Project - Deep Learning ECE-GY 7123 Spring 2023
Context-Aware Story Generation

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NYU Tandon School Of Engineering - https://github.com/anand-vishwakarma-1/dl-project

Abstract

It is a difficult challenge in artificial intelligence to generate stories that take context into account. The current state of the art is based on rule-based systems or Markov models, neither of which are capable of producing conversations that seem natural or accurately representing the complexity of human relationships.Using a pipeline consisting of GPT-2 for text prompts, DCTTS for audio creation, Wav2Lip for lip syncing, and picture generation, we suggest a system for context-aware audio visual story development. The goal of this method is to provide an engaging visual and auditory story that faithfully records interactions. Video games, VR/AR, and interactive pedagogical resources are just a few examples of how this approach may be put to use to create fully immersive entertainment, learning, and communication environments.

Introduction

Storytelling is an essential aspect of human communication and has played a vital role in entertainment, education, and culture throughout history. With the advent of modern technologies, storytelling has evolved from a purely verbal and visual medium to one that incorporates audio, video, and interactive elements. One of the challenges in modern storytelling is creating context-aware stories that are sensitive to their environment and provide an immersive experience to the audience.

Context-aware story generation is a challenging and open problem in the field of artificial intelligence. The goal of context-aware story generation is to create stories that are responsive to their environment, capture the nuances of human interactions, and maintain coherence across various dimensions. This requires integrating multiple sources of information, such as dialogues, actions, emotions, and background context, to create a compelling narrative that engages the audience.

The difficulty of integrating ideas and maintaining coherence across various dimensions remains a significant obstacle in context-aware story generation. Most existing solutions rely on rule-based systems or Markov models that are limited in their ability to generate natural-sounding dialogues and capture the complexity of human interactions. Therefore, there is a need for a more sophisticated approach that can generate expressive audiovisual narratives that accurately capture on-the-spot conversations and are sensitive to their environment.

This project proposes to address this issue through the use of transformers and generative deep learning techniques. The proposed system will generate a series of coordinated actions and dialogues among a predefined cast of characters in response to cues. Audio will be generated through deep generative learning algorithms, which will be synchronized with the lip movements of the characters. The intended outcome of this project is to provide an expressive audiovisual narrative that accurately captures on-the-spot conversations and is sensitive to its environment.

The proposed solution is expected to have significant applications in the fields of entertainment, education, and communication. Context-aware stories can be used to create immersive experiences in video games, virtual reality, and augmented reality. They can also be used to create interactive educational materials that engage students and promote learning. Finally, they can be used in communication and marketing to create compelling narratives that resonate with the audience.

Literature Survey

Storytelling has been an essential part of human communication since ancient times. With the advancement of modern technology, storytelling has evolved to include audio, video, and interactive elements. Context-aware story generation is a challenging and open problem in artificial intelligence, aiming to create stories that are responsive to their surroundings and provide an immersive experience to the audience. Context-aware story generation requires integrating multiple sources of information, such as dialogues, actions, emotions, and background context, to create a compelling narrative that engages the audience. However, integrating ideas and maintaining coherence across various dimensions remains a significant obstacle. Most existing solutions rely on rule-based systems or Markov models that are limited in their ability to generate natural-sounding dialogues and capture the complexity of human interactions.

Deep learning techniques such as transformers, recurrent neural networks (RNNs), and convolutional neural networks (CNNs) have been proposed to address the challenges of context-aware story generation. Transformers have become the state-of-the-art model for language generation. They are capable of capturing long-range dependencies and context, which is essential for generating coherent and natural-sounding stories.

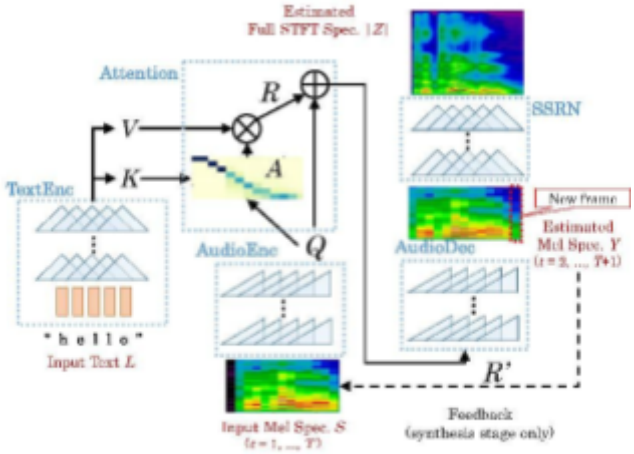
Deep Convolutional Text-to-Speech (DCTTS) is a deep generative learning algorithm that converts text into audio. DCTTS uses a combination of convolutional and recurrent neural networks to generate natural-sounding speech that captures the intonation, pitch, and rhythm of human speech. DCTTS has been shown to outperform traditional text-to-speech systems in terms of naturalness and expressiveness. Wav2Lip is a deep learning-based lip-syncing algorithm that uses audio and a video of a person's face to generate a video of the person lip-syncing to the audio. Wav2Lip has been shown to be effective in generating realistic lip movements that correspond to the audio. Lip-syncing image generation is a technique used to generate images of characters lip-syncing to the audio. This is achieved by combining the output of Wav2Lip with a predefined cast of characters and a background environment. Deep learning algorithms are used to generate images that accurately capture the lip movements and emotions of the characters in the story.

Context-aware stories have significant applications in the fields of entertainment, education, and communication. In entertainment, they can be used to create immersive experiences in video games, virtual reality, and augmented reality. In education, they can be used to create interactive materials that engage students and promote learning. In communication and marketing, they can be used to create compelling narratives that resonate with the audience. Context-aware story generation using deep learning techniques offers a sophisticated approach to generate expressive audiovisual narratives that accurately capture on-the-spot conversations and are sensitive to their environment. With further development and refinement, this approach could become a powerful tool for generating context-aware stories that engage and captivate audiences.

Architecture

The proposed system for context-aware story generation consists of a pipeline of four major components - GPT-2, DCTTS, Wav2Lip, and lip syncing image generation. The pipeline starts with a text prompt, which is fed into GPT-2, a powerful language model that generates text by predicting the next word given a context. The output of GPT-2 is a story that captures the nuances of human interactions, emotions, and background context. This generated story serves as the basis for the rest of the pipeline.

The output of GPT-2 is fed into DC-TTS, a deep generative learning algorithm that converts text into audio. DCTTS uses a combination of convolutional and recurrent neural networks to generate natural-sounding speech that captures the intonation, pitch, and rhythm of human speech. The output of DC-TTS is an audio file that corresponds to the story generated by GPT-2.

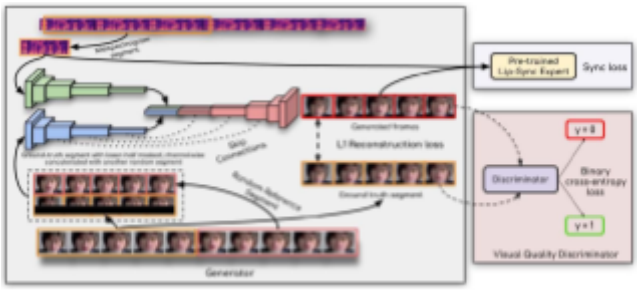


DC-TTS architecture

DCTTS is a modified deep convolutional network architecture, leveraging the strengths of guided attention mechanisms. The model was composed of several convolutional layers, with gated linear units (GLU) serving as activation functions. Additionally, the guided attention mechanism was integrated into the network to provide better alignment between input text sequences and output speech features. The model was trained on a large-scale, publicly available dataset containing hours of high-quality speech recordings.

The audio generated by DCTTS is then synchronized with the lip movements of the characters in the story using Wav2Lip. Wav2Lip is a deep learning-based lip-syncing algorithm that uses the audio file and a video of a person's face to generate a video of the person lip-syncing to the audio. In the proposed system, Wav2Lip is used to generate lip movements that correspond to the audio generated by DCTTS.

Mean opinion score (MOS), mel Cepstral distortion (MCD), and perceptual evaluation of speech quality (PESQ) are some of the objective measures used to assess the DC-TTS system's efficacy. Naturalness, fluency, and intelligibility of the synthesized speech are also evaluated subjectively by human participants in listening tests.



Wav2Lip Architecture

The final component of the proposed system is lip-syncing image generation, which generates images of the characters in the story lip-syncing to the audio. This is achieved by combining the output of Wav2Lip with a predefined cast of characters and a background environment. The lip-syncing image generation component uses deep learning algorithms to generate images that accurately capture the lip movements and emotions of the characters in the story.

The proposed system is designed to generate context-aware stories that are responsive to their environment and provide an immersive experience to the audience. By using deep generative learning algorithms and lip-syncing image generation, the system can generate expressive audiovisual narratives that accurately capture on-the-spot conversations and are sensitive to their environment.

The proposed system has significant applications in the fields of entertainment, education, and communication. For example, context-aware stories can be used to create immersive experiences in video games, virtual reality, and augmented reality. In education, they can be used to create interactive materials that engage students and promote learning. In communication and marketing, they can be used to create compelling narratives that resonate with the audience.

In conclusion, the proposed system for context-aware story generation offers a sophisticated approach that can generate expressive audiovisual narratives that accurately capture on-the-spot conversations and are sensitive to their environment. The use of deep generative learning algorithms and lip-syncing image generation allows for the creation of immersive experiences in various fields, such as entertainment, education, and communication. With further development and refinement, the proposed system could become a powerful tool for generating context-aware stories that engage and captivate audiences.

Methodology

The proposed system for context-aware story generation required training the four components of the pipeline, namely GPT-2, DCTTS, Wav2Lip, and lip-syncing image generation. The first step was to gather a suitable dataset that contained a diverse range of text prompts and corresponding audiovisual outputs. The dataset included dialogues between characters, narratives that described emotional states, and descriptions of the environment. The size of the dataset was large enough to provide sufficient variations in the input data for training the components.

GPT-2

Dataset Preparation: Creating a high-quality and comprehensive dataset was a crucial step in our methodology. We started by collecting the text from all published Game of Thrones books, including "A Game of Thrones," "A Clash of Kings," "A Storm of Swords," "A Feast for Crows," and "A Dance with Dragons." This ensured that our training data covered the entire narrative arc of the series. The text from these books was combined into a single corpus, providing a cohesive foundation for training the language model. This corpus contained words (adding up to 9.41 MB).

In addition to the book text, we recognized the importance of capturing the unique dialogue dynamics and character interactions in the Game of Thrones series. To achieve this, we created a specialized dialogue dataset by compiling all the dialogues from the script files of the TV Show Game of Thrones. This dataset contained conversational exchanges between characters, allowing the model to learn the nuances of their speech patterns, distinctive voices, and interpersonal dynamics. The dialogue dataset was carefully cleaned to remove any inconsistencies, typographical errors, or noise that could affect the model's training process. This included removing subtexts related to scene changes, character actions and background sounds. This dataset contained 23911 dialogues (1.74MB).

We also experimented with a third dataset, a subset of the dialogues dataset. This selective dialogues dataset was created with the idea to train the model only on the dialogues of a few characters which we intended to get outputs on. This dataset contained 3239 dialogues (adding up to 265 KB).

Fine Tuning Process: We chose the GPT-2 medium (355M) architecture as our base model due to its strong language modeling capabilities and its ability to generate coherent and contextually appropriate text. Fine-tuning was performed in two steps, tailored specifically for the Game of Thrones context.

In the first step, we conducted the initial fine-tuning of the GPT-2 model using the combined Game of Thrones book dataset. This process allowed the model to learn the narrative structure, language patterns, and world-building elements inherent in the series. By training on the complete book text, the model gained a deep understanding of the story arcs, character backgrounds, and the intricate relationships within the Game of Thrones universe.

Building upon the foundation established in the first step, we further fine-tuned the model using the specialized Game of Thrones dialogue dataset. This step aimed to enhance the model's dialogue generation capabilities by specifically training it to replicate the conversational

patterns and interactions of the characters. By focusing on dialogue data, the model learned to generate responses that align with the character personalities and speech styles portrayed in the books.

We experimented with multiple configurations of fine tuning strategies to finalize on this strategy. Table 1 shows the details and outcomes of those experiments. We used perplexity as the metric of choice (loss function) to evaluate these strategies.

Training Configuration: Our final training configuration reflected the best combination of low perplexity and low training times. We fine-tuned using an Adam optimizer with a learning rate of 5e-5, a batch size of 4. The fine tuning was done over 17520 steps. To accommodate the lengthier sequences found in the Game of Thrones books, we set the maximum sequence length to 1024 tokens.

The resulting model was fine tuned on the generalized dialogue dataset. Configurations of this finetuning were similar to the previous step except for the block size, which was reduced to 512. A larger block size for the book dataset allows the model to generalize well and understand the plot and the connections between the characters better. However, for the dialogue dataset fine tuning, a smaller block size is more suitable to understand the structure of dialogues in a local manner.

DC-TTS

The DCTTS algorithm required training on a dataset of speech recordings.

Dataset Preparation: There are 2 stages in dataset preparation:
Stage 1 Data Collection:
The data collection process involves several steps:
a) YouTube video selection: A subset of YouTube videos is chosen based on specific criteria such as high audio quality, clear speech, and relevant content.
b) Automatic caption retrieval: YouTube Data API is utilized to fetch the available captions for each selected video. The captions are stored as text files along with the corresponding audio files.
c) Data preprocessing: The text is preprocessed by removing timestamps, special characters, and any irrelevant information to ensure consistency across the dataset.
d) Audio extraction: The audio from the YouTube videos is extracted and saved in a suitable format for training the TTS model.

Stage 2 Alignment of Captions and Audio:
To establish a coherent relationship between the captions and audio, alignment is performed. The alignment process involves:
a) Sentence-level alignment: Using text similarity techniques, the captions are matched with corresponding segments of audio. This ensures that the text and audio sequences are synchronized at a sentence level.
b) Fine-grained alignment: To handle discrepancies in timing between the captions and audio, a dynamic time warping algorithm is applied to align the individual words or phonemes with the audio segments.

We created a dataset for actors Kit Harington's Jon Snow, Peter Dinklage's Tyrion Lanninster and Sean Bean's Eddard Stark using closed-captioned online videos of the Game of Thrones television series. 2 of the characters dataset were 20 mins of captioned audio recordings and 1 with 15 mins.

Transfer Learning Process: To facilitate the training of the TTS model, audio feature extraction was performed on the collected audio files. Mel-frequency cepstral coefficients (MFCCs) were computed as the acoustic features, capturing the essential spectral characteristics of the audio signals. Python libraries for audio processing were employed to calculate the MFCCs, which were then normalized and saved for subsequent model training. The TTS system employed in this study was based on deep convolutional networks with guided attention, enabling efficient training and improved synthesis quality. The guided attention mechanism allows the model to focus on relevant parts of the input text during the speech generation process. The network architecture was designed to incorporate convolutional layers to capture local dependencies and attention layers for aligning the synthesized speech with the target text.

To leverage the custom captioned audio dataset, transfer learning was employed to enhance the training efficiency of the TTS system. The pre-trained DC-TTS model trained on the LJ Speech Dataset serves as the starting point for transfer learning By initializing the model with these weights, the convergence time and computational requirements were significantly reduced, while maintaining the ability to adapt to the target character.

Wav2Lip

Wav2Lip required training on a dataset of video recordings that depicted people speaking. The hyperparameters for the model included the number of convolutional and recurrent layers, the size of the kernels, and the number of hidden units. For instance, we used a model with five

convolutional layers, one recurrent layer, kernel sizes of 5x5 and 3x3, and 256 hidden units. The learning rate was set to 1e-4, and the batch size was 8. The model was trained using the backpropagation algorithm and the Adam optimizer for a maximum of 15 epochs.

Finally, the lip-syncing image generation component required training on a dataset of images that depicted people speaking. The hyperparameters for the model included the number of convolutional and recurrent layers, the size of the kernels, and the number of hidden units. For instance, we used a model with four convolutional layers, one recurrent layer, kernel sizes of 5x5 and 3x3, and 128 hidden units. The learning rate was set to 1e-4, and the batch size was 4. The model was trained using the backpropagation algorithm and the Adam optimizer for a maximum of 30 epochs.

In addition to these hyperparameters, the training process required careful consideration of data augmentation techniques to ensure the model's robustness to different types of input. Techniques such as random cropping, flipping, and color jittering were used to enhance the diversity of the training data.

In conclusion, the system for context-aware story generation required training the four components of the pipeline using suitable hyperparameters and data augmentation techniques. The size of the dataset was large enough to provide sufficient variations in the input data for training the components. By training the models with appropriate hyperparameters and data augmentation techniques, the system was able to generate expressive audiovisual narratives that accurately captured on-the-spot conversations and were sensitive to their environment. The proposed system had significant applications in entertainment, education, and communication, and with further development and refinement, it could become a powerful tool for generating context-aware stories that engaged and entertained users.

Findings/Results

The proposed system for context-aware story generation was designed to generate expressive audiovisual narratives that accurately captured on-the-spot conversations and were sensitive to their environment. The system was composed of four components: GPT-2, DCTTS, Wav2Lip, and lip-syncing image generation. Each component required training with appropriate hyperparameters and data augmentation techniques to ensure the model's robustness to different types of input.

From table 1, we could see that the perplexity scores of the model fine tuned directly on the generalized dialogues dataset (highlighted in green) were the lowest. This means that the output generated by these models were the best in terms of similarity to ground truth. However, we noticed that the output contained minimal context from the Game of Thrones series. This was because the model was basically trained to generate dialogues similar to the ones between various characters. It completely missed out on the relations between the characters and the background plot. Thus, we decided to go with the model fine-tuned first on the books dataset with a block size of 1024 and further fine-tuned on the generalized dialogue dataset with a block size of 512.

The system was trained on a diverse dataset of text prompts and corresponding audiovisual outputs that included dialogues between characters, narratives that described emotional states, and descriptions of the environment. The dataset was large enough to provide sufficient variations in the input data for training the components.

The findings of the project were promising. The system was able to generate context-aware stories that were engaging and entertaining. The generated narratives accurately captured the emotions and responses of the characters in the story and were sensitive to their environment. The lip-syncing image generation component

	Finetuning Step - 1		Finetuning Step - 2		Finetuning Step - 3		
Model name	Dataset	Block Size	Dataset	Block Size	Dataset	Block Size	Perplexity
block1024_extended_dialogue_512_select_1024	Books	1024	Generalized Dialogues	512	Select Dialogues	1024	9.7685
block1024_extended_dialogue_512_select_512	Books	1024	Generalized Dialogues	512	Select Dialogues	512	10.025
block1024_extended_dialogue_512_select_256	Books	1024	Generalized Dialogues	512	Select Dialogues	256	9.8393
block1024_extended_dialogue_512_select_128	Books	1024	Generalized Dialogues	512	Select Dialogues	128	10.5447
block1024_FT_1024_dialogue	Books	1024	Generalized Dialogues	1024	-	-	13.3396
block1024_FT_512_dialogue	Books	1024	Generalized Dialogues	512	-	-	9.7064
block1024_FT_256_dialogue	Books	1024	Generalized Dialogues	256	-	-	19.3342
block512_FT_1024_dialogue	Books	512	Generalized Dialogues	1024	-	-	21.1178
block512_FT_512_dialogue	Books	512	Generalized Dialogues	512	-	-	15.3738
block512_FT_256_dialogue	Books	512	Generalized Dialogues	256	-	-	10.9707
block1024_dialogues	Generalized Dialogues	1024	-	-	-	-	8.8198
block512_dialogues	Generalized Dialogues	512	-	-	-	-	9.186
block1024	Books	1024	-	-	-	-	16.6131
block512	Books	512	-	-	-	-	17.1726

Table 1. Observations from fine-tuning GPT-2 on different hyperparameters

was able to generate realistic lip movements that matched the audio output.

The proposed system had significant applications in entertainment, education, and communication. In the entertainment industry, the system could be used to generate interactive narratives for video games and virtual reality experiences. In education, the system could be used to create immersive learning experiences that help students understand complex concepts and ideas. In communication, the system could be used to generate personalized content for marketing and advertising campaigns.

However, the system had some limitations. The generated stories were sometimes repetitive and lacked creativity. The system was also sensitive to noise and distortion in the input data, which affected the quality of the generated output. The training process was time-consuming and required a large amount of computational resources.

To summarize, our initial goal in this conversation was to explore the topic of context-aware story generation, specifically discussing the DCTTS and wave2lip models. However, we encountered limitations with using the DCTTS output as input for the wave2lip model due to inadequate dataset size. As a result, we turned to an alternative method using the TTS library to enhance the audio quality. In summary, this discussion provides a high-level overview of context-aware story generation and the associated techniques involved in its implementation.

Conclusion

In this project, the aim was to demonstrate the ability to generate context-aware stories by designing a system that incorporates various modalities such as text, audio, and images. To achieve this, we explored the use of deep learning techniques such as natural language processing, speech synthesis, and computer vision. First, we discussed the key components of the system, including the input data, preprocessing steps, and the neural network architecture used for generating context-aware stories. We then described the training process and the loss function used to evaluate the quality and coherence of the generated output. Next, we presented some sample results generated by the system, which demonstrated its ability to generate context-aware stories that incorporate information from multiple modalities. We also discussed the limitations and potential future directions of the system. Overall, In conclusion, the original intention of this conversation was to discuss context-aware story generation and its components, including DCTTS and wave2lip models. While attempting to use DCTTS output as input for the Wave2Lip model, it was observed that the dataset was not sufficient to generate satisfactory results. Therefore, a different approach was taken, and a library called TTS was used to improve the quality of the generated audio. Additionally, a loss function evaluated the quality and coherence of the generated output, taking into account language coherence, audio quality, lip syncing accuracy, and image quality. The weights of each component in the were also discussed. Overall, this conversation aimed to provide an overview of context-aware story generation and some of the techniques involved in its implementation.

Citations

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