Features: Same as HW1P2 (13D)
- Labels: Order synchronous but not time synchronous
- Should output sequence of phonemes
 - ['B', 'IH', 'K', 'SH', 'A'] (precisely the indexes)
- Loss: CTCLoss
- Metric: mean Levenshtein distance
 - Can import (given in starter notebook)
 - Sequence of Phonemes -> String and then calculate distance (Use PHONEMES and PHONEMES MAP)

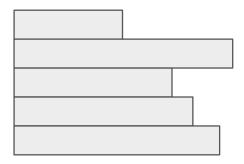


- HW1, HW2: Equal length inputs
- HW3: Variable Length sequences
- Steps:
 - Padding
 - Packing



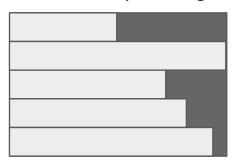


Padding

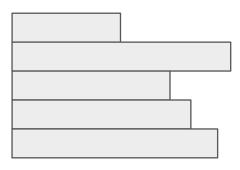


Need to store unpadded lengths as well. Have the variables *lengths_x*, *lengths_y* in the starter notebook

Padded to equal lengths

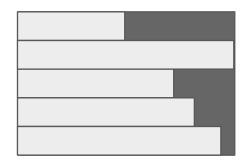


Padding



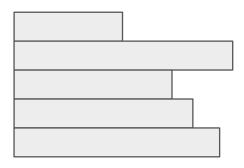
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Padded to equal lengths



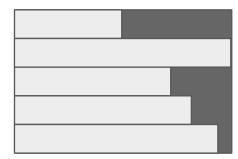
$$(B, *, 13) \rightarrow (B, T, 13)$$

Padding



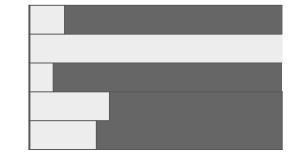
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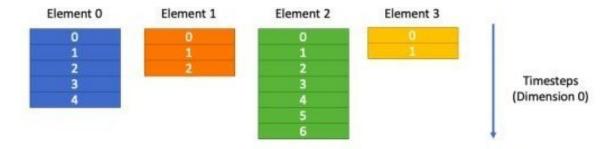




$$(B, *, 13) \rightarrow (B, T, 13)$$

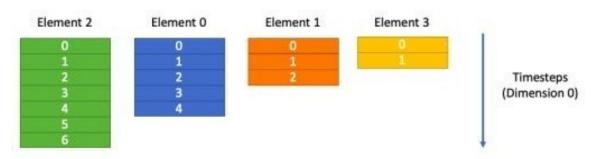
 Not for the whole dataset (instead we pack after padding)





List of Tensors to be packed. Each has same number of features but different time steps.

Figure 2: List of tensors we want to pack



Tensors sorted in descending order based on the number of time steps in each sample.

Figure 3: First we sort the list in a descending order based on number of timesteps in each

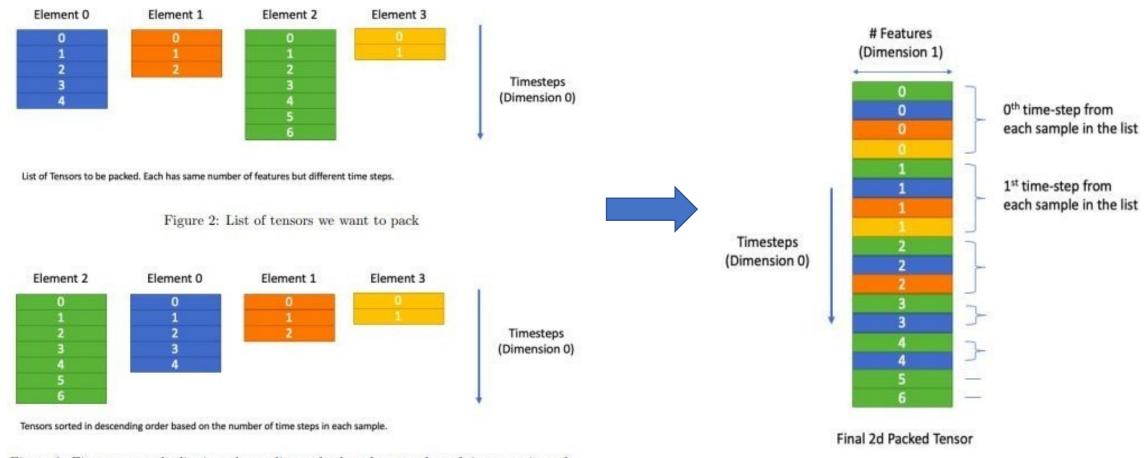
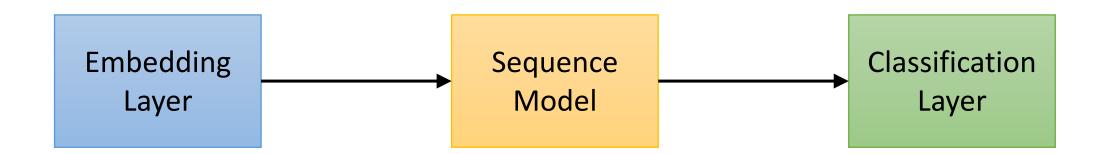


Figure 3: First we sort the list in a descending order based on number of timesteps in each

Figure 4: Final Packed 2d Tensor

Parts of a Sequence Model

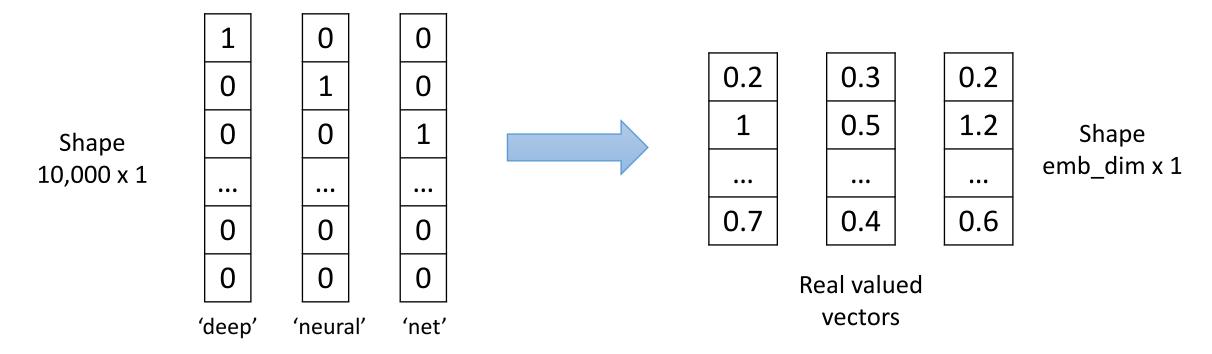


Embedding Layer

- Optional but recommended
- Used to increase/decrease the dimensionality of the input

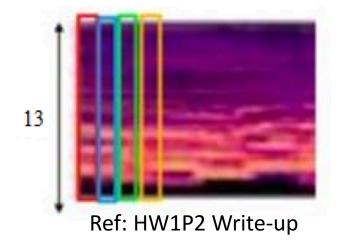
Embedding Layer

- Optional but recommended
- Used to increase/decrease the dimensionality of the input
- Eg. In NLP, 10k vocabulary represented as 1 hot vectors with 10k dim

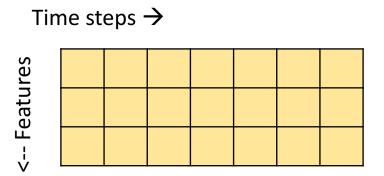


Embedding Layer

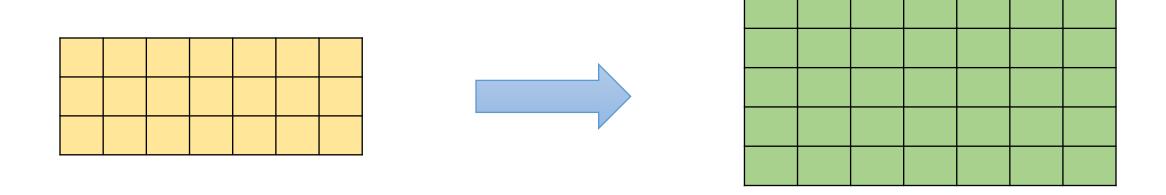
- Optional but recommended
- Used to increase/decrease the dimensionality of the input
- Our task:
 - Input dim = 13
 - Expand to emb_dim > 13 for feature extraction



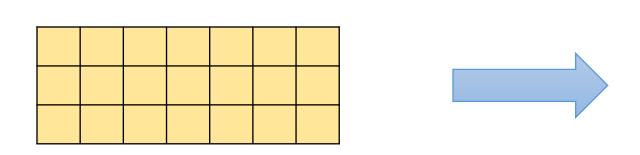
• Consider the below as an input having 3 features at each time instant

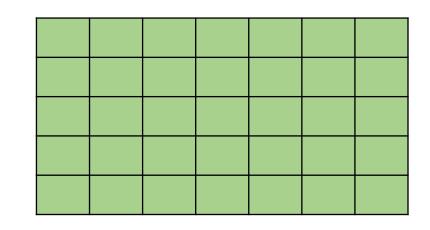


• We can use Convolution to which increases the channels of the input as we go deeper.



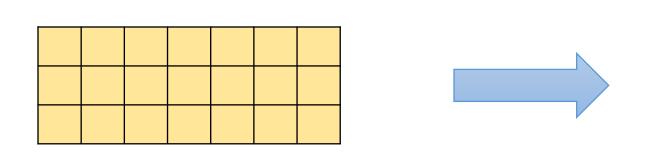
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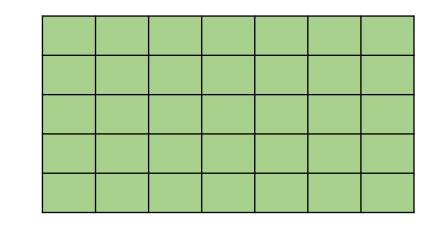




- No. Filters = 5
- Kernel= 3; Padding= 1; Stride= 1
- Kernel= 5; Padding= 2; Stride= 1
 (Or anything similar)

 We can use Convolution to which increases the channels of the input as we go deeper.





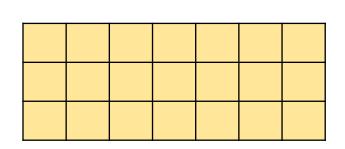
 $3D \rightarrow 5D$

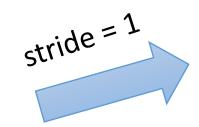
- No. Filters = 5
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 (Or anything similar)

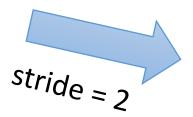
• Our input is of shape (B, T, 13) (after padding). How can we change it to (B, T, 64)?

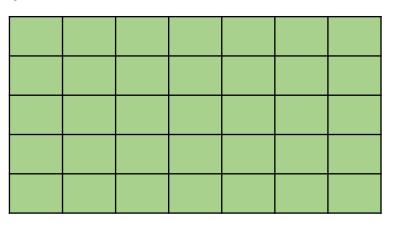
- Our input is of shape (B, T, 13) (after padding). How can we change it to (B, T, 64)?
- Transpose/Permute: (B, T, 13) → (B, 13, T) which makes #channels = 13 (Conv1d)
- Apply convolution (B, 13, T) \rightarrow (B, 64, T)
- Transpose/Permute: (B, 64, T) → (B, T, 64) (pack and pass to LSTM/GRU)
- Note: This is done in the forward function

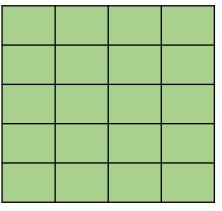
If stride > 1, we effectively reduce the time steps







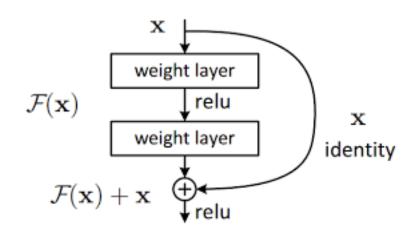




- Stride > 1 reduces computation for LSTM and training is faster.
- However, too much reduction in time steps will lead to loss of information (we don't recommend downsampling more than 4x)

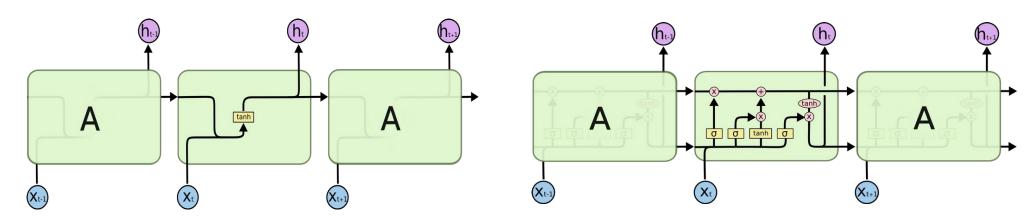
- Stride > 1 reduces computation for LSTM and training is faster.
- However, too much reduction in time steps will lead to loss of information (we don't recommend downsampling more than 4x)
- Note: Stride > 1 alters number of time steps. You need to change lengths_x accordingly
 - Use convolution formula (X K + 2*P)//S (or)
 - Clamp lengths to length of embedding (torch function)

- You can try convolution layers based on residual blocks
- Our observation: Deeper embedding layers without skip connections are not so fruitful
- Hint: Remember HW2P2!



Sequence Model

• Can use RNN, GRU, LSTM (recommended) from torch.nn



http://colah.github.io/posts/2015-08-Understanding-LSTMs/

Sequence Model

- Important parameters/hyper parameters in nn.LSTM()
 - input_size (13 or emb_size)
 - hidden_dim
 - num_layers
 - dropout
 - bidirectional
 - Note: when bidirection = True, LSTM outputs a shape of hidden_dim in the forward direction and hidden_dim in the backward direction (in total, 2*hidden_dim)

Classification Layer

- Same as HW1P2
- Output from the sequence model goes to the classification layer
- Variations
 - Deeper
 - Wider
 - Different activations
 - Dropout

• In this HW,

ARCHITECTURES >> HYPERPARAMETERS

• Don't stick with one architecture and vary the hyperparameters

*** The following suggestions might or might not work. You may want to run a proper ablation study as suggested in the previous homeworks***

Cepstral Normalization:

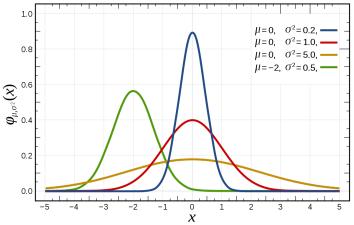
$$X \rightarrow (X - mean)/std$$

- Different weight initialization (for Conv and Linear layers)
- Weight decay with optimizer

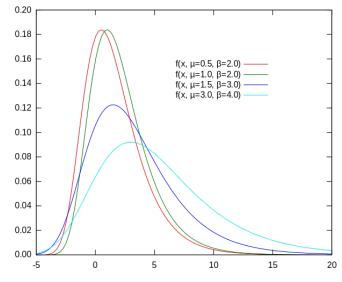
- Scheduler is very important
 - ReduceLRonPlateau (Most of our ablation)
 - Lev distance might start to oscillate at lower values
 - Can have a somewhat higher patience
 - Cosine Annealing
 - Try with higher number of epochs

- Dropout is key
 - Can use dropout in all the 3 layers: Embedding, Sequence model and classification
 - You can also start with a small dropout rate and increase after the model gets trained
- Locked Dropout for LSTM layer

- Addition of Noise (only during training)
 - Gaussian Noise
 - Gumbel Noise
- Need not add to all samples.
 Implement your module
 AddNoise(nn.module) in such a way that it adds noise to random inputs

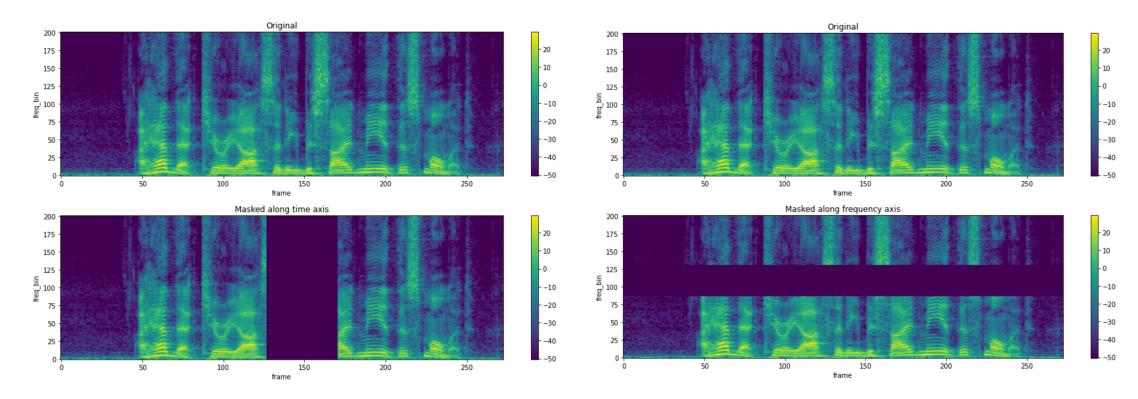


https://en.wikipedia.org/wiki/Normal_distribution



https://en.wikipedia.org/wiki/Gumbel_distribution

- Torch Audio Transforms [docs]
 - Time Masking
 - Frequency Masking



Beam width

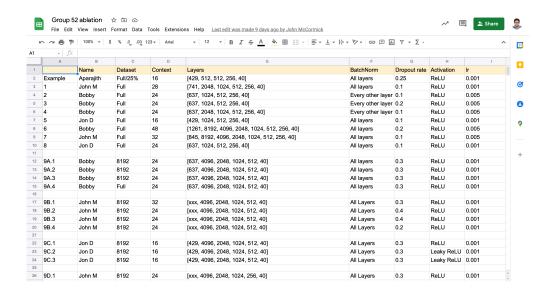
- Higher beam width may give better results (advisable to keep test beam width below 50 for computation purposes)
- Sometimes bw = 1 (greedy search) also gives good results
- Tip: Don't use a high beam width while validating in each epoch (time per epoch will be higher)

Final Tips

- More work by varying architectures
- Make proper ablation by varying just one parameter/hyperparameter to observe its influence
- Have multiple notebooks running:
 - Colab Pro users: 1 with high ram and 3 with standard ram
 - AWS: Can run multiple notebooks when some GPU memory is left
- Private leader board is worse (gives at least 0.1 higher distance than public)

Final Tips

Make sure to split work within your study groups



• Start Early - High cut-off is tougher than last homework

Medium Cut-off Architecture

Medium Cut-off Architecture

- Embedding: 2 Conv1d Layers (Final emb size 256)
- Sequence model: 4 layer Bi-directional LSTM with dropout (256)
- Classification: 2 Linear layers (2048, 41)
- Optimizer: Adam (Ir = 2e-3) with a scheduler
- Epochs: 50 100
- Beam width: 30 50 (Only for testing)

All the best!