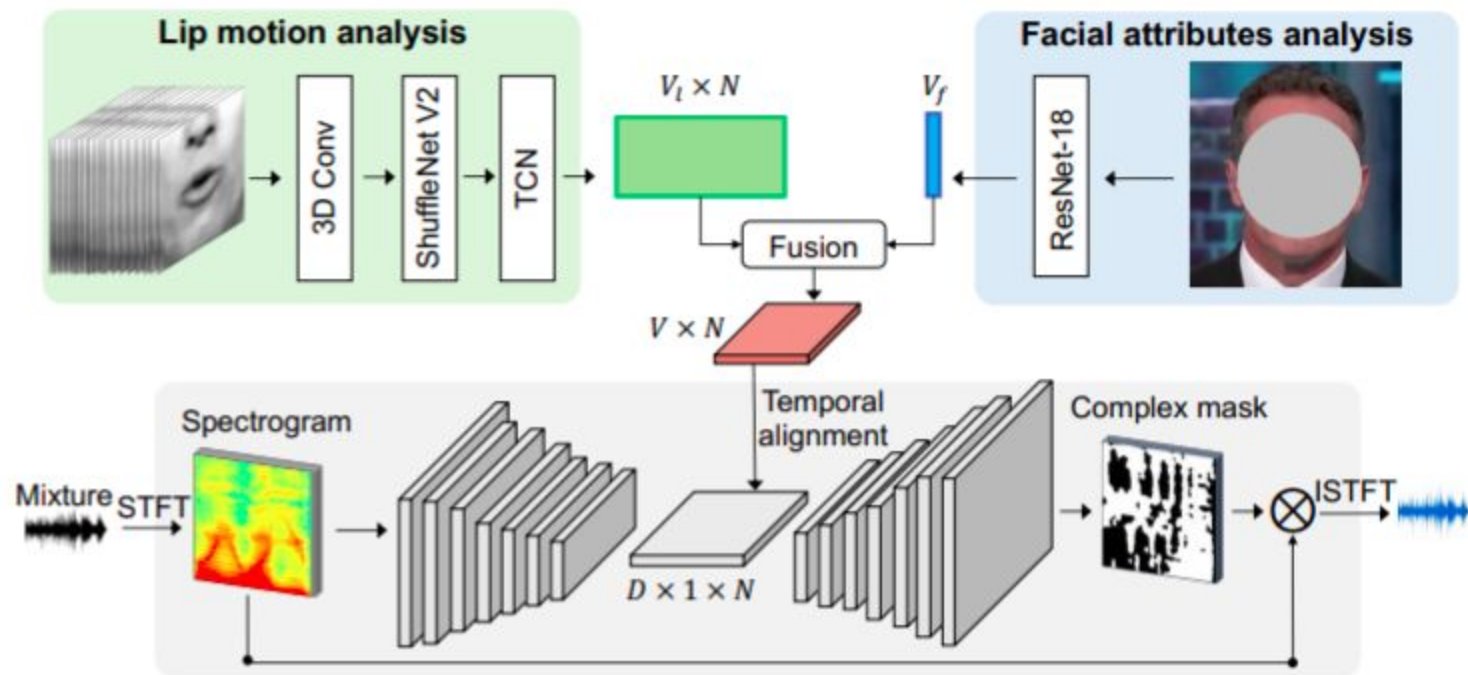


# VISUALVOICE: Audio-Visual Speech Separation with Cross-Modal Consistency

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**Objective :** A multi-task learning framework to jointly learn audio-visual speech separation.



# Synthesising Training data

Two speech segments  $s_{\mathcal{A}_1}(t)$ ,  $s_{\mathcal{A}_2}(t)$  from video  $V_A$  for speaker A, and  $s_{\mathcal{B}}(t)$  from video  $V_B$  for speaker B. (Speaker consistency loss is used)

Let  $F_{\mathcal{A}_1}, F_{\mathcal{A}_2}, F_{\mathcal{B}}$  denote the face tracks associated with the speech segments  $s_{\mathcal{A}_1}(t)$ ,  $s_{\mathcal{A}_2}(t)$ ,  $s_{\mathcal{B}}(t)$  respectively.

We create two mixture signals  $x_1(t)$  and  $x_2(t)$ :

$$x_1(t) = s_{\mathcal{A}_1}(t) + s_{\mathcal{B}}(t), \quad x_2(t) = s_{\mathcal{A}_2}(t) + s_{\mathcal{B}}(t).$$

The mixture speech signals are then transformed into complex audio spectrograms  $X_1$  and  $X_2$ .

# Mixing two audio samples

1. **Get audio timeseries** (1D array representing amplitudes) using `librosa.core.load()`
2. **Average amplitudes** both audio samples , it represents audio time series of mixture
3. Get **complex frequency-time spectrogram** of mixed audio using `librosa.load.stft()`
4. This spectrogram is fed into audio-visual speech separation network

# Visual Features Extraction

Face tracks are given a input to lip motion and facial attributes analysis network.

The lip motion analysis network takes as input the lip motion features  $F_{A_1}, F_{A_2}, F_B$  and outputs the lip motion features  $F_{L_1}, F_{L_2}, F_{L_3}$ . The facial attributes analysis network takes as input the facial attributes features  $F_{A_1}, F_{A_2}, F_B$  and outputs the facial attributes features  $F_{F_1}, F_{F_2}, F_{F_3}$ . The lip motion features  $F_{L_1}, F_{L_2}, F_{L_3}$  and the facial attributes features  $F_{F_1}, F_{F_2}, F_{F_3}$  are fused to produce the final visual features  $F_{V_1}, F_{V_2}, F_{V_3}$ .

Replicate the feature map (and concatenate with the feature map)  $n$  times along the time dimension. We then have a 2D feature map of dimension  $64 \times 64 \times n$ . So for each of the face track, we have a 2D visual feature map

$$F_{A_1}, F_{A_2}, F_B$$

# Audio Feature Extraction

Encoder and a decoder network.

Frequency consists of convolutional and frequency pooling layer to downsample the input. The output of the encoder is the complex spectrogram of the mixture signal of dimension  $2 \times F \times T$ , where  $F$  is the frequency and  $T$  is the time.

Encoder Output is a audio feature map of dimension  $D \times 1 \times N$ , where  $D$  is the number of channels and  $N$  is the number of time steps.

Concatenate the visual and audio features along the channel dimension to generate an audio-visual feature map of dimension  $(V+D) \times 1 \times N$ .



**Complex mask** of dimension  $2 \times F$  as input and **predicts** a  
So using mixture spectrogram  $X_1$  and  $F^{A1}$ , mask  $M_{A1}$  predicted for speaker A  
using  $X_1$  and  $F^B$ , mask  $M_{B1}$  predicted for speaker B  
using  $X_2$  and  $F^{A2}$ , mask  $M_{A2}$  predicted for speaker A  
using  $X_2$  and  $F^B$ , mask  $M_{B2}$  predicted for speaker B

Using **Tanh** layer to map the output feature map values to the range of  $[-1, 1]$ .  
And then using scaling by 5 so that output mask values are in range  $[-5, 5]$   
because the real and imaginary parts of the ground-truth complex mask typically  
lie between -5 and 5.

# Speech separation by Masking

The predicted **spectrograms for the separated speech** signals can be obtained by complex masking the mixture spectrograms:

$$S_{\mathcal{A}_i} = X_i * M_{\mathcal{A}_i}, \quad S_{\mathcal{B}_i} = X_i * M_{\mathcal{B}_i}, \quad i \in \{1, 2\},$$

Finally, using the inverse short-time Fourier transform (ISTFT) , we reconstruct the separated speech signals.

## Mask Prediction Loss -

$$L_{mask-prediction} = \sum_{i \in \{\mathcal{A}_1, \mathcal{A}_2, \mathcal{B}_1, \mathcal{B}_2\}} \|M_i - \mathcal{M}_i\|_2,$$

where  $\mathcal{M}_i$  denotes the ground truth complex masks, which are obtained by taking the common spectrogram of the clean speech and the corresponding complex spectrogram.

# Cross-modal matching loss-

The predicted spectrograms  $\mathbf{S}_{A1}, \mathbf{S}_{B1}, \mathbf{S}_{A2}, \mathbf{S}_{B2}$  for separated speech signals are fed into ResNet-18 to extract audio embeddings

This loss is to link voice and face.

$\mathbf{a}^{\mathcal{A}_1}, \mathbf{a}^{\mathcal{A}_2}, \mathbf{a}^{\mathcal{B}_1}, \mathbf{a}^{\mathcal{B}_2}$  denote the face image embeddings,  $\mathbf{i}^{\mathcal{A}_1}, \mathbf{i}^{\mathcal{A}_2}, \mathbf{i}^{\mathcal{B}_1}, \mathbf{i}^{\mathcal{B}_2}$  denote the spectrogram embeddings.

$$L_t(\mathbf{a}, \mathbf{i}^+, \mathbf{i}^-) = \max\{0, D(\mathbf{a}, \mathbf{i}^+) - D(\mathbf{a}, \mathbf{i}^-) + m\},$$

The distance between the embedding of the generated utterance and the speaker, by analogy, is also considered, and the loss is defined as follows:

The **cross-modal matching loss** is defined as follows:

$$\begin{aligned} L_{cross-modal} = & L_t(\mathbf{a}^{A_1}, \mathbf{i}^A, \mathbf{i}^B) + L_t(\mathbf{a}^{A_2}, \mathbf{i}^A, \mathbf{i}^B) \\ & + L_t(\mathbf{a}^{B_1}, \mathbf{i}^B, \mathbf{i}^A) + L_t(\mathbf{a}^{B_2}, \mathbf{i}^B, \mathbf{i}^A). \end{aligned}$$

# Speaker Consistency Loss-

The audio embeddings for the separated parts of speaker B for speaker A; a speaker consistency loss on the audio embeddings of the separated speech;

$$L_{consistency} = L_t(\mathbf{a}^{\mathcal{A}_1}, \mathbf{a}^{\mathcal{A}_2}, \mathbf{a}^{\mathcal{B}_1}) + L_t(\mathbf{a}^{\mathcal{A}_1}, \mathbf{a}^{\mathcal{A}_2}, \mathbf{a}^{\mathcal{B}_2}).$$