#  for Deep Learning Based Speech Recognition 

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## Overview

Problem: Linguistic search takes over 50\% computation in Automatic Speech Recognition (ASR).

## Approach:

Reduce the Search Complexity by End-to end Modeling
Accelerate the Search Speed using Parallel Computing

Experiments \& Discussion: 5 times and 50 times speedup respectively; able to combine

## From HMM to CTC model

From HMM to CTC: do better in sequential modeling
\(\xrightarrow[\substack{Stronger classifier <br>

\& more data}]{\)|  Better sequential  |
| :---: |
|  modeling  |$}$

CTC model: learn the many-to-one function
$\square$
(a) Traditional нмм
(b) CTC
peaky distribution and concentrated information output


## Frame Sync. To Phone Sync.

## Frame synchronous Viterbi beam search in CTC

$\mathbf{w}^{*}=\underset{\mathbf{w}}{\operatorname{argmax}}\{P(\mathbf{w}) p(\mathbf{x} \mid \mathbf{w})\}=\underset{\mathbf{w}}{\operatorname{argmax}}\left\{P(\mathbf{w}) p\left(\mathbf{x} \mid \mathbf{l}_{\mathbf{w}}\right)\right\}$
$=\underset{\mathbf{w}}{\mathbf{w}}\left\{P(\mathbf{w}) \max _{\mathbf{1}_{\mathbf{w}}} \frac{P\left(\mathbf{1}_{\mathbf{w}} \mid \mathbf{x}\right)}{P\left(\mathbf{l}_{\mathbf{w}}\right)}\right\}$
$\cong \underset{\mathbf{w}}{\operatorname{argmax}}\left\{P(\mathbf{w}) \max _{\pi: \pi \in L^{\prime}, \mathcal{B}\left(\pi_{1: T}\right)=\mathbf{l}_{\mathbf{w}}} \frac{1}{P\left(\mathbf{l}_{\mathbf{w}}\right)} \prod_{t=1}^{T} y_{\pi_{t}}^{t}\right\}$
$\pi_{1: T}=\left(\pi_{1}, \ldots, \pi_{T}\right)$ is the frame-wise decoding path
$\mathrm{l}_{\mathrm{w}}$ is phone sequence corresponding to w in dictionary
$l \in L$ and $L$ is the phone se
Frame synchronous to phone synchronous decoding $\mathbf{w}^{*} \cong \underset{\mathbf{w}}{\operatorname{argmax}}\left\{P(\mathbf{w}) \max _{\pi: \pi \in L^{\prime}, \mathcal{B}\left(\pi_{1: T}\right)=\mathbf{l}_{\mathbf{w}}} \frac{1}{P\left(\mathbf{l}_{\mathbf{w}}\right)}\{\right.$

$U=\left\{u: y_{\mathrm{blank}}^{u} \simeq 1\right\}$ is the set of common blank time indexes $J=T-|U| \quad$ is the number of output phone labels

## Parallel Viterbi Decoding



Three levels of parallelism: future, history, utterance


Atomic Token Recombination


- Dynamic Load Balancing


Lattice Processing

- Linkedlist $\rightarrow$ vector
- Atomic operations e.g. memory allocation
- Parallel lattice pruning


## Experiments

Experimental Setup
Switchboard 300 hours corpus, Cross Entropy \& LF-MMI acoustic models (AM)
30 k -vocabulary, several tri-gram language models (LM)
Baseline: Kaldi 1 -best decoder, Kaldi lattice decoder
GPU Optimization: Fast memcpy; merge GPU kernels by adding grid sync.; etc. (rel. 20\% speedup)

| subset | performance$\mathrm{FSD} \mapsto \mathrm{PSD}$ |  | search speed-up |  |  |  |
| :---: | :---: | :---: | :---: | :---: | :---: | :---: |
|  |  |  | FSD $\mapsto$ PSD |  | FSD $\mapsto$ PSD |  |
|  | WER | $\Delta(\%)$ | SRTF | $\Delta(\%)$ | \#AT | $\Delta(\%)$ |
| swb | 18.7 | +0.5 | 0.075 | $-717$ | 2221 | -77 |
| callhm | 33.3 | +0.0 | 0.073 | -70 | 2211 | -77 |



Table 2: Speedup of the Proposed Method (beam=14).

| system | 1-best |  | + lattice |  |
| :---: | :---: | :---: | :---: | :---: |
|  | RTF | $\Delta$ | RTF | $\Delta$ |
| CPU | 0.16 | 1.0X | 0.27 | 1.0X |
| +8 -sequence (1 socket) | - | - | 0.15 | 1.8X |
| GPU | 0.016 | 10X | 0.080 | 3.3X |
| + atomic operation | 0.015 | 11X | 0.077 | 3.5 X |
| + dyn. load balancing | 0.011 | 15X | 0.075 | 3.6X |
| + lattice prune | - | - | 0.028 | 9.7X |
| +8 -sequence (MPS) | 0.0035 | 46X | 0.0080 | 34X |

- 34 times speedup from parallel computing


Varieties of GPU arch., WFST sizes and acoustic models.

## Conclusions

General speedup of linguistic search in speech recognition
End-to-end Modeling


Parallel Computing
Future works:
Inspire more researches in GPU decoding
Combination of Both Techniques

