VID2SPEECH: SPEECH RECONSTRUCTION FROM SILENT VIDEO

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ABSTRACT

Speechreading is a notoriously difficult task for humans to perform. In this paper we present an end-to-end model based on a convolutional neural network (CNN) for generating an intelligible acoustic speech signal from silent video frames of a speaking person. The proposed CNN generates sound features for each frame based on its neighboring frames. Waveforms are then synthesized from the learned speech features to produce intelligible speech. We show that by leveraging the automatic feature learning capabilities of a CNN, we can obtain state-of-the-art word intelligibility on the GRID dataset, and show promising results for learning out-of-vocabulary (OOV) words.

Index Terms— Speechreading, visual speech processing, articulatory-to-acoustic mapping, speech intelligibility, neural networks

1. INTRODUCTION

Speechreading is the task of obtaining reliable phonetic information from a speaker’s face during speech perception. It has been described as “trying to grasp with one sense information meant for another”. Given the fact that often several phonemes (phonetic units of speech) correspond to a single viseme (visual unit of speech), it is a notoriously difficult task for humans to perform.

Several applications come to mind for automatic video-to-speech systems: Enabling videoconferencing from within a noisy environment; facilitating conversation at a party with loud music between people having wearable cameras and earpieces; maybe even using surveillance video as a long-range listening device.

Much work has been done in the area of automating speechreading by computers [1 2 3]. There are two main approaches to this task. The first, and the one most widely attempted in the past, consists of modeling speechreading as a classification problem. In this approach, the input video is manually segmented into short clips which contain either whole words from a predefined dictionary, or parts of words comprising phonemes or visemes [4]. Then, visual features are extracted from the frames and fed to a classifier. Wand et al. [5] recently showed state-of-the-art word-classification results by using CNN-features fed to an LSTM model for classification.

The second approach, and the one used in this work, is to model speechreading as an articulatory-to-acoustic mapping problem in which the “label” of each short video segment is a corresponding feature vector representing the audio signal. Kello and Plaut [6] and Hueber and Bailly [7] attempted this approach using various sensors to record mouth movements. Le Cornu and Milner [8] took this direction in a recent work where they used hand-crafted visual features to produce intelligible audio.

A major advantage of this model of learning is its non-dependency on a particular segmentation of the input data into words or sub-words. It does not either need to have explicit manually-annotated labels, but rather uses “natural supervision” [9], in which the prediction target is derived from a natural signal in the world. A regression-based model is also vocabulary-agnostic. Given a training set with a large enough representation of the phonemes/visemes of a particu-
speech production [10]. LPC analysis is applied to overlapping audio frames of the original speech signal, resulting in an LPC coefficient vector whose order \( P \) can be tuned. Linear Spectrum Pairs (LSP) [11] are a representation of LPC coefficients which are more stable and robust to quantization and small coefficient deviations. LSPs are therefore useful for speech coding and transmission over a channel, and indeed proved to be well suited to the task at hand.

We apply the following procedure to calculate audio features suitable for use as neural network output: First, the audio from each video sequence is downsampled to 8kHz and split into audio frames of 40ms (320 samples) each, with an overlap of 20ms. 8th-order LPC analysis is applied to each audio frame, as done by [8], followed by LSP decomposition, resulting in a feature vector of length 9 per frame. While 8th-order LPC is relatively low for high-fidelity modeling of the speech spectrum, we did so in order to isolate the effect of using CNN-learned visual features versus the hand-crafted ones of [8]. Each video frame has two successive corresponding feature vectors, which are concatenated to form a sound vector, \( S_i \in \mathbb{R}^{18} \). See Figure 1 for an illustration of this procedure. Finally, the vectors are standardized element-wise by subtracting the mean and dividing by the standard deviation of each element.

3. PREDICTING SPEECH

3.1. Regressing sound features

Given a sequence of input frames \( I_1, I_2, \ldots, I_N \) we would like to estimate a corresponding sequence of sound features \( S_1, S_2, \ldots, S_N \) where \( S_i \in \mathbb{R}^{18} \).

Input representation Our goal is to reconstruct a single audio representation vector \( S_i \) which corresponds to the duration of a single video frame \( I_i \). However, instantaneous lip movements such as those in isolated video frames can be significantly disambiguated by using a temporal neighborhood as context. Therefore, the input to our network is a clip of \( K \) consecutive grayscale video frames, out of which the speaker’s face is cropped and scaled to 128 \( \times \) 128 pixels. This results in an input volume of size 128 \( \times \) 128 \( \times \) \( K \) scalars, which is then normalized by dividing by maximum pixel intensity and subtracting the mean.

Figure 2 illustrates the importance of allowing the network to learn visual features from the speaker’s entire face, as opposed to the mouth region only, as widely done in the past. The two lines in the graph represent final network test error as a function of the length \( K \) of the clip used as input to the CNN. We tested the values of \( K \in \{1, 3, 5, 7, 9\} \), while the output \( S_i \) always remained the sound features of the center frame. Not surprisingly, the largest gain in performance for both face and mouth regions is when clip length is increased from \( K = 1 \) frame to \( K = 3 \) frames, highlighting the importance of con-
Fig. 3. Examples of original (top) and reconstructed (bottom): (a) LSP coefficients, (b) waveform and (c) spectrogram. The vertical columns of (a) are the actual output of the CNN. Spectral envelope of reconstructed audio (c) is relatively accurate, however unvoiced excitation results in the lack of formants (horizontal lines inside spectral envelope, representing frequency of voiced speech).

text. The advantage of learning features from the full facial information is also evident, with the best face region error 40% lower than the best mouth region error (both at K = 9). We hypothesize that this is as result of our CNN using the increased amount of visual information to disambiguate similar mouth movements.

Sound prediction model We use a convolutional neural network (CNN) that takes the aforementioned video clip of size 128 × 128 × K as input. Our network uses VGG-like [12] stacks of small 3 × 3 receptive fields in its convolutional layers. The architecture comprises five consecutive conv3 − conv3 − maxpool blocks consisting of 32 − 32 − 64 − 128 − 128 kernels, respectively. These are followed by two fully connected layers with 512 neurons each. The last layer of our CNN is of size 18 which corresponds to the size of the sound representation vectors we wish to predict. The network is trained with backpropagation using mean squared error (MSE) loss.

3.2. Generating a waveform

Source-filter speech synthesizers such as [13] use both filter parameters as well as an excitation signal to construct an acoustic signal from LPC features. Predicting excitation parameters is out of the scope of this work, and we therefore use Gaussian white noise as the excitation signal. This produces an unvoiced speech signal and results in unnatural sounding speech. Although this method of generating a waveform is relatively simplistic, we found that it worked quite well for speech intelligibility purposes, which is the focus of our work.

4. EXPERIMENTS

We applied our speech-reconstruction model to several tasks, and evaluated it with a human listening study[1]

| Command | Color | Preposition | Letter | Digit | Adverb |
|---------|-------|-------------|--------|-------|--------|
| bin     | blue  | at          | A-Z    | 0-9   | again  |
| lay     | green | by          | minus W|       | now    |
| place   | red   | in          |        |       | please |
| set     | white | with        |        |       | soon   |

Table 1. GRID sentence grammar.

Implementation details Our network implementation is based on the Keras library [14] built on top of TensorFlow [15]. Network weights are initialized using the initialization procedure suggested by He et al. [16]. We use Leaky ReLU [17] as the non-linear activation function in all layers but the last two, in which we use the hyperbolic tangent (tanh) function. Adam optimizer [18] is used with a learning rate of 0.003. Dropout [19] is used to prevent overfitting, with a rate of 0.25 after convolutional layers and 0.5 after fully connected ones. We use mini-batches of 32 training samples each and stop training when the validation loss stops decreasing (around 80 epochs). Training is done using a single Nvidia Titan Black GPU. We use a cascade-based face detector from OpenCV [20], and crop out the mouth region for the comparison in Figure 2 by using a hard-coded mask. For LPC analysis/resynthesis, as well as excitation generation, we used pysptk, a Python wrapper for Speech Signal Processing Toolkit (SPTK) [21].

4.1. GRID corpus

We performed our experiments on the GRID audiovisual sentence corpus [22], a large dataset of audio and video (facial) recordings of 1000 sentences spoken by 34 talkers (18 male, 16 female). Each sentence consists of a six word sequence of the form shown in Table[1] e.g. “Place green at H 7 now”.

A total of 51 different words are contained in the GRID corpus. Videos have a fixed duration of 3 seconds at a frame
rate of 25 FPS with 720×576 resolution, resulting in sequences comprising 75 frames. These videos are preprocessed as described in Section 2.1 before feeding them into the network. The acoustic part of the GRID corpus is used as described in Section 2.

In order to accurately compare our results with [8], we performed our experiments on the 1000 videos of speaker four (S4, female) as done there. The training/testing split for each experiment will be described in the following sections.

4.2. Sound prediction tasks

Reconstruction from full dataset. The first task, proposed by [8], is designed to examine whether reconstructing audio from visual features can produce intelligible speech. For this task we trained our model on a random 80/20 train/test split of the 1000 videos of S4 and made sure that all 51 GRID words were represented in each set. The resulting representation vectors were converted back into waveform using unvoiced excitation, and two different multimedia configurations were constructed: the predicted audio-only and the combination of the original video with reconstructed audio.

Reconstructing out-of-vocabulary words. As cited earlier, regression-based models can be used to reconstruct out-of-vocabulary (OOV) words. To test this, we performed the following experiment: The videos in our dataset were sorted according to the digit uttered in each sentence, and our network was trained and tested on five different train/test splits - each with two distinct digits left out of the training set. For example, the network was trained on all sequences with the numbers 1 – 8 uttered, and tested only on sequences containing the numbers 9 and 0.

4.3. Evaluating the speech predictions

We assessed the intelligibility of the reconstructed speech using a human listening study done using Amazon Mechanical Turk (MTurk). Each job consisted of transcribing one of three types of 3-second clips: audio-only, audio-visual and OOV audio-visual. The listeners were unaware of the differences between the clips. For each clip, they were given the GRID vocabulary and tasked with classifying each reconstructed word into one of its possible options. All together, over 400 videos containing 38 distinct sequences were transcribed by 23 different MTurk workers, which is comparable to the 20-listener study done by [8].

4.4. Results

Table 2 shows the results of our first task, reconstruction from the full dataset, along with a comparison to [8]. Our reconstructed audio is significantly more intelligible than the best results of [8], as shown by both audio-only and audio-visual tests. The final column shows the result of retraining and testing our model on another speaker from the GRID corpus, speaker two (S2, male), whose speech clarity is comparable to S4, as reported by [22]. We used the same listening test methodology described above, however this time only using combined audio and video. Examples of original vs. reconstructed LSP coefficients, waveform and spectrogram for this task can be seen in Figure 3.

Table 3. Out-of-vocabulary (OOV) intelligibility results. We tested this by reconstructing spoken digits which were left out of the training set. Listeners were five times more likely to choose the correct digit than randomly guessing, however only slightly more than half as likely compared to having all digits represented in the training set.

5. CONCLUDING REMARKS

This work has proven the feasibility of reconstructing an intelligible audio speech signal from silent videos frames. OOV word reconstruction was also shown to hold promise by modeling automatic speechreading as a regression problem, and using a CNN to automatically learn relevant visual features.

The work described in this paper can serve as a basis for several directions of further research. These include using a less constrained video dataset to show real-world reconstruction viability and generalizing to speaker-independent and multiple speaker reconstruction.
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