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Blizzard 2014 spoke task submission: Dual acoustic models and probabilistic cross-lingual speaker adaptation

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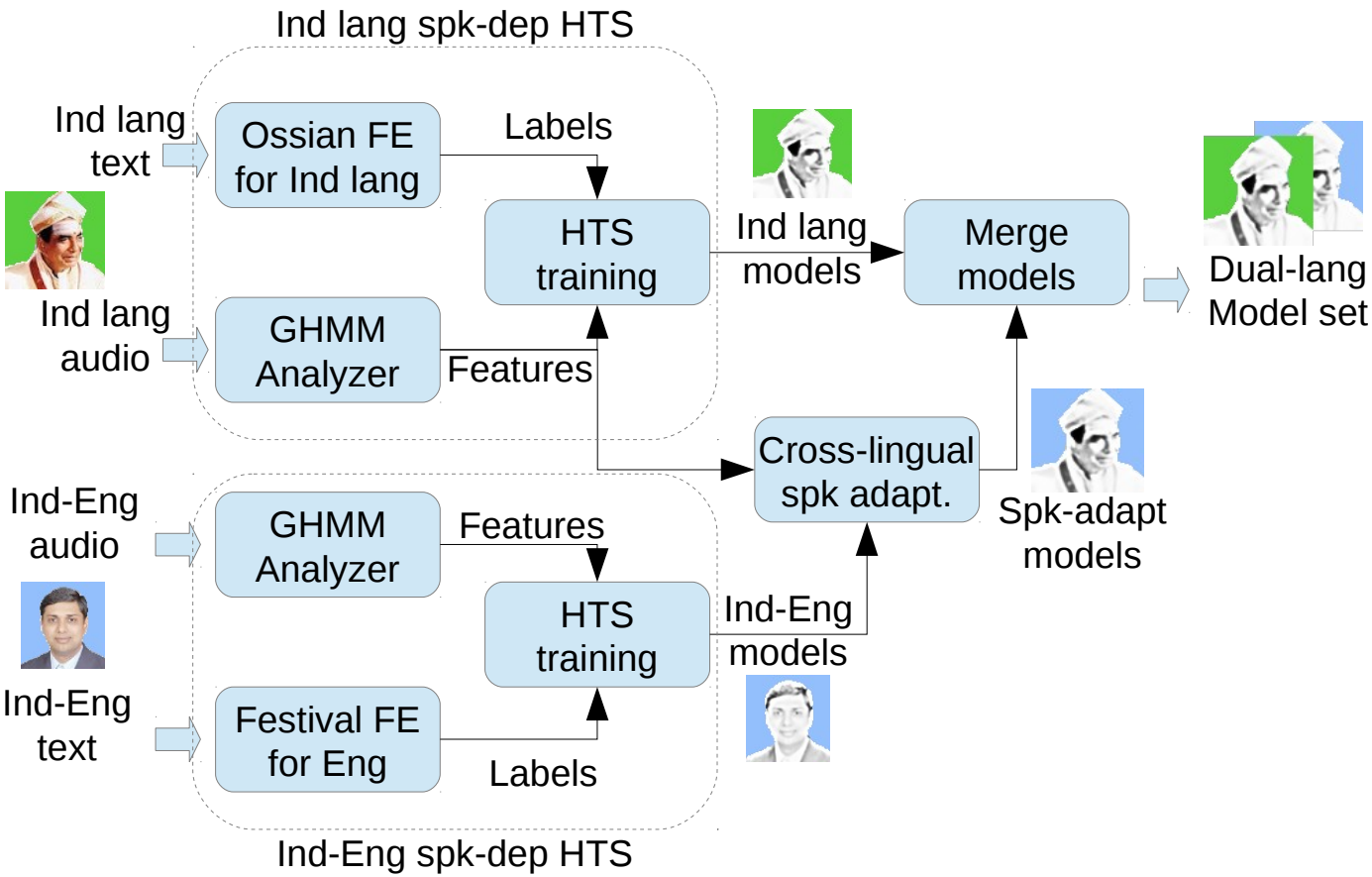
Outline

- Blizzard 2014 Spoke Task - Overview
- Dual language synthesis – overview
- Label generation
  - Method 1: Transcribe English in Indian script
  - Method 2: Dual front-end with filler words
- Acoustic model training
  - Speaker-dependent models
  - Cross-lingual adaptation
- Model set merging and synthesis
- Spoke Task - Demo

Blizzard 2014: Spoke Task

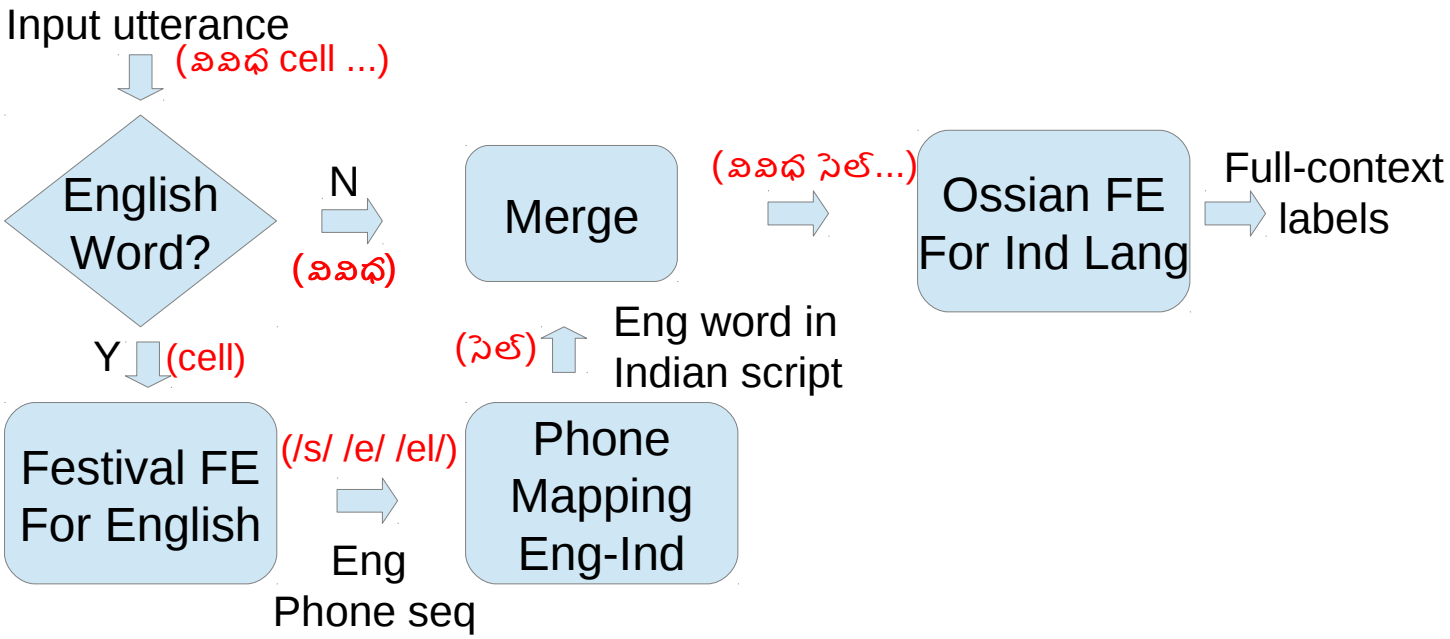
- Task: To sythesize dual-language utterances, primarily a native language (Indian) intersperced with words from a non-native language (English)
  - Training data
    - Single speaker data only in Indian language (a few hundred utterances)
      - Example: "प्रसिद्ध कबीर अध्येता, पुरुषोत्तम अग्रवाल का यह शोध आलेख, उस रामानंद की खोज करता है “
    - Audio data (16kHz, 16 bits) along with text in Indian script (UTF-8)
  - Test data
    - Example: “Under 19 cricket world cup में सोमवार को अफ़गानिस्तान ने ऑस्ट्रेलिया को हराकर, बड़ा उलटफेर किया है”

Dual-language synthesizer



Label generation - Method 1: Eng-to-Ind transcription

- Transcribe all English words in the target Indian langauge script and use the OSSIAN front-end (FE)



Input utterance

(వివిధ cell ...)

English Word?

N

(వివిధ)

Y

(cell)

Festival FE For English

(/s/ /e/ /el/)

Eng Phone seq

Phone Mapping Eng-Ind

(సెల్)

Eng word in Indian script

Merge

(వివిధ సెల్...)

Ossian FE For Ind Lang

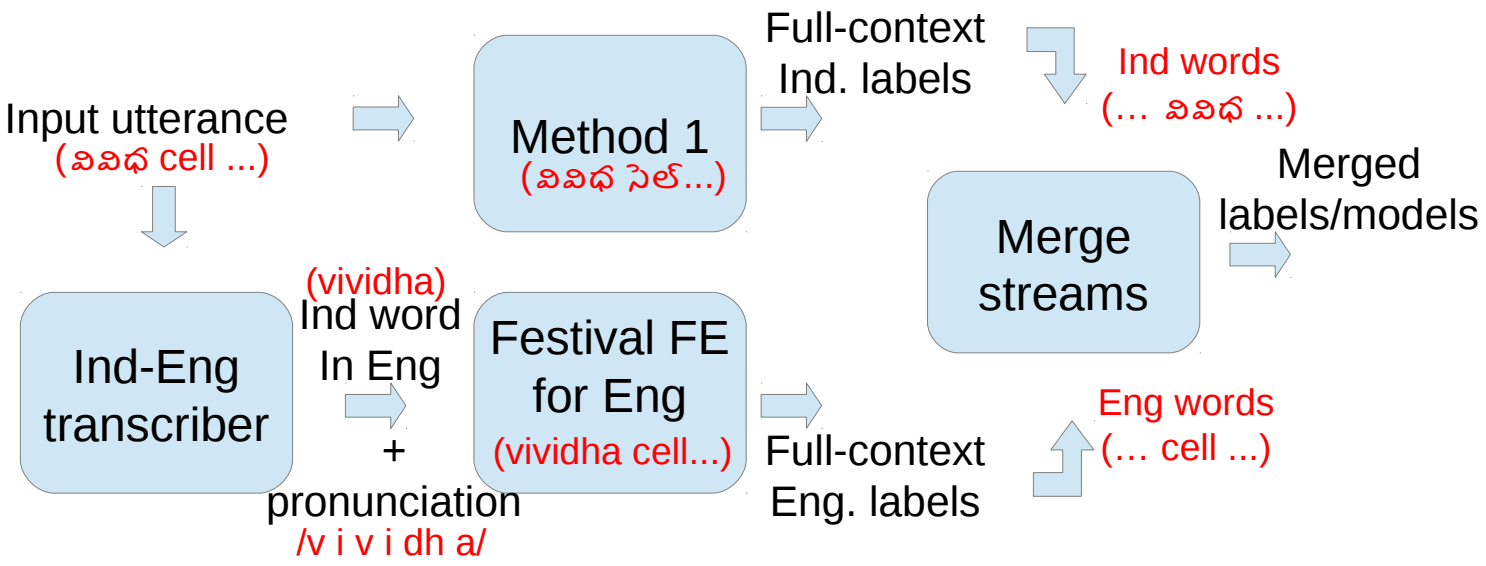
Full-context labels

Issues with Method 1: Eng-to-Ind transcription

- Works best for Telugu
  - Telugu script most phonetic
- Worst for Tamil
  - Context dependent phonemes in Tamil
  - Eg: single phoneme [k] represents both phones /k/ and /g/
- Not so good for Hin, Guj, Raj, and Asm.
  - Schwa insertion an isuue in Hin, Guj, and Raj.
  - Not able to assess Assamese

Label generation - Method 2: Dual front-end with filler words

- Use independent front-ends for English and the Ind. Lang., with filler words and merge the labels

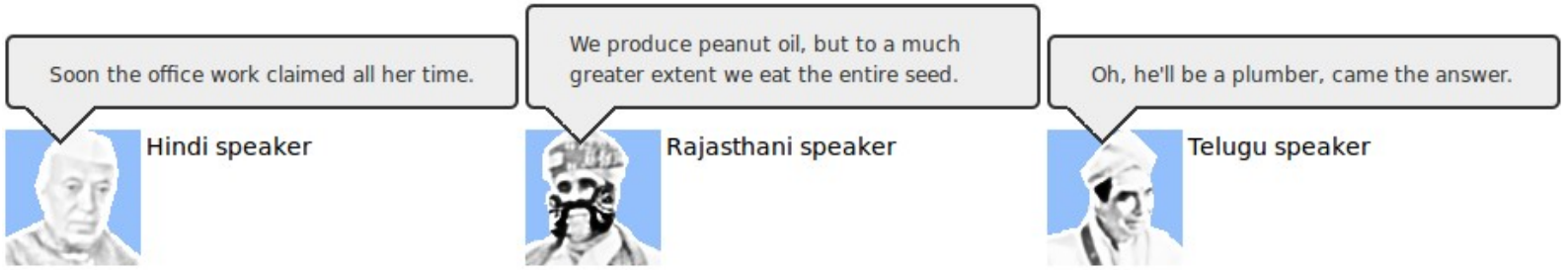


Acoustic model training 1: Indian language speaker dependent HTS

- Speaker dependent models for 3 different Indian languages – Hindi, Rajasthani and Telugu
- Training data (audio + UTF-8 text)
  - No. of uterrances - Hindi: 875, Rajasthani: 1369, Telugu: 1470
- GlottHMM acoustic modeling with Ossian front-end
- Features – LSFs:30, LSFsource:10, HNR:5, Gain:1, F0:1
- Trained with modified UEDIN Blizzard 2010 scripts.
  - Number of reclusterings = 3

Acoustic model training 2: Cross-lingual adaptation 0/4

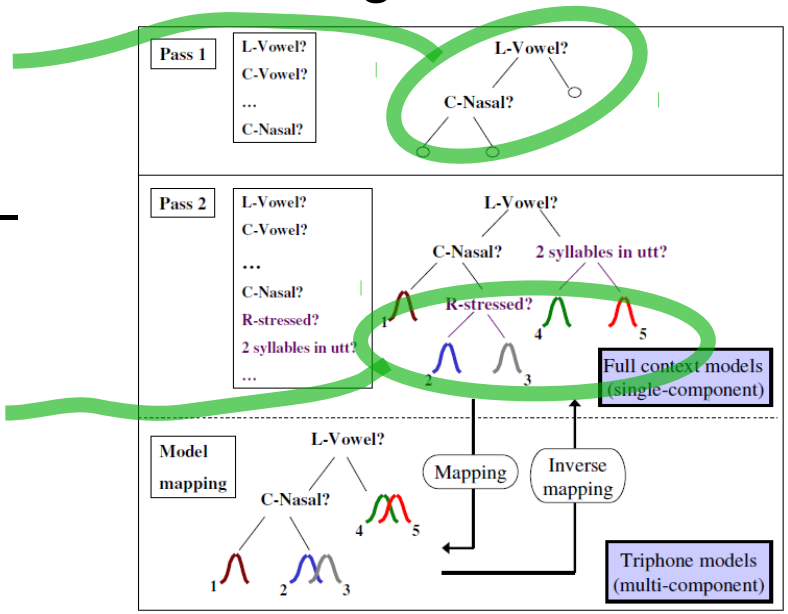
- We want to create an English model set for each of the Indian language speakers:



- We'll use the Arctic KSP speaker's Indian accented English data as a starting point

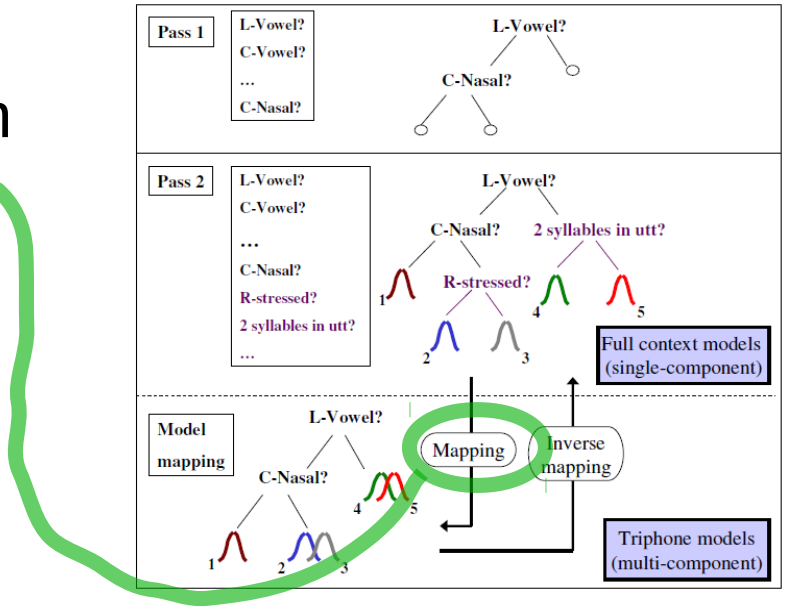
Acoustic model training 2: (Ind-Eng HTS) Cross-lingual adaptation 1/4

- English speaker-dependent model trained with Matt Gibson's code for two-pass decision tree generation:
- Top part of tree is populated by ASR-style questions related to triphone contexts – No leaf nodes!
- Bottom part of tree contains TTS style questions about quinphone context, stress, position in phrase etc... and leaf nodes with Gaussians



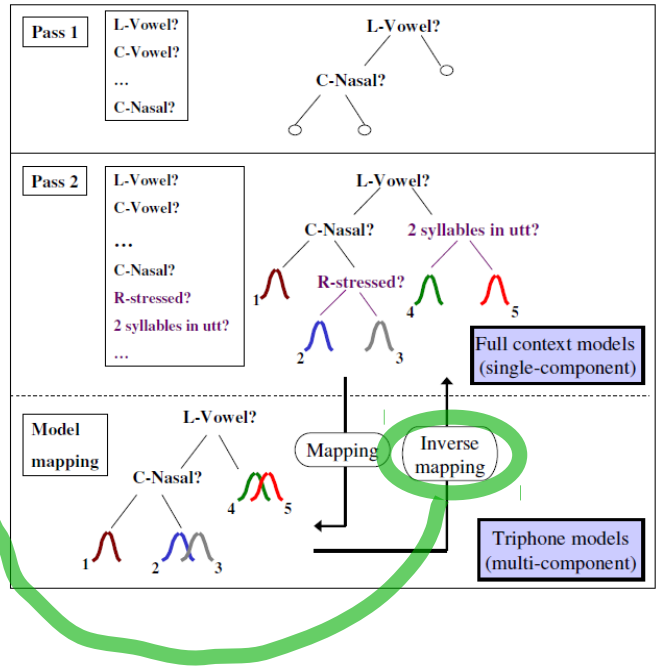
Acoustic model training 2: Cross-lingual adaptation 2/4

- After training, the Gaussians of the TTS model are collected based on the triphone contexts to make an ASR style model set
- This model set can be used to decode audio



Acoustic model training 2: Cross-lingual adaptation 3/4

- Indian language speakers' utterances are decoded using the triphone set
- A simple phoneme loop is used as the language model
- Triphone labels are then aligned using the different associated full-context TTS-models as alternative pronunciations for each triphone.
- This gives a mapping between Indian language audio and English TTS-models and allows adaptation to be carried out normally.



Acoustic model training 2: Cross-lingual adaptation 4/4

- Potential trouble
  - Lack of data for an average voice
  - We're adapting a speaker-dependent voice
- But at least we have a fair amount of training data for adaptation
- Could complete only 3 out of 6 Indian languages
  - Hindi, Rajasthani and Telugu completed. Stay tuned for samples!
  - Tamil: Lack of an Indian-English female voice
  - Gujarathi: Bad cross-lingual adaptation
  - Assamese: Bad speaker dep. voice for Indian lang.

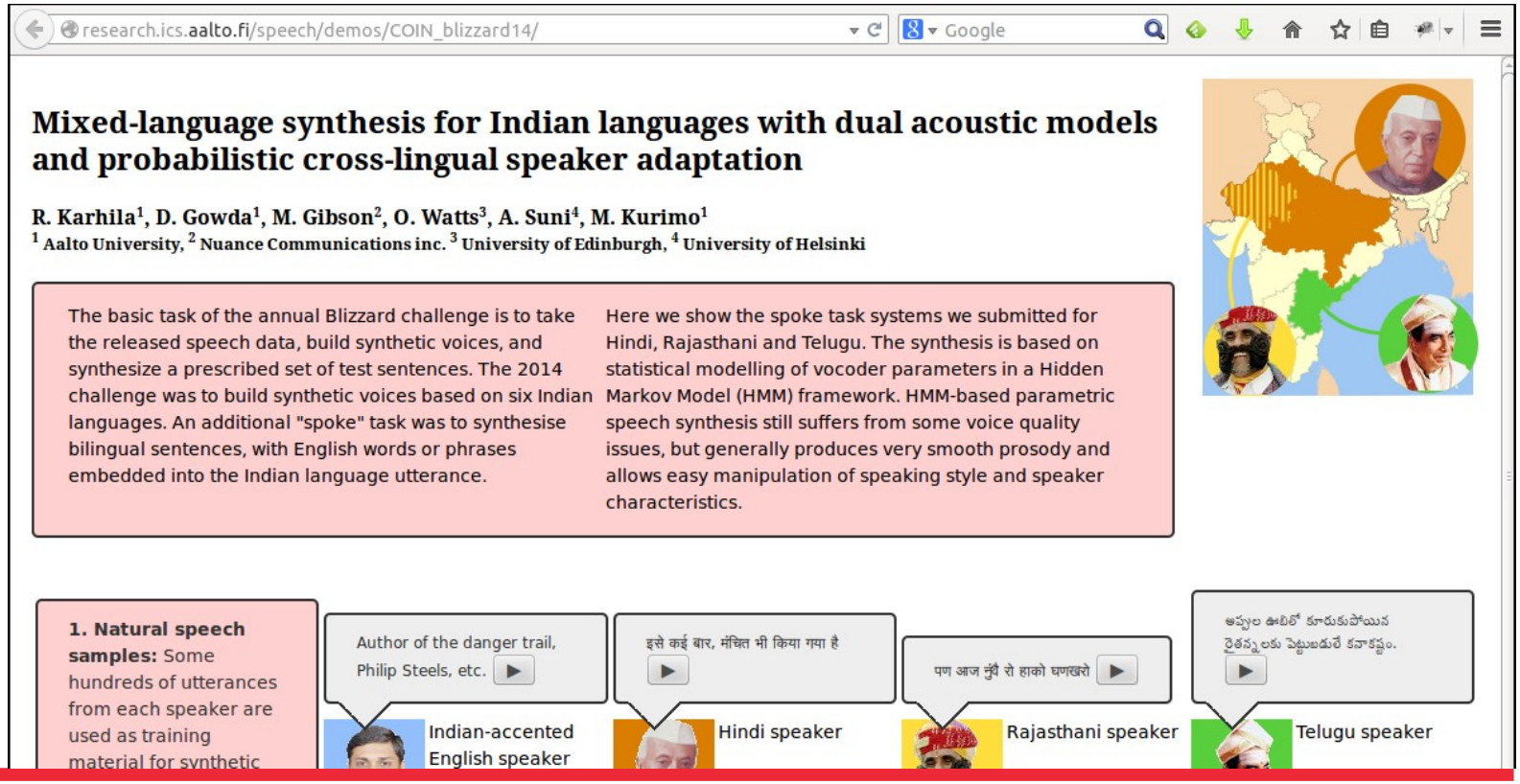
Merging model sets

- Unseen models need to be synthesised using the decision trees, independently for both languages.
- The required models from both language model sets are concatenated:
  - Macros are renamed to retain uniqueness
  - Duplicate headers are removed
- Otherwise synthesis is done in a normal way

```
~o
<STREAMINFO> 6 93 30 15 1 1 1
<MSDINFO> 6 0 0 0 1 1 1
<VECSIZE> 141<NULLD><USER><DIAGC>
~t "TrP_10"
<TRANSP> 7
...
~p "hnr_s4_1440"
<STREAM> 3
<MEAN> 15
...
~h "-
s+/0:v/1:ee/2:s/3:u/4:k/5:15/6:11/7:35/8:25/9:23/10:61/11:56
/12:65/13:4/14:66/15:64/16:57/17:64/18:8/19:5/20:1/21:3/22:2
/23:2/24:2/25:2/26:5/27:0/28:0/29:15/30:0/31:10/32:0/33:11/3
4:3/35:15/36:7/37:10/38:0/39:0/40:1/41:0/42:5/43:0/44:0/45:3
6/46:0/47:26/48:0/49:4/50:0/51:29/52:4/53:0/54:8/55:0/56:0/5
7:38/58:5/59:1/"
<BEGINHMM>
<NUMSTATES> 7
...
<ENDHMM>
~t "enTrP_10"
<TRANSP> 7
...
~p "en(flow_s2_301)169"
<STREAM> 2
<MEAN> 30
...
~h "e~n-t^+@=n:4_1/A/0_0_2/B/1-1-4:3-2&4-8#1-2$2-2>0-2<2-7|
e/C/0+0+3/D/content_1/E/content+4:2+6&2+5#1+1/F/content_2/G/
0_0/H/11=7:1=1&L-L%/I/0_0/J/11+7-1"
<BEGINHMM>
...
```

Demo page

http://research.ics.aalto.fi/speech/demos/COIN_blizzard14/



Issues to Ponder Over?

- How can a simple thing like this work?
- Role of filler words in label generation
  - for smooth transition
  - word count, accurate phone/context at boundaries
  - what else?
- How significant is syllable and stress related information for Indian languages?

References

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