Convolutional Neural Networks in Speech



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Lecture 20



- Fully connected (dense) layers have no awareness of spatial information
- Key concept behind convolutional layers is that of kernels or filters
- Filters slide across an input space to detect spatial patterns (translation invariance) in local regions (locality)

Fully Connected Layers

32x32x3 image -> stretch to 3072 x 1

input



3072





32x32x3 image



5x5x3 filter



Image from:http://cs231n.stanford.edu/slides/2019/cs231n_2019_lecture05.pdf





32x32x3 image

convolve (slide) over all spatial locations







convolve (slide) over all spatial locations

activation maps









Convolutional Neural Network





What do these layers learn?

















Convolution Layers: Summary

- Accepts a volume of size $W_1 \times H_1 \times D_1$
- Requires four hyperparameters:
 - \circ Number of filters K_{i}
 - \circ their spatial extent F,
 - \circ the stride S,
 - \circ the amount of zero padding P.
- Produces a volume of size $W_2 \times H_2 \times D_2$ where:
 - $W_2 = (W_1 F + 2P)/S + 1$
 - $\circ D_2 = K$
- and *K* biases.

 $\circ H_2 = (H_1 - F + 2P)/S + 1$ (i.e. width and height are computed equally by symmetry)

• With parameter sharing, it introduces $F \cdot F \cdot D_1$ weights per filter, for a total of $(F \cdot F \cdot D_1) \cdot K$ weights

Summary from: http://cs231n.github.io/convolutional-networks/

Pooling Layer

Pooling Layer

- Accepts a volume of size $W_1 \times H_1 \times D_1$ Requires two hyperparameters: \circ their spatial extent F,
- - \circ the stride S,
- - $W_2 = (W_1 F)/S + 1$ $H_2 = (H_1 - F)/S + 1$ $D_{2} = D_{1}$

• Produces a volume of size $W_2 \times H_2 \times D_2$ where:

CNNs for Speech

Speech features to be fed to a CNN

$$Q_j = \sigma \left(\sum_{i=1}^{I} O_i * \mathbf{w}_i \right)$$
$$p_{i,m} = \max_{n=1}^{G}$$

Illustrating a CNN layer

Convolution operations involve a large sparse matrix

feature maps in each band

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Shared weight matrices (W)

CNN Architecture used in a hybrid ASR system

Performance on TIMIT of different CNN architectures (Comparison with DNNs)

ID	Network structure	Average PER	min-max PER	# param's	# 0
1	DNN $\{2000 + 2 \times 1000\}$	22.02%	21.86-22.11%	6.9M	6.
2	DNN $\{2000 + 4 \times 1000\}$	21.87%	21.68-21.98%	8.9M	8.
3	CNN {LWS(m:150 p:6 s:2 f:8) + 2×1000 }	20.17%	19.92-20.41%	5.4M	10
4	CNN {FWS(m:360 p:6 s:2 f:8) + 2×1000 }	20.31%	20.16-20.58%	8.5M	13

More recent ASR system: Deep Speech 2

Image from Amodei et al., "Deep speech 2: End-to-end speech recognition in English and Mandarin", ICML 2016

Chann	els	Filter dimension	Stride	Reg	gular Dev	Nois
1280		11	2		9.52	19
640, 640 5, 5		5, 5	1, 2		9.67	19
512, 512, 512		5, 5, 5	1, 1, 2		9.20	20
32		41x11	2x2		8.94	1
32, 32		41x11, 21x11	2x2, 2x1		9.06	1.
32, 32	, 96	41x11, 21x11, 21x11	2x2, 2x1, 2x	2x1	8.61	14
	Test set		Ours	Human		
	WSJ ev	al'92	3.10	5.03		
ad	WSJ ev	al'93	4.42	8.08		
Re	LibriSp	eech test-clean	5.15	5.83		
	LibriSp	eech test-other	12.73	12.69		
pa	VoxFor	ge American-Canadian	7.94	4.85	_	
nte	VoxFor	ge Commonwealth	14.85	8.15		
cce	VoxFor	ge European	18.44	12.76		
A	VoxFor	ge Indian	22.89	22.15		
Sy	CHiME	z eval real	21.59	11.84	_	
Noi	CHiME	eval sim	42.55	31.33		

TTS: Wavenet

- Speech synthesis using an auto-regressive generative model Generates waveform sample-by-sample:16kHz sampling rate
- ullet•

1 Second

Gif from https://deepmind.com/blog/wavenet-generative-model-raw-audio/

Causal Convolutions

- Fully convolutional •
- •

Prediction at timestep t cannot depend on any future timesteps

Dilated Convolutions

- Wavenet uses "dilated convolutions" •
- Enables the network to have very large receptive fields

Gif from https://deepmind.com/blog/wavenet-generative-model-raw-audio/ ¹https://techcrunch.com/2017/10/04/googles-wavenet-machine-learning-based-speech-synthesis-comes-to-assistant/

Conditional Wavenet

- Condition the model on input variables to generate audio with the required characteristics
- Global (same representation used to influence all timesteps) • Local (use a second timeseries for conditioning)

Output	•	•	•	•	•
Hidden Layer	0	0	0	0	(
Hidden Layer	0	0	0	0	(
Hidden Layer	0	0	0	0	(
Input	0	0	0	0	

Image from Wang et al., "Tacotron: Towards end-to-end speech synthesis", 2017. "https://arxiv.org/pdf/1703.10135.pdf

Tacotron

Image from Wang et al., "Tacotron: Towards end-to-end speech synthesis", 2017. "https://arxiv.org/pdf/1703.10135.pdf

Tacotron: CBHG Module

Grapheme to phoneme (G2P) conversion

Grapheme to phoneme (G2P) conversion

- Produce a pronunciation (phoneme sequence) given a written word (grapheme sequence)
- Learn G2P mappings from a pronunciation dictionary
- Useful for:
 - ASR systems in languages with no pre-built lexicons
 - Speech synthesis systems
 - Deriving pronunciations for out-of-vocabulary (OOV) words

G2P conversion (I)

- One popular paradigm: Joint sequence models [BN12]
 - Grapheme and phoneme sequences are first aligned • using EM-based algorithm
 - Results in a sequence of graphones (joint G-P tokens)
 - Ngram models trained on these graphone sequences
- WFST-based implementation of such a joint graphone model [Phonetisaurus]

[BN12]:Bisani & Ney, "Joint sequence models for grapheme-to-phoneme conversion", Specom 2012 [Phonetisaurus] J. Novak, Phonetisaurus Toolkit

G2P conversion (II)

- for G2P
 - layer [Rao15]. Comparable to Ngram models.

 - above systems [Toshniwal16].

Neural network based methods are the new state-of-the-art

Bidirectional LSTM-based networks using a CTC output

Incorporate alignment information [Yao15]. Beats Ngram

No alignment. Encoder-decoder with attention. Beats the

LSTM + CTC for G2P conversion [Rao15]

Model	Word Error Rate (%)
Galescu and Allen [4]	28.5
Chen [7]	24.7
Bisani and Ney [2]	24.5
Novak et al. [6]	24.4
Wu et al. [12]	23.4
5-gram FST	27.2
8-gram FST	26.5
directional LSTM with Full-delay	30.1
DBLSTM-CTC 128 Units	27.9
DBLSTM-CTC 512 Units	25.8
BLSTM-CTC 512 + 5-gram FST	21.3

[Rao15] Grapheme-to-phoneme conversion using LSTM RNNs, ICASSP 2015

G2P conversion (II)

- Neural network based methods are the new state-of-the-art for G2P
 - Bidirectional LSTM-based networks using a CTC output layer [Rao15]. Comparable to Ngram models.
 - Incorporate alignment information [Yao15]. Beats Ngram models.
 - No alignment. Encoder-decoder with attention. Beats the above systems [Toshniwal16].

Seq2seq models (with alignment information [Yao15])

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Method	PER (%)	WER (%)
encoder-decoder LSTM	7.53	29.21
encoder-decoder LSTM (2 layers)	7.63	28.61
uni-directional LSTM	8.22	32.64
uni-directional LSTM (window size 6)	6.58	28.56
bi-directional LSTM	5.98	25.72
bi-directional LSTM (2 layers)	5.84	25.02
bi-directional LSTM (3 layers)	5.45	23.55

Data	Method	PER (%)	WER (%)
CMUDict	past results [20]	5.88	24.53
	bi-directional LSTM	5.45	23.55
NetTalk	past results [20]	8.26	33.67
	bi-directional LSTM	7.38	30.77
Pronlex	past results [20, 21]	6.78	27.33
	bi-directional LSTM	6.51	26.69

G2P conversion (II)

- for G2P
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 - above systems [Toshniwal16].

[Rao15] Grapheme-to-phoneme conversion using LSTM RNNs, ICASSP 2015 [Yao15] Sequence-to-sequence neural net models for G2P conversion, Interspeech 2015

[Toshniwal16] Jointly learning to align and convert graphemes to phonemes with neural attention models, SLT 2016.

Neural network based methods are the new state-of-the-art

Incorporate alignment information [Yao15]. Beats Ngram

No alignment. Encoder-decoder with attention. Beats the

Encoder-decoder + attention for G2P [Toshniwal16]

Encoder-decoder + attention for G2P [Toshniwal16]

Method	PER (%)
BiDir LSTM + Alignment [6]	5.45
DBLSTM-CTC [5]	-
DBLSTM-CTC + 5-gram model [5]	-
Encoder-decoder + global attn	5.04 ± 0.03
Encoder-decoder + local- m attn	5.11 ± 0.03
Encoder-decoder + local- p attn	5.39 ± 0.04
Ensemble of 5 [Encoder-decoder + global attn] models	4.69
BiDir LSTM + Alignment [6]	6.51
Encoder-decoder + global attn	6.24 ± 0.1
Encoder-decoder + local- m attn	5.99 ± 0.11
Encoder-decoder + local- p attn	6.49 ± 0.06
BiDir LSTM + Alignment [6]	7.38
Encoder-decoder + global attn	7.14 ± 0.72
Encoder-decoder + local- m attn	7.13 ± 0.11
Encoder-decoder + local- p attn	8.41 ± 0.19