

DIGITAL AUDIOBOOK

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Abstract:

The idea of this paper is to build an automated virtual reader. In this modern era, there is an urge for an automated reader which is cost-effective, accurate, and also portable at the same time. The major project involves the complete design of the Digital Audio Book.

In this digital audiobook, we aimed to develop a convertor and a translator of an eBook. An eBook is a digital representation of a printed book. In this, we can convert an eBook into an audio source. Web Speech module is used to convert the text to audio format and vice versa. Here we are with the Digital audiobook. It has features like text to speech, speech to text, and a translator. During the pandemic period, everything went through online itself. Especially in Educational Institutions and they provide notes to students in the form of text. It's very difficult to study by looking and learning on a screen. So, with the help of Digital audiobooks, we can overcome these problems.

Keywords —Web Speech API , JavaScript, HTML, TTS, SST, Translator

I. INTRODUCTION

For many years, scientists and engineers have had the speech production process with the goal of building a system that can start with text and produce speech automatically. Sometimes we prefer listening to the content instead of reading we can multitask while listening to the critical file data.

Whenever you try to translate a word or a sentence from a certain language to another language. Web Speech provides many APIs to convert text to speech, speech to text, and translator. It is very easy to use the tools and provides many built-in functions which are used to save the text file an mp3 file.

The text to speech synthesis is Synthesizing speech from the text (TTS). TTS systems have based on the complex pipeline. A speech synthesizer is one of the major applications of NLP. The conversation of text to speech involved three important stages: text analysis, text processing, and waveform generation, i.e., speech formation.

A speech synthesizer is an application that converts written text into speech, the user enters a text and gets output as a sound. In this survey, Speech Synthesis is becoming one of the most important steps toward improving the human interface to the computer. The objective of text to speech is to convert arbitrary text into spoken

waveform. The quality of a speech synthesizer is based on two factors naturalness and intelligibility.

II. LITERATURE SURVEY

In 2018 a paper was published by Sneha.C , S.B. Gundre about “OCR Based Image Text toSpeech Conversion Using MATLAB” in this project There are millions of blind people inthe world who are visually impaired. The disability to read has a large impact on the life ofvisuallyimpairedpeople.TheProposedsystemisco st-efficientandhelpsthevisuallyimpaired person to hear the text. The main idea of this project is optical Character recognitionwhich is used toconverttextcharactersinto the audiosignal.

In 2017 a paper was published by H Rithika, B Nithya Santhoshi “Image text to speech conversion in the desired language by translation with Raspberry Pi”in this project The mainproblem incommunication islanguagebias betw eenthecommunicators.Thisdevicebasically can be used by people who do not know English and want it to be translated into their native language

In 2019 a paper was published by Jing-Xuan Zhang, Zhen-Hua Ling, Yuan Jiang “Improving sequence-to-sequence voice conversion by adding text supervision”.Voice conversion (VC) is a task that alters the voice of a person to suit different styles while conserving linguistic content. Previous state-of-the-art technology used in VC was based on the sequence-to-sequence (seq2seq) model, which could lose linguistic information.

In 2021 a paper published by D. C. Tran, H. S. Ha, and A. Khalyasmaa, “A Question Detection Algorithm for Text Analysis,” In this paper, an FPT.AI-based text-to-speech (TTS) application is developed that converts Vietnamese text into spoken words. The application is developed based on Django for Python and in the form of an interactive web page which is connected to an FPT.

III. EXISTING SYSTEM

Previously so many members used the tts module in order to convert the text to speech and some members used Optical Character Recognition to convert the text to speech. So many of the authors using the tts module in python for thisproject. Some authors give features like text to speech and speech to text. But tts only reads the typed text it does not read the text. They encouraged todo this text to speech for impaired persons

IV. PROPOSED SYSTEM

In this proposed method system, we are overcome to solve the problems faced by the students during the learning process. Here we are with the Digital audio book. It has features like text to speech, speech to text, and a translator. During the pandemic period, everything went through online itself. Especially in Educational Institutions and they provide notes to students in the form of text.It’s very difficult to study by looking and learning on a screen. So, with the help of Digital audio books, we can overcome these problems

A. BLOCK DIAGRAM

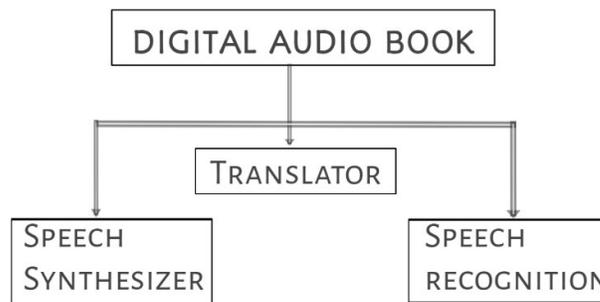


Fig1:Digital Audio Book

Digital Audio Book consists of a speech synthesizer, speech recognition, and translator. we are going to do this with the help of a web speech API in JavaScript.

B. SPEECH TO TEXT

The speech is converted into text by using web speech API.

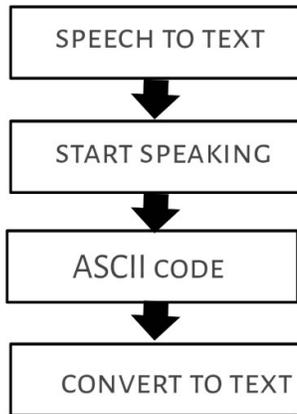


Fig 2: *speech to text*

D. TRANSLATOR

The translator can translate the information provided by the user

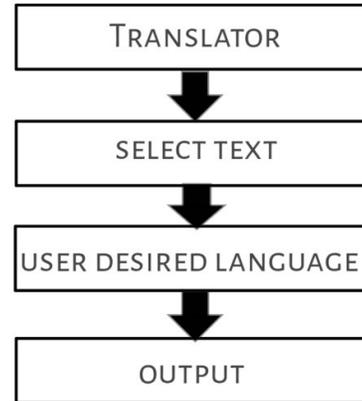


Fig4: *Translator*

C. TEXT TO SPEECH

The text is converted into speech by using web speech API.

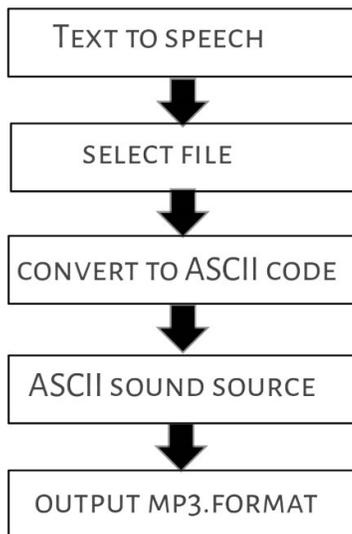


Fig 3: *text to speech*

V. SOFTWARE USED WEBSPEECH API

The Web Speech API enables you to incorporate voice data into web apps. The Web Speech API has two parts: Speech Synthesis (Text-to-Speech), and speech Recognition (Asynchronous Speech Recognition).

A. SPEECH SYNTHESIS

Let's create an instance of the `SpeechSynthesisUtterance` class. We'll configure this instance with various properties.

The Speech Synthesis Interface of the Web Speech API is the controller interface for the speech service; this can be used to retrieve information about the synthesis voices available on the device, start and pause speech, and other commands besides.

i. PROPERTIES

`SpeechSynthesis` also inherits properties from its parent interface, `EventTarget`.

SpeechSynthesis.paused Read-only

A Boolean value that returns true if the `SpeechSynthesis` object is in a paused state

SpeechSynthesis.pending Read-only

A Boolean value that returns true if the utterance queue contains any unspoken utterances.

SpeechSynthesis.speaking Read-only

A Boolean value that returns true if an utterance is currently in the process of being spoken even if Speech Synthesis is in a paused state.

ii. METHODS

SpeechSynthesis also inherits methods from its parent interface, **EventTarget**.

SpeechSynthesis.cancel()

Removes all utterances from the utterance queue.

SpeechSynthesis.getVoices()

Returns a list of **SpeechSynthesisVoice** objects representing all the available voices on the current device.

SpeechSynthesis.pause()

Puts the **SpeechSynthesis** object into a paused state.

SpeechSynthesis.resume()

Puts the **SpeechSynthesis** object into a non-paused state: resumes it if it was already paused.

SpeechSynthesis.speak()

Adds an utterance to the utterance queue; it will be spoken when any other utterances are queued before it has been spoken.

B. SPEECH RECOGNITION

It is an interface to the web speech API; it is used to convert speech to text.

i. PROPERTIES

SpeechRecognition also inherits properties from its parent interface, **EventTarget**.

SpeechRecognition.Grammar

Returns and sets a collection of **SpeechGrammar** objects that represent the grammar that will be understood by the current **SpeechRecognition**.

SpeechRecognition.Lang

Returns and sets the language of the current **SpeechRecognition**. If not specified, this defaults to the HTML **Lang** attribute value, or the user agent's language setting if that isn't set either.

SpeechRecognition.Continuous

Controls whether continuous results are returned for each recognition, or only a single result. Defaults to **false**.

SpeechRecognition.interimResults

Controls whether interim results should be returned (**true**) or not (**false**). Interim results are results that are not yet final (e.g., the **SpeechRecognitionResult.final** property is **false**).

ii. METHODS

SpeechRecognition also inherits methods from its parent interface, **EventTarget**.

SpeechRecognition.Abort()

Stops the speech recognition service from listening to incoming audio, and doesn't attempt to return a **SpeechRecognitionResult**.

SpeechRecognition.Start()

Starts the speech recognition service by listening to incoming audio with the intent to recognize grammar associated with the current speech recognition.

SpeechRecognition.Stop()

Stops the speech recognition service from listening to incoming audio, and attempts to return a speech recognition result using the audio captured so far.

SpeechGrammar()

Creates a new **SpeechGrammar** object.

SpeechGrammar.src

Sets and returns a string containing the grammar from within the **SpeechGrammar** object instance.

C. TRANSLATOR

QuerySelector()

It is used to add the CSS property to the HTML element (or) query selector method returns the first element that matches a CSS property

Google Translate API

Google Translate is a free multilingual machine translation service. It can translate the Website's text content from one language to another. It offers a huge list of languages to translate and has an efficient, reliable, and easy way to translate the webpage into whatever language the user wants. It supports over 100 languages. Use this website translator to convert webpages into your choice of language

VI. RESULTS AND DISCUSSION

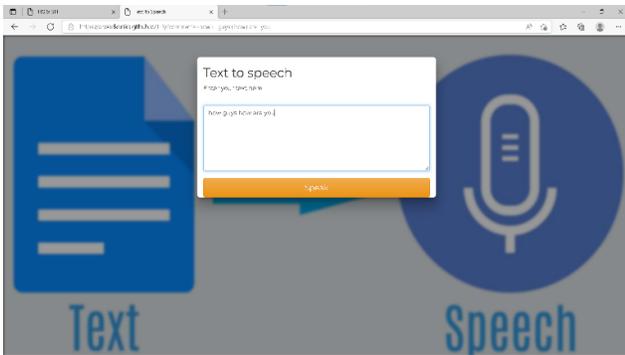


Fig 5: text to speech output

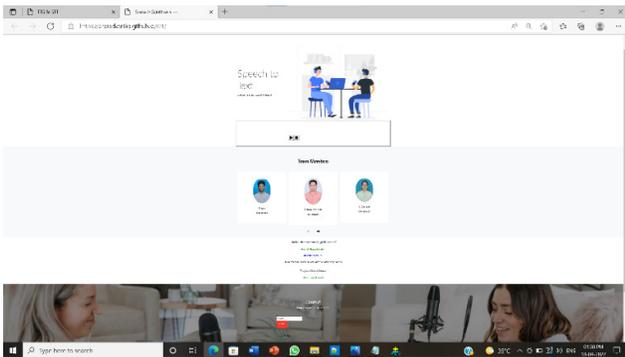


Fig 6: Speech to text output

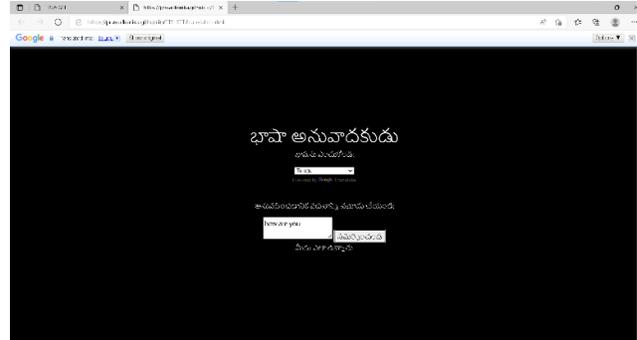


Fig 7: Translator output

The main motto of TTS is to convert arbitrary text into waveform. Speech generation is the generation of an acoustic waveform corresponding to text and each of these units in the sequence. This involved text analysis, text normalization, text processing, grapheme-to-phoneme conversion, and speech synthesis.

Text analysis which analyses the input text such as dividing the text into words and sentences. Text normalization is the transformation of text into the pronounceable form, it is the front end of TTS that assigns phonetic transcription to each and every word. The process of assigning phonetic transcription to a word is called grapheme to phoneme conversion.

The phonetic analysis also known as word analysis focuses on the phone within the word. Finally symbolic linguistic representation produces sound.

The general objective of the project is to develop a digital audiobook for the students. The specific objectives are:

- It is able to help people who struggle with reading
- It is able to convert the speech to text
- It is able to help you in learning languages that you do not know.
- It is able to help the students in listening to e-

booksore-materialduring the examination.

VII CONCLUSION

Speechsynthesis has been developed steadily over the last decades and it has been incorporated into several new applications. For most applications, the intelligibility and comprehensibility of synthetic speech have reached an acceptable level. However, in prosodic, text preprocessing, and pronunciation fields there is still much work and improvements to be done to achieve more natural-sounding speech. Natural speech has so many dynamic changes that perfect naturalness may be impossible to achieve. However, since the markets for speech synthesis-related applications are increasing steadily, the interest in giving more effort and funds to this research area is also increasing. Present speech synthesis systems are so complicated that one researcher cannot handle the entire system. With good modularity, it is possible to divide the system into several individual modules whose developing process can be done separately if the communication between the modules is made carefully.

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