# RNN-T Based ASR Systems

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#### Outline

• Automatic Speech Recognition (ASR) Systems

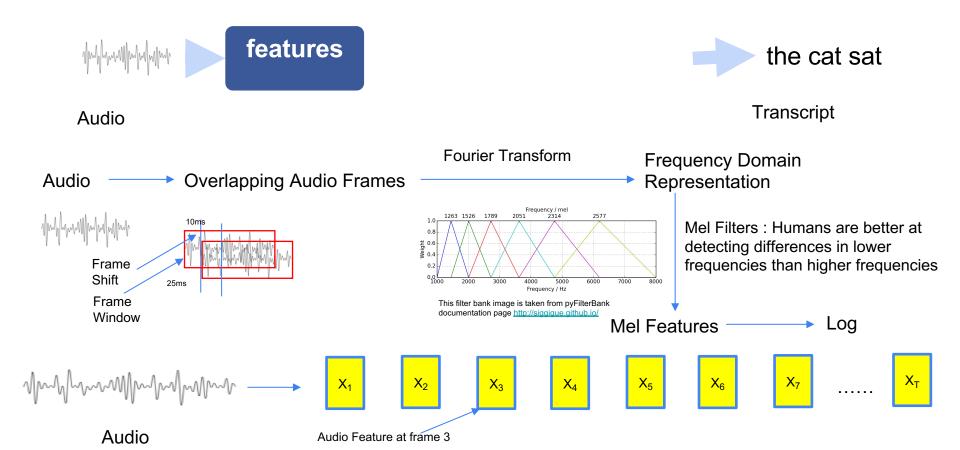
• Different Approaches For Building ASR Systems

 RNN-T (Recurrent Neural Network Transducer) Based ASR Training And Decoding

• Practical Considerations For deploying RNN-T ASR Systems in Production

Use pptx version if possible to see slides in Slide Show (Animation View) for best viewing as there are animations throughout

# Automatic Speech Recognition



LSTM image is taken from https://colah.github.io/posts/2015-08-Understanding-LSTMs/

#### ASR As Sequence To Sequence Problem

- Input Sequence: Audio frame features
- Output Sequence : Words from True Transcript Text written as letters
- Example Training Instance: Input Sequence : 7 audio frames and True Transcript Text: "BE"
- At each audio frame index, network\* generates probability of producing each output symbol. E. g. Probability of generating output symbol "K" at 1<sup>st</sup> frame.

 $P_K^1 = prob(s_1 = K \mid f_1)$ 

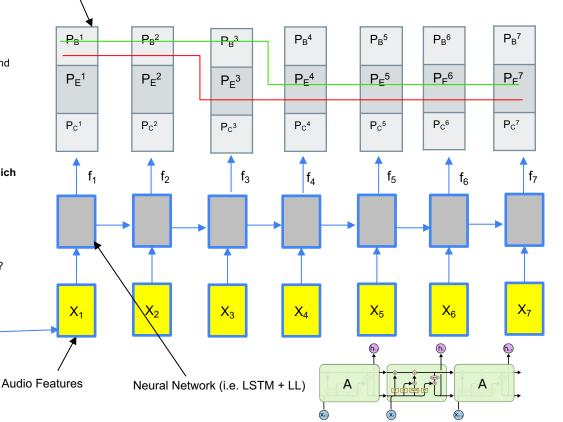
- #audio frames != # letters. We don't know alignment i.e. which portion of audio aligns to what letter in true transcript.
- Collapse continuous occurrences of same letter into one:
  - First alignment choice (BBEEEEE -> BE)
  - Second alignment choice (BBBEEEE -> BE)
  - and so on....
- How to deal with repeated letters (as in word BEE) and slience? (next slide)

MWM MWWMWW

Audio

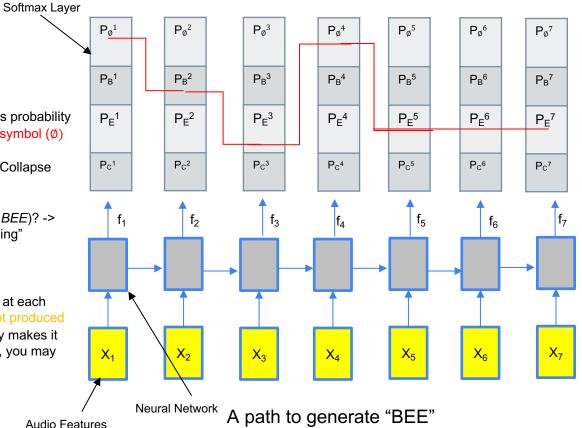
network\* -> Softmax layer and neural network that produces ft

Softmax Layer (Toy example: If we assume there are just three letters in the language (B, E, C))

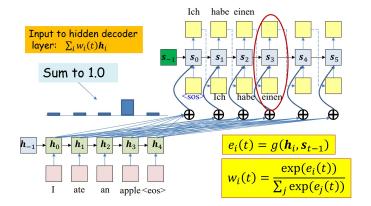


# **Connectionist Temporal Classification (CTC)**

- Input Sequence: Audio frame features
- Output Sequence: Letters
- At each audio frame index, network generates probability of producing each output symbol and blank symbol (Ø)
- How to deal with #audio frame != # letters -> Collapse letters
- How to deal with repeated letters (as in word BEE)? -> blank symbol (Ø). Ø also indicates "emit nothing" (silence). ØBEØEEØ
- Conditional Independence: Probability output at each frame does not depend on history of transcript produced so far, this non auto-regression on text history makes it less effective on learning LM information and, you may still need language model -> RNN-T.



#### ASR And Machine Translation As Sequence To **Sequence Problem** Speller (eas)

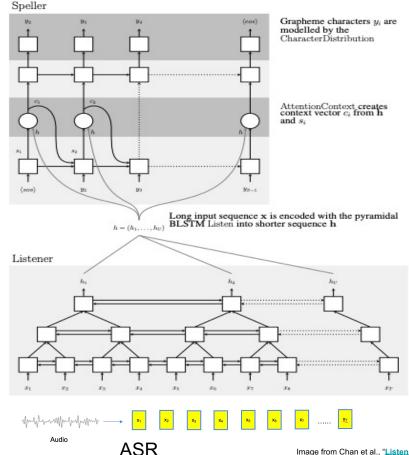


#### Machine Translation

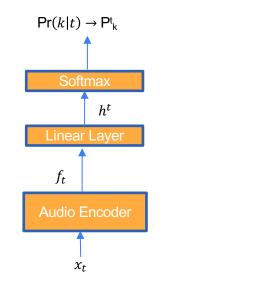
Image from Bhiksha Raj's lecture on attention models

LAS attends to all audio embeddings and uses text history produced so far to generate probability distribution over output units

$$f(y_u | y_{u-1}, y_{u-2}, ..., y_0, h_0, h_2, h_3, ..., h_v$$

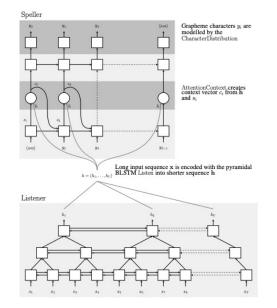


#### **Conditional Independence Assumption**



#### CTC:

 Only uses audio embedding at time "t" (x<sub>t</sub>) to generate probability distribution over output units. CTC assumes that each output unit is conditionally independent of others.



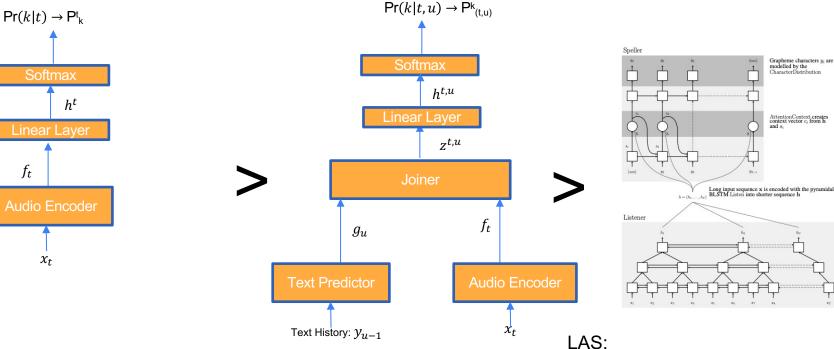
#### LAS:

Attends to all audio embeddings and uses text history produced so far to generate probability distribution over output units.

$$f(y_u | y_{u-1}, y_{u-2}, ..., y_0, h_0, h_2, h_3, ..., h_v)$$

Streaming

## **Conditional Independence Assumption**



#### RNN-T:

Uses both audio embedding at time "t"  $(x_t)$  and text history produced so far  $(y_{\mu-1})$  to generate probability distribution over output units.

Attends to all audio embeddings and uses text history produced so far to generate probability distribution over output units.

 $f(y_u|y_{u-1}, y_{u-2}, ..., y_0, h_0, h_2, h_3, ..., h_v)$ 

Streaming

Only uses audio embedding at time "t"  $(x_t)$  to

CTC assumes that each output unit is

conditionally independent of others.

generate probability distribution over output units.

CTC:

•

#### Why RNN-T ASR From Production Point Of View

- Single deployable non modularized neural model, all components of ASR in one model. ٠
- Allows compact on-device streaming ASR. Does not need a decoder graph which can be ٠ large. Unlimited words in vocab. Standard on-device ASR choice across industry.



An All-Neural On-Device Speech Recognizer Tuesday, March 12, 2019 Posted by Johan Schalkwyk, Google Fellow, Speech Team

Image from https://ai.googleblog.com/2019/03/an-all-neural-on-devicespeech.html

Achieves comparable accuracy and compute with much smaller size model compared to ٠ modularized (hybrid) systems for production when training data is the same.

Table 3. Comparison of Hybrid model with RNN-T				
Test Set	System	WER	Throughput	rtf@40
vid-clean	hybrid	14.0	55	.70
vid-clean	RNN-T	14.0	63	.60
vid-noisy	hybrid	20.7	55	.71
vid-noisy	RNN-T	21.0	65	.60

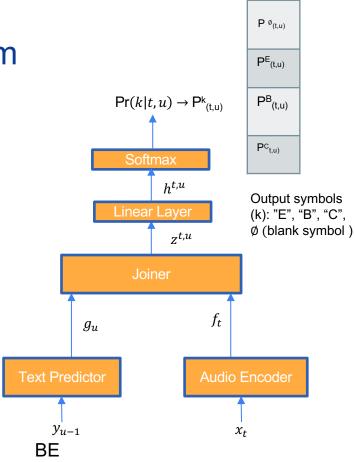
Table 2 Communication of Hadrid and John MALT

Example Study

Image from Jain et al., "RNN-T For Latency Controlled ASR With Imoroved Beam Search'

# Components Of RNN-T ASR System

- Audio Encoder : To encode sequence of audio features into audio embeddings. Long short-term memory (LSTM), B-LSTM (bi-directional LSTM), Transformer are commonly used audio encoders. Acts as acoustic model
- Text Predictor : To encode transcript produced so far into text embedding. Typically a LSTM. Acts as language model
- Joiner combines output of audio encoder and text predictor
- A linear layer followed by Softmax produces probability distribution over output units.
   Pr(k|t,u)(P<sup>k</sup><sub>(t,u</sub>) is probability of emitting "k" from (t, u).
- No collapsing of symbols: If emitted output symbols is blank (Ø) then move to next time frame else stay in same time frame. Ø also indicates "emit nothing"



# Training



# **Training Objective**

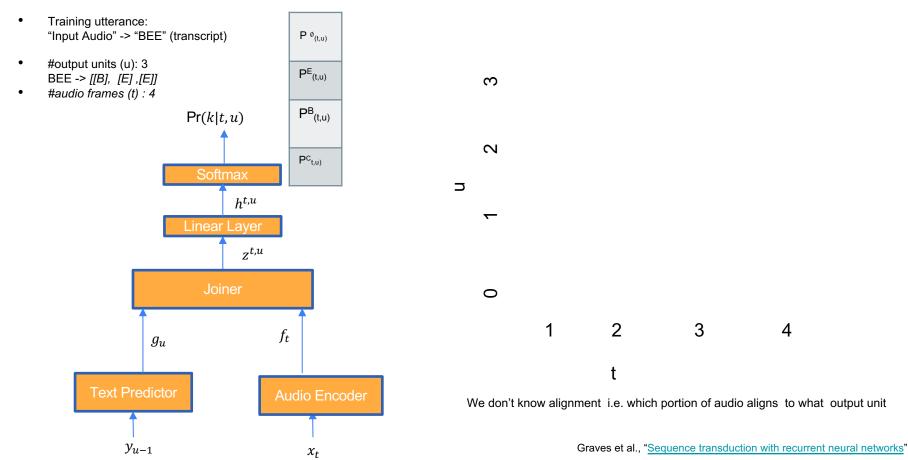
- Training utterance:
   "Input Audio" -> "BEE" (transcript)
- #output units (u): 3 BEE -> [[B], [E], [E]]
- #audio frames (t) : 4
- We don't know alignment i.e. which portion of audio aligns to what output unit (A path taken in lattice)
- Training Objective

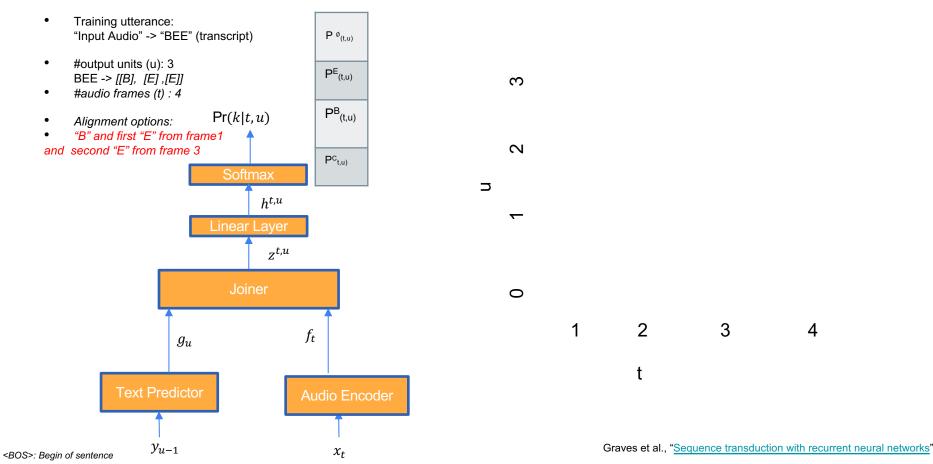
$$\begin{split} P(BEE|X) &= \sum_{\substack{alignment}} P(alignment, BEE|X) \\ P(BEE|alignment, X) &= 1 & \sum_{\substack{alignment\\alignment}} P(BEE|alignment, X) P(alignment|X) \\ \sum_{\substack{alignment}} P(alignment|X) \end{split}$$

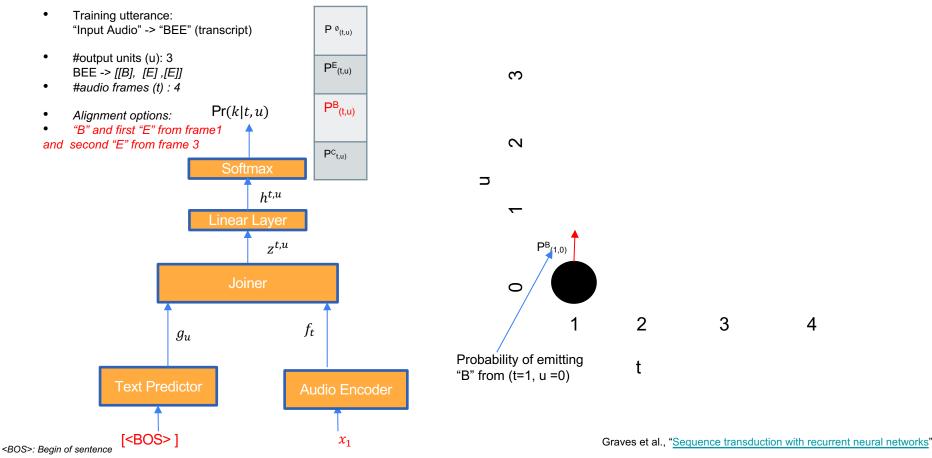


Audio

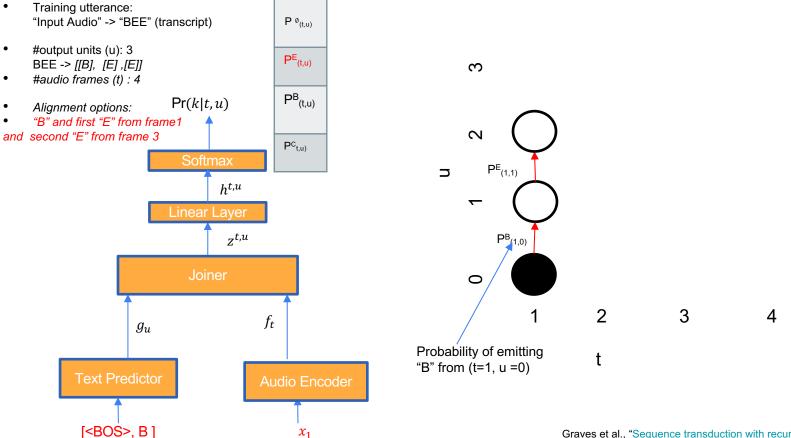
# Training

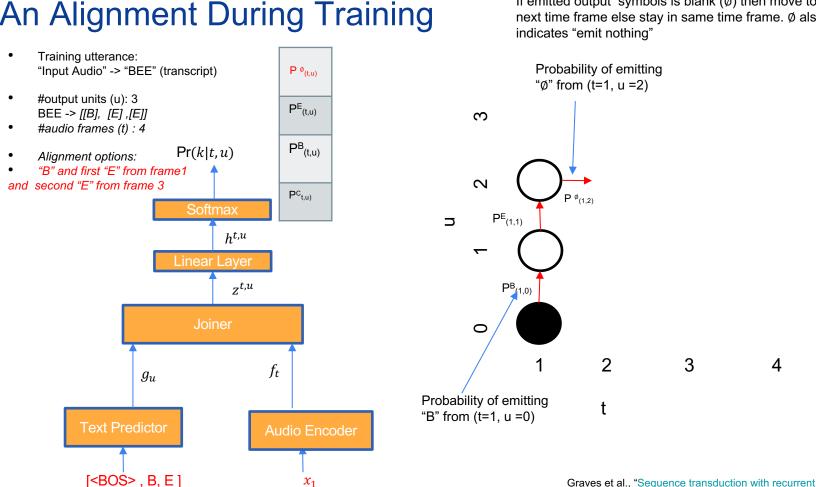




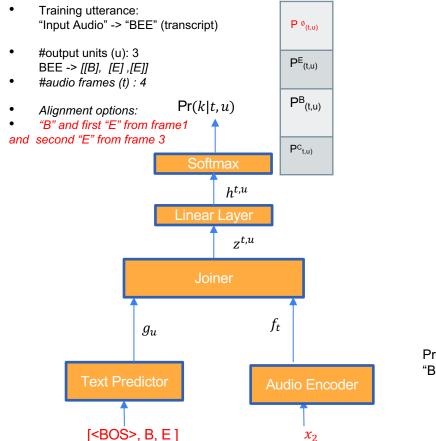


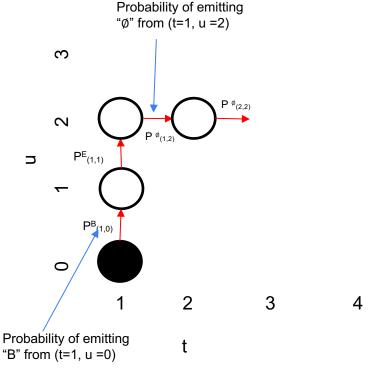
٠

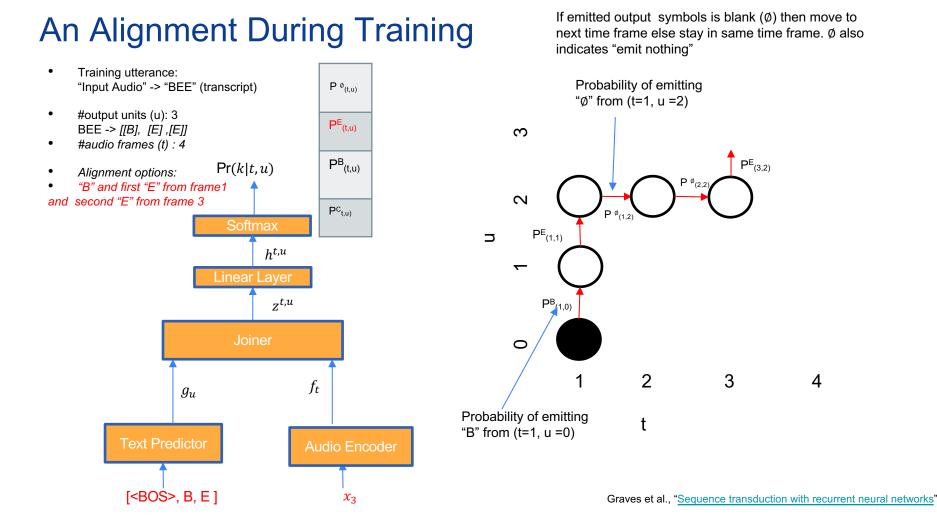


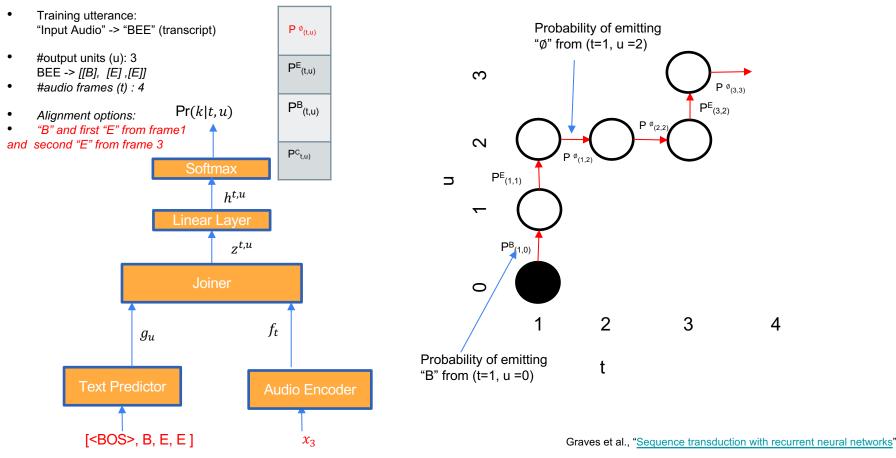


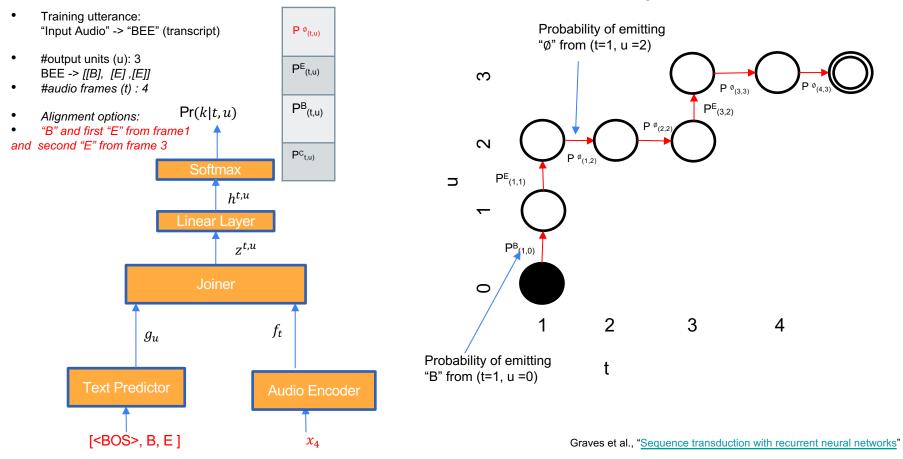
If emitted output symbols is blank (Ø) then move to next time frame else stay in same time frame. Ø also











# **RNN-T Lattice And Training**

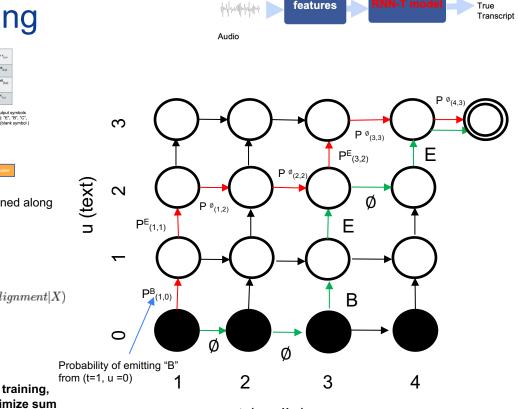
- Training utterance: "Input Audio" -> "BEE" (transcript)
- #output units (u): 3 ٠ BEE -> [[B], [E], [E]]
- #audio frames (t) : 4
- We don't know alignment i.e. ٠ which portion of audio aligns to what output unit
- Probability of alignment is multiplication of probabilities assigned along the path of alignment

 $Pr(k|t,u) \rightarrow P^{*}_{nul}$ 

Output symbols (k): "E", "B", "C", Ø (blank symbol )

$$\begin{aligned} & P(BEE|X) = \sum_{\substack{alignment}} P(alignment, BEE|X) \\ & P(BEE|alignment, X) = 1 \\ & \sum_{\substack{alignment}\\alignment}} P(BEE|alignment, X) P(alignment, X) \end{aligned}$$

- Lattice contains all valid alignment paths(traversals). During training, we change (optimize) neural network parameters to maximize sum of probabilities of all alignment paths
- Computation is done efficiently through dynamic programming (slide 51 to 56)



features

True

t (audio)

# Inference: Find a Transcript given an audio and Trained RNN-T model

Goal: Find candidate transcript of a given audio during test time. Challenge: There are infinitely many alignments that can be assigned to a test audio.



## **Operation: Extend Hypothesis**

- · Hypothesis is defined as a candidate output sequence during search
- Example Hypothesis: "EB" at time frame "t"
- Output symbols: "E", "B", "C", "Ø"

Time = t + 1

- Extend Hypothesis: Append hypothesis with each of the output symbols (k) and  $\emptyset$
- The Ø extension goes to next time frame (t + 1) and non-blank extensions remain in the same time frame (t)
- Every extended hypothesis has lower probability compared to the hypothesis it was extended from

EB

EBE

EB

**p** \* **P**<sup>Ø</sup><sub>(t, "BE")</sub>

Time = t

**p** \* **P**<sup>B</sup><sub>(t, "BE")</sub>

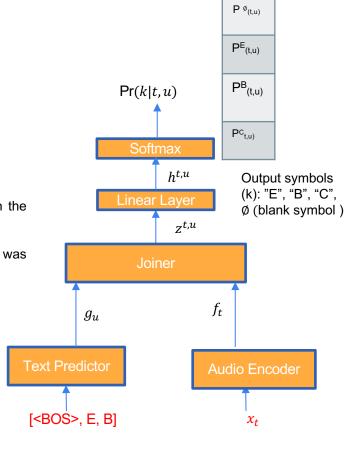
EB

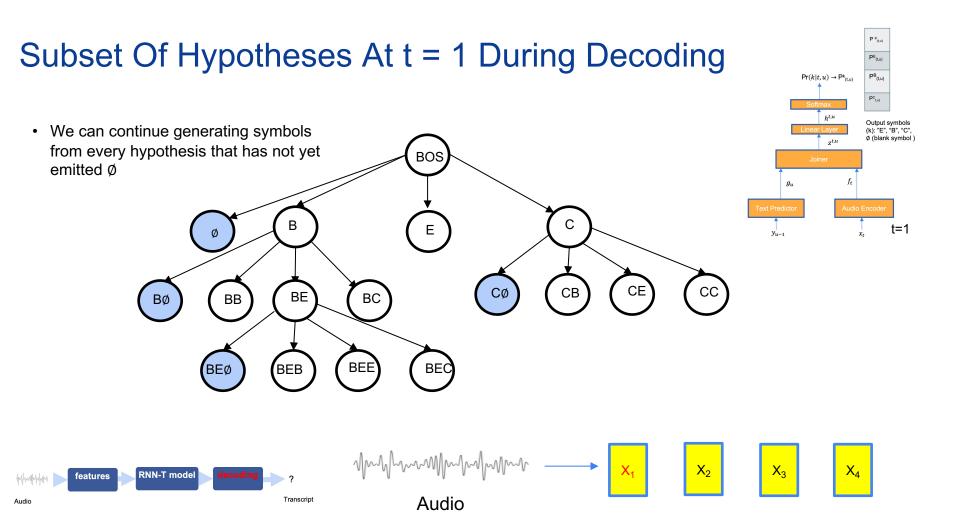
EB

**p** \* **P**<sup>E</sup><sub>(t, "BE")</sub>

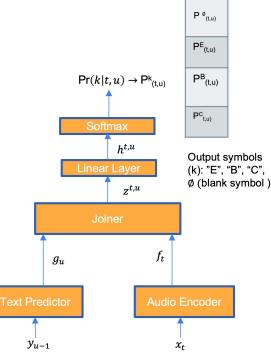
**p** \*

Р<sup>С</sup>(t, "BE")





• Goal: Get a candidate final transcript by making local optimal choice at each distinct "t" and "u"



Е 0.2 Lets make a local optimal • В 0.3 choice: If highest С 0.1 h<sup>t,u</sup> probability is given to Linear Layer blank (Ø), move to the  $z^{t,u}$ next audio frame else stay in the same audio frame Do it until all audio frames ٠ ft  $g_u$ [< *BOS* are processed Audio Encoder (E Ø [ < BOS > ] $x_1$ Mm Mmmm MM  $X_1$  $X_2$ **RNN-T model** features helpedledet. ? Transcrip Audio Audio

 $\Pr(k|t,u)$ 

probability

t = 1

В

 $X_3$ 

С

 $X_4$ 

0.4

k

Ø

Е 0.04 Lets make a local optimal ٠ В 0.3 choice: If highest С 0.01 h<sup>t,u</sup> probability is given to Linear Layer blank (Ø), move to the  $z^{t,u}$ next audio frame else stay in the same audio frame Do it until all audio frames ٠ ft  $g_u$ are processed [ < BOS ]Audio Encoder (E Ø [ < BOS > ] $x_2$ 1mhhmmm/mlm MM  $X_3$  $X_1$ **X**<sub>2</sub> features **RNN-T model** helpedledet. ? Transcrip Audio Audio

 $\Pr(k|t,u)$ 

probability

t = 2

В

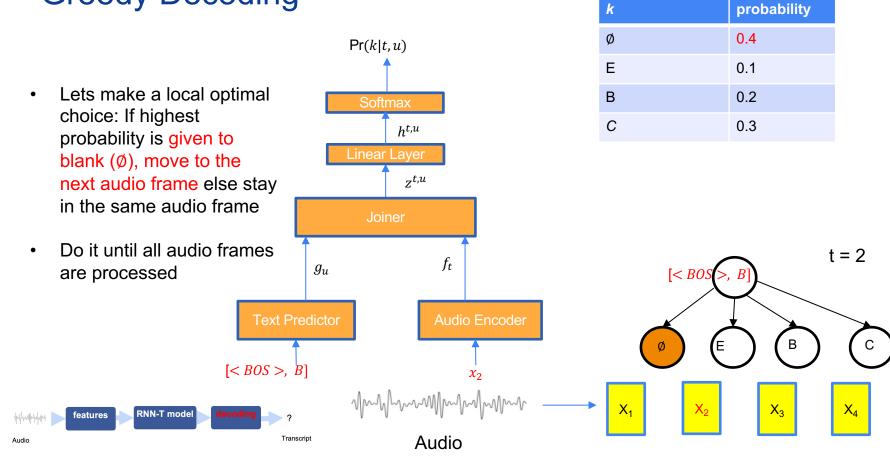
С

 $X_4$ 

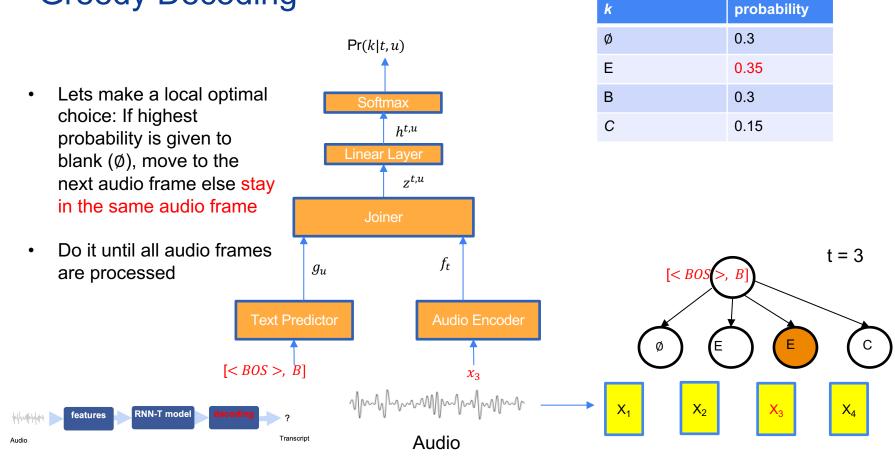
0.04

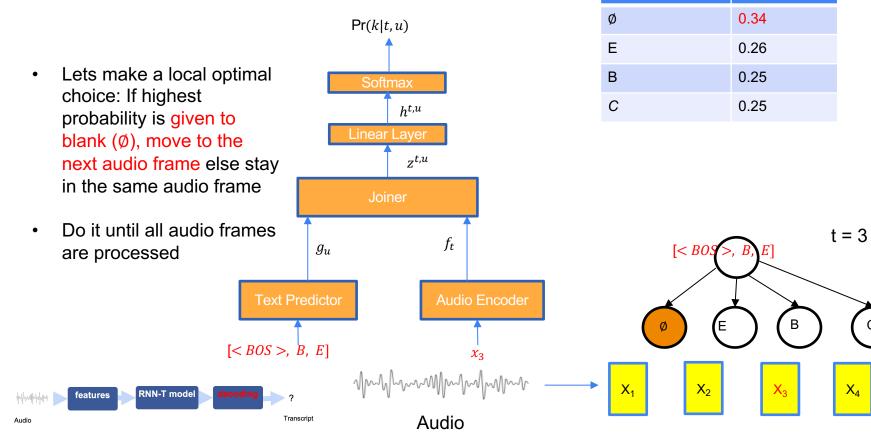
k

Ø



k

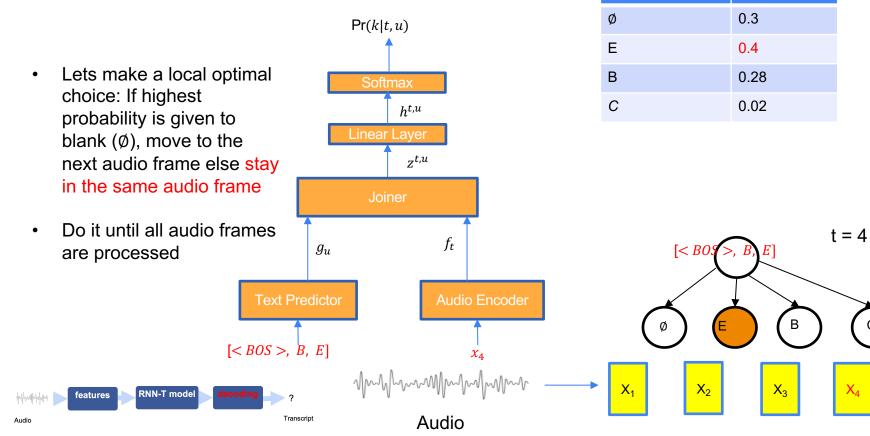




probability

С

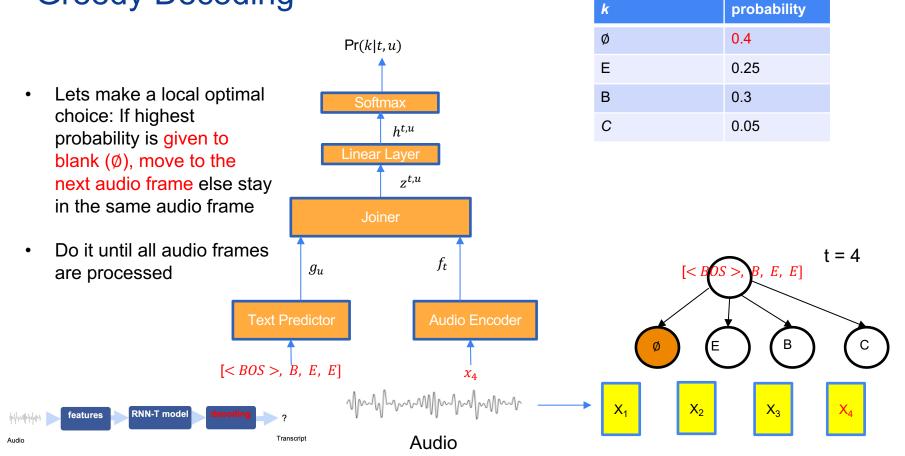
k



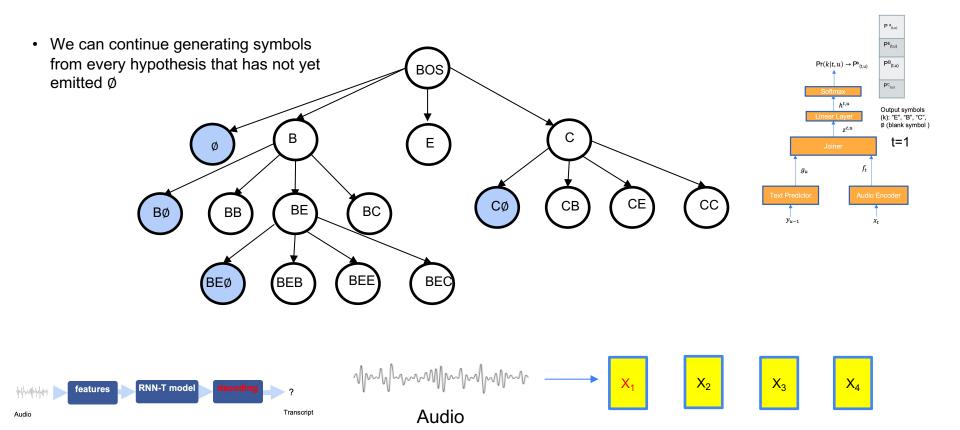
probability

С

k



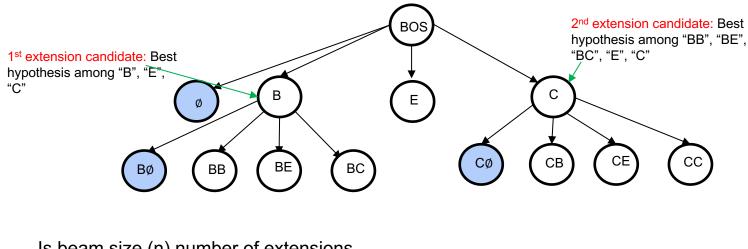
## Subset Of Hypotheses At t = 1 During Decoding





- We don't want to stay in time frame "t" forever. But we do not know when to move to "t+1" as there are infinitely many candidates that can be explored in time frame "t".
- Goal: We would like to obtain *n* candidate hypotheses at time frame "t+1" before exiting to decode at "t" which would be better than all possible future extensions at "t" that could go to "t+1".
- *n* is hyper parameter

## Beam Search: Expand the best hypothesis among candidate hypotheses



Is beam size (n) number of extensions enough? No.



MWMMWWW



 $X_1$ 





(n) "3"

P<sup>®</sup>IW PE(LU)

P<sup>B</sup><sub>0</sub>... P<sup>C</sup><sub>tab</sub>

Output symbols (k): "E", "B", "C", Ø (blank symbol

 $\Pr(k|t, u) \rightarrow \Pr_{(t,u)}^{k}$ 

ht,u

 $z^{t,u}$ 

BØ CØ

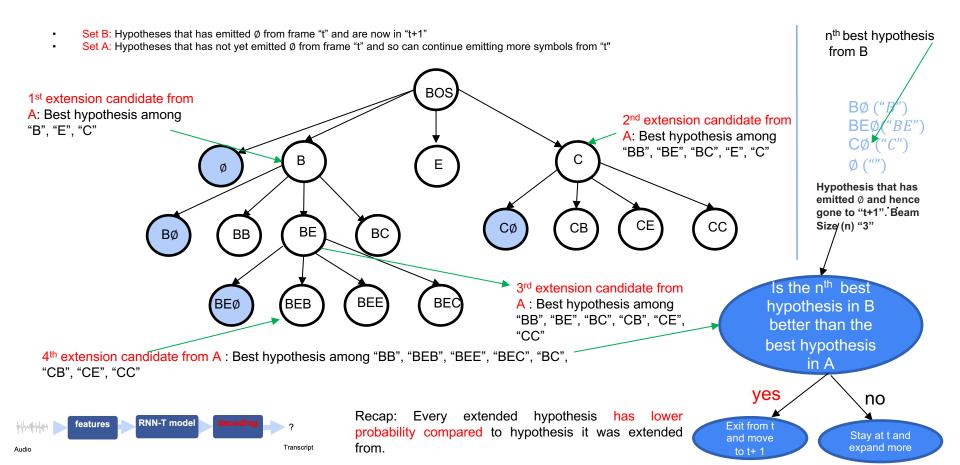
Hypothesis that has emitted Ø and hence gone to "t+1". Beam Size

x.

 $g_u$ 

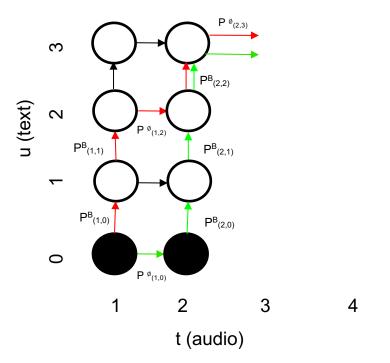
 $y_{u-1}$ 

## Beam Search: When to exit search at time frame t



# Beam Search: How To Deal With hypotheses that results in same partial transcript in B

- Combine probabilities of different alignments that result in same partial transcript
  - BB Ø B Ø (First frame emitted two Bs and Second Frame emitted one B)
  - Ø BBB Ø (First frame emitted nothing and Second Frame emitted three Bs)
  - The total probability of "BBB" at exit of time index "2" would be some of probability of alignments : BB Ø B Ø and Ø BBB Ø
    - These are two different alignments that result in "BBB" at t=2.



## How To Deal With Hypotheses That Share Prefix

- The three best hypotheses from previous time frame corresponds to following partial transcripts
   : B & BE & C
- RNN-T allows emission of any number of non blank symbols from a time frame and as hypothesis "B" from previous time frame can be extended to "BE" by emitting "E" at the current time frame, should the total probability of text sequence "BE" be contributed from both 1) BE and 2) B by emitting "E"? Yes!!
- · Add prefix accumulation for each of the alignment responsible for generating partial transcript
- We are limiting prefix accumulation to only best hypotheses kept from previous time frame, which works well in practice.

Transcrir

RNN-T mode

features

Audio

ΒØ

ΒEØ

BOS

X<sub>3</sub>

CØ

X₄

В

ΒE

X<sub>1</sub>

 $X_2$ 

### **Beam Search: Algorithm**

- Set B: Hypotheses that a blank has been output from frame t
- Set A: Hypotheses that a blank has not been output from frame t
- Take the best hypothesis from A and extend it with each of the output symbols and  $\ensuremath{\varnothing}$
- Exit the beam search at audio frame *t* if B has more than W (beam size) hypotheses that are more probable than most probable hypothesis in A.
- When starting the beam search at audio frame *t+1*: 1) Empty A, 2) Move all Hypotheses from B to A, 3) compute the prefix (*pref*) completion probability and, 4) do prefix accumulation.
- Prefix completion probability (**Pr(y**|ŷ,t)) for proper prefixes(ŷ ∈ pref(y)) of each hypothesis (y) is computed by outputting the symbols of symbol index (u) from (|ŷ| + 1) to (|y|) at audio frame (t). (slide 60 has more context about it)

$$\Pr(oldsymbol{y}|\hat{oldsymbol{y}},t) = \prod_{u=|\hat{oldsymbol{y}}|+1}^{|oldsymbol{y}|} \Pr(y_u|oldsymbol{y}_{[0:u-1]},t)$$

· Prefix accumulation for each hypothesis in A is done by following:

 $\begin{array}{l} \text{for } \boldsymbol{y} \text{ in } A \text{ do} \\ \Pr(\boldsymbol{y}) \mathrel{+}= \sum_{\hat{\boldsymbol{y}} \in pref(\boldsymbol{y}) \cap A} \Pr(\hat{\boldsymbol{y}}) \Pr(\boldsymbol{y} | \hat{\boldsymbol{y}}, t) \\ \text{end for} \end{array}$ 



Algorithm 1 Output Sequence Beam Search Initalise:  $B = \{ \boldsymbol{\varnothing} \}$ ;  $\Pr(\boldsymbol{\varnothing}) = 1$ for t = 1 to T do A = B $B = \{\}$ for y in A do  $\Pr(\boldsymbol{y}) \mathrel{+}= \sum_{\hat{\boldsymbol{y}} \in pref(\boldsymbol{y}) \cap A} \Pr(\hat{\boldsymbol{y}}) \Pr(\boldsymbol{y} | \hat{\boldsymbol{y}}, t)$ end for while B contains less than W elements more probable than the most probable in A do  $y^* = \text{most probable in } A$ Remove  $\boldsymbol{y}^*$  from A  $\Pr(\boldsymbol{y}^*) = \Pr(\boldsymbol{y}^*) \Pr(\boldsymbol{\varnothing}|\boldsymbol{y}, t)$ Add  $\boldsymbol{y}^*$  to B for  $k \in \mathcal{V}$  do  $\Pr(\boldsymbol{y}^* + k) = \Pr(\boldsymbol{y}^*) \Pr(k|\boldsymbol{y}^*, t)$ Add  $y^* + k$  to A end for end while Remove all but the W most probable from Bend for **Return:**  $\boldsymbol{y}$  with highest log  $\Pr(\boldsymbol{y})/|\boldsymbol{y}|$  in B

Transcrip

# **Practical Considerations**

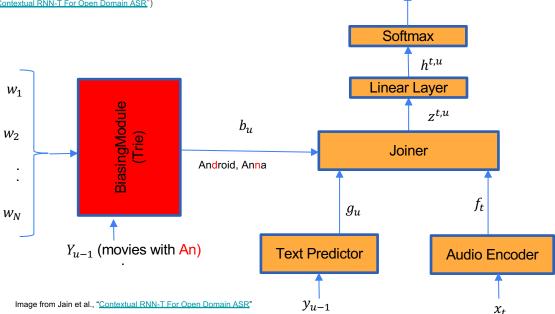
### **Utilizing Utterance Specific Context: Contextualization**

- Use utterance specific list of context words along with audio during RNN-T ASR training and/or inference
- Improve recognition of rare words for ASR systems

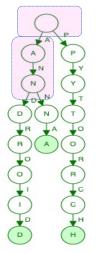
Reference Snippet		Baseline Output	Contextualization Output	Metadata Words (truncated)
its very intuitive so when	n	its very intuitive so when	its very intuitive so when	experiences, novel, PyTorch
you look at <b>PyTorch</b> itself		you look at <b>pie towards</b> itself	you look at <b>PyTorch</b> itself	updates, Facebook, machine,
				AI, language, research,
hey assistant call dharashivkar	ukar	hey assistant call <b>dharashakar</b>	hey portal call sachin dharashivkar	dharashivkar, john, kelly
ney assistant call <b>unai asinvkai</b>		ney assistant can unarasnakar	ney portai can saciiii <b>unarasiiivkar</b>	tom, singh,

## Utilizing Utterance Specific Context: Biasing Module

- Train RNN-T model using: Utterance specific context words list along with Audio, True Transcript
- We want to let model know what could be next possible output units if the next word produced in transcript was from context words list.
- **Biasing Module:** Built using Trie (data structure) of all utterance specific context words. Trie is used to find what words from context list can be finished from last unfinished word in text history so far
- Text history so far: Movies with An
- Queries:
  - Last unfinished word suffix: Android, Anna (Jain et al., "Contextual RNN-T For Open Domain ASR")



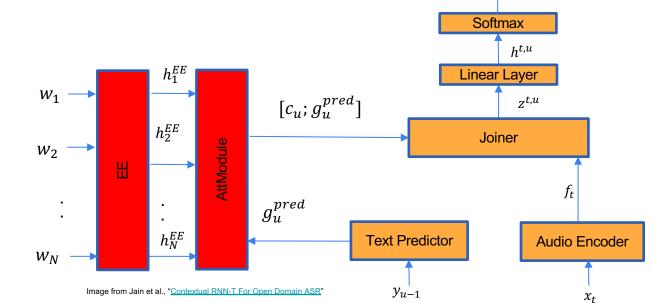
 $\Pr(k|t,u)$ 



Triie with Context Words: Android, Anna, Pytorch

### **Utilizing Utterance Specific Context: Attention Module**

- Train RNN-T model using: Utterance specific embeddings (i.e. embeddings of context words: Android, Anna, Pytorch) along with Audio, True Transcript (Jain et al., "<u>Contextual RNN-T For Open Domain ASR</u>")
- Embedding Extractor: Extracts embeddings of relevant context information. Here we are using embeddings of context words but we could use other embeddings such as visual embeddings.
   Pr(k|t, u)
- Attention Module: Attends between contextual embeddings and output of RNN-T Text Predictor



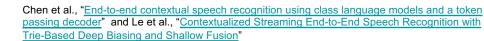
#### What Does Attention Module Learn?

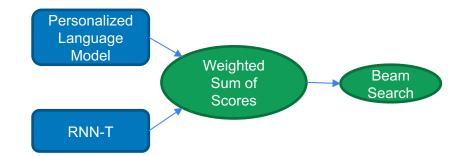


Figure 3: Visualizing attention weights,  $\alpha_{u,n}$  from Equation (5c), for the example in Table 1, row 1. The x-axis shows the target units from the hypothesis output by the Contextual RNN-T Model, and the y-axis shows the contextual metadata words ( $w_i$ ). Darker colors represent values close to zero while brighter colors represent values closer to 1.

### **Utilizing Utterance Specific Context: Shallow Fusion**

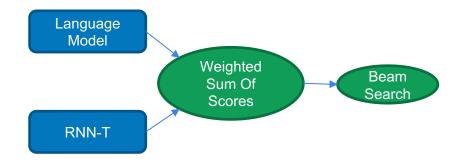
- Build a personalized Language Model (LM) using WFST
  - Patterns:
    - Hey Assistant call @entity
      - Hey Assistant call Nayan
  - In the wild messaging :
    - … @name …
      - message Nayan that I will be late...
- Language Model (LM) has same output units as RNN-T's one.
- Shallow Fusion is performed by computing a weighted sum of scores from Language Model and RNN-T model.
- Compute weighted sum of scores from RNN-T(Pr(k|t, y\*)) and personalized Language Model (Pr\_LM(k|y\*)) and use it in beam search.
- Boost occurrence of contextual entity (@entity) using Personalized Language Model (PLM)





# Contextualization: Utilizing Large Text Only data with Shallow Fusion

- Language models (LM) built with external text only data allows to model knowledge of the world into ASR system. Helpful when ASR training data is limited. Language model can estimate for example, Pr\_LM("m"] ["P", "r", "e", "s", "I", "d", "e", "n", t", " ", "B", "a", "r", "a", "c", "k", " ", "O", "b", "a"]) from text LM only data.
- Compute weighted sum of scores from RNN-T(Pr(k|t, y\*)) and Language Model (Pr\_LM(k|y\*)) and use it in beam search



## Choice Of Output Units For RNN-T ASR

- RNN-T ASR produces probability distribution over output units.
- Letters do not take co-occurrence into account.
- Words
  - Large number of output units.
  - Out of vocabulary concern for unseen words in ASR training data.
- Sentence Pieces
  - Configurable number of output units.
  - Combines co-occurring characters in single unit.
  - Example:
    - Hello World encoded as [[Hello] [\_Wor] [ld]]
  - Number of output units larger than letters but less number of runs for text predictor network.
    - If a phrase is of 11 letters but could be re-presented by 3 sentence pieces then we only need to run the text predictor network ~1/4th time.

# Optimization: Beam Search Decoding

 Limit number of expanded hypothesizes that are added in A

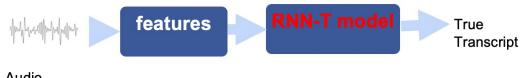
Expand best hypo from A only with the output symbols that are within **expand\_beam** value of best expansion output symbol

• Additional Conditions for exiting beam search at audio frame *t* 

Exit if the best hypothesis in B is better by more than *state\_beam* compared to best hypothesis in A

```
Algorithm 1 Improved RNNT Beam Search
Initialize: B = \{\emptyset\}; Pr(\emptyset) = 1
for t = 1 to T do
   A = B
  B = \{\}
  for y in A do
     Pr(y) + = \sum_{\hat{y} \in pref(y) \cap A} Pr(\hat{y}) Pr(y \mid \hat{y}, t)
   end for
   while B contains less than W elements more probable
   than the most probable in A do
     y^* = most probable in A
     a\_best\_prob = \max \text{ probability in } A
     b\_best\_prob = \max probability in B
     if log(b_best_prob)
                                           state_beam +
                                  >
     log(a\_best\_prob) then
        break {meet beam search and break while loop}
     end if
     Remove y^* from A
     Pr(y^*) = Pr(y^*)Pr(\emptyset \mid y, t)
     Add y^* to B
     best\_prob = \max_{k \in non\_blank} Pr(k \mid y^*, t)
     for k \in Y do
        if log(Pr(k \mid y^*, t)) \geq log(best\_prob) -
        expand_beam then
          Pr(y^* + K) = Pr(y^*)Pr(k \mid y^*, t)
          Add y^* + k to A
        end if
     end for
   end while
   Remove all but the W most probable from B
end for
return y with highest log Pr(y)/|y| in B
```

# More Details On Training **RNN-T** model

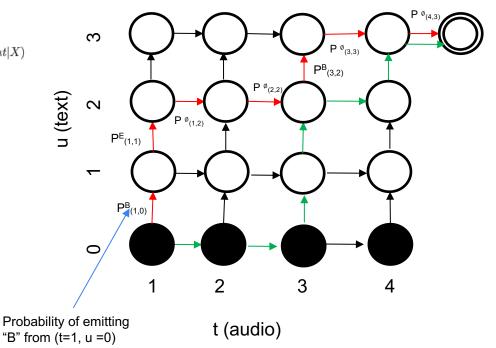


Audio

### Training: Lattice With Complete Set Of Alignments

$$\begin{split} P(BEE|X) &= \sum_{\substack{alignment\\alignment}} P(alignment, BEE|X) \\ P(BEE|alignment, X) &= 1 \\ \sum_{\substack{alignment\\alignment}} P(BEE|alignment, X) P(alignment|X) \end{split}$$

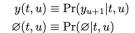
• A naïve implementation is computationally heavy: Use dynamic programming to efficiently compute this.



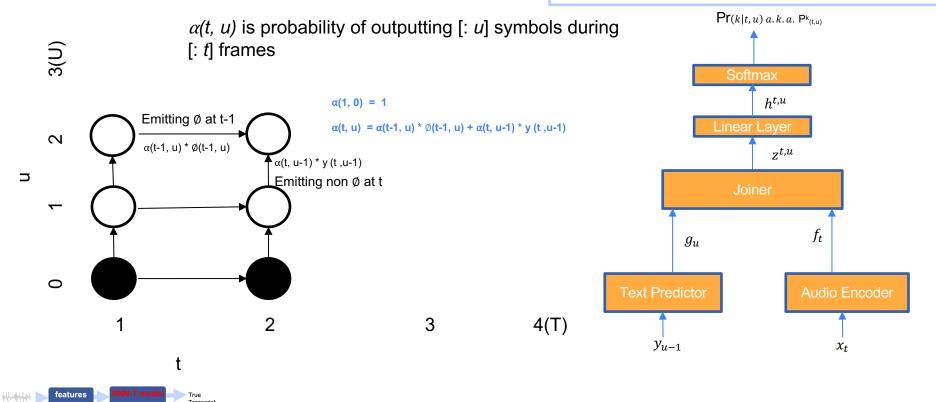


### Training: Forward Variable (alpha)

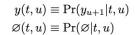
Transcript



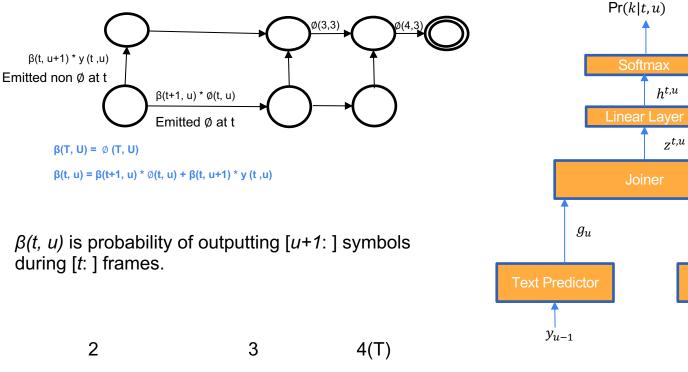
y (t,u) is probability of emitting next symbol (u + 1) in output sequence from (t, u)

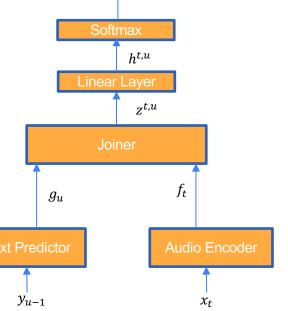


# Training: Backward Variable (beta)



y (t, u) is probability of emitting next symbol (u + 1) in output sequence from (t, u)





3(U)

2

0

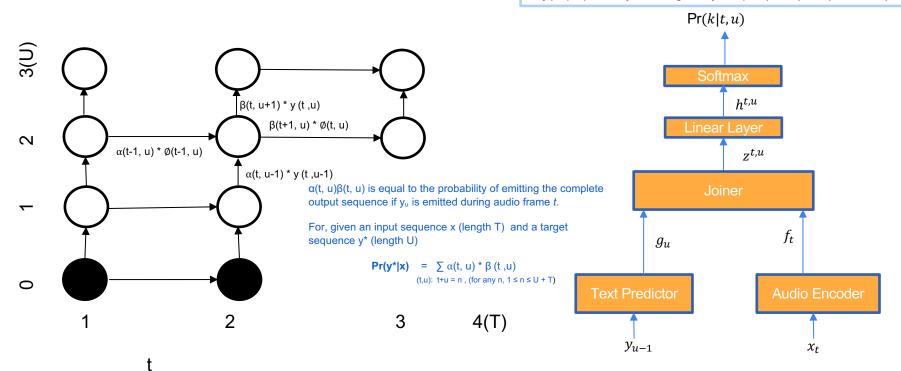
features

True Transcript

# Training: $\alpha(t, u) * \beta(t, u)$

$$y(t, u) \equiv \Pr(y_{u+1}|t, u)$$
  
 $\emptyset(t, u) \equiv \Pr(\emptyset|t, u)$ 

y (t,u) is probability of emitting next symbol (u + 1) in output sequence from (t, u)

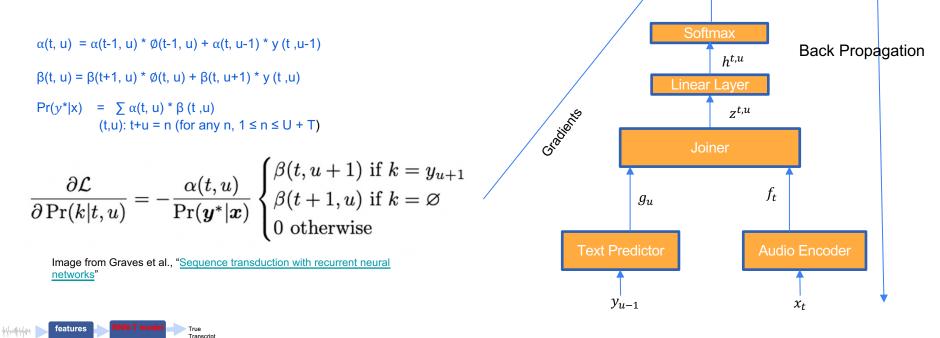


features RNN-T model True

### **Training: Gradient Descent**

For Target sequence  $y^*$ 

 $\mathcal{L} = -\ln \Pr(\boldsymbol{y}^* | \boldsymbol{x})$ 



 $y(t, u) \equiv \Pr(y_{u+1}|t, u)$ 

 $arnothing(t,u)\equiv\Pr(arnothing|t,u)$ y (t ,u) is probability of emitting next symbol (u + 1) in output sequence from (t, u)

Pr(k|t,u)

Audio

### Improving Training Optimization And Emission Delays

it

is

time

what

hev

0.0

0.5

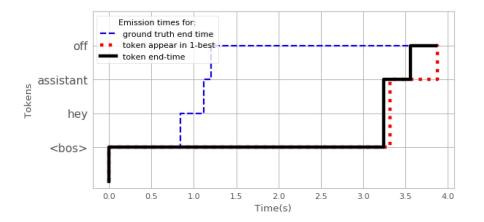
1.0

1.5

<bos>

assistant

Tokens



Token Emission Delay Problem: During inference, tokens are emitted with considerable delay after they are spoken

Restrict alignment paths: Helps in token emission delay and improving training speed.

2.0

Time(s)

2.5

Ar-RNN-T:  $b_l = 0.2$ ,  $b_r = 0.5$ Ground truth Path

3.5

40

3.0

Mahadeokar et al. "Alignment Restricted Streaming Recurrent Neural Network Transducer"

## **Further Reading**

- CTC: Graves et al., Connectionist Temporal Classification: Labelling Unsegmented Sequence Data with Recurrent Neural Networks
- RNN-T: Graves et al., <u>Sequence Transduction with Recurrent Neural Networks</u>
- On Device RNN-T ASR:
  - He et al., <u>Streaming end-to-end speech recognition for mobile devices</u>
  - Yuan et al., <u>Optimizing speech recognition for the edge</u>
- Improving RNN-T beam search
  - Jain et al., <u>RNN-T For Latency Controlled ASR With Imoroved Beam Search</u>
- Contextualization:
  - Jain et al., Contextual RNN-T For Open Domain ASR
  - Le et al., "Contextualized Streaming End-to-End Speech Recognition with Trie-Based Deep Biasing and Shallow Fusion"
- RNN-T Variants:
  - Variani et al., <u>Hybrid Autoregressive Transducer (HAT)</u>
  - Tripathi et al., Monotonic RNN-T
- Comparing RNN-T with other ASR techniques
  - Zhang et al., <u>Benchmarking LF-MMI, CTC and RNN-T Criteria for Streaming ASR</u>
  - Jain et al., <u>RNN-T For Latency Controlled ASR With Imoroved Beam Search</u>
- Sentence Piece
  - Kudo et al., <u>A simple and language independent subword tokenizer and detokenizer for Neural Text Processing</u>
- <u>Good read on tries: https://medium.com/basecs/trying-to-understand-tries-3ec6bede0014 (Vaidehi Joshi)</u>

### Thanks

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