"Speech Interface for Form Filling with Biometric Recognition and Authentication"

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ABSTRACT

Speech recognition technology has several real-world applications and security is imp for this paper discusses the algorithm that is used in speech to text online form fill up application for different organization. Our aim to aware more people to know about this technology has become familiar with this concept. These applications that are speech to text can drastically improve the way users, especially those with disabilities, perform tasks. In a website, users could navigate pages or populate form fields using their voice. Users could also interact with a page while driving, without taking their eyes off of the road. These are not trivial use cases.

Keyword: Speech to text , HTML , Java script

I INTRODUCTION

The speech to text technology is introduced at the end of 2012,initially it allows web developers to provide speech input and text-to-speech output features in a web browser. Typically, these features aren't available when using standard speech recognition or screen reader softwaretechnology .This API takes care of the privacy of the users and keep record of the speech to text conversions happen . Before allowing and recognizing the speech. A website to access the voice via microphone, the user must explicitly grant permission. Interestingly, the permission request is the same as the user get access the microphone , although it doesn't need the webcam because the speech conversion is our main purpose. If the page that runs this speech totext technology uses the HTTPS protocol, the browser asks for the permission only once, otherwise it does every time a new process starts.The speech to text conversion which helps to recognize the voice of the person and convert into desirable text as like as to fill up the form.

The Web Speech to text technology which is very necessary part of our technology defines a complex interface, called Speech Recognition, whose structure can be seenin the complex form .This paper won't cover all the properties and methods described in the specification for two main reasons. The first is that if we have seen the interface, it's too complex to be covered in one paper. Secondly, as we'll see in the next sections, there is only one browser that supports this and its implementation is very limited. Therefore, we'll cover only the implemented methods and properties.

The specification asserts that the speech to text itself is agnostic of the underlying speech recognition and synthesis implementation technology and it can support both server-based and client-based/embedded recognition and technology . It allows two types of recognition: one-shot and continuous both necessary for different situations. In the first type of technology the recognition ends as soon as the user stops talking, while in the second technology it ends when the stop() method is called. In the second case, we can still allow our users to end the recognition by attaching a handler that calls the stop() method (via a button for example). The results of the recognition are provided to our code as a list of hypotheses and that could be understand , along with other relevant information for each hypothesis needed

Another interesting feature of the Web Speech API that is speech t text technology is that it allows you to specify a grammar objectinterpretive in different languages . Explaining in detail what a grammar iswhich is very important aspects. You can think of it as a set of rules and instructions which form a program for defining a language. The advantage of using a grammar is that it usually leads to better results due to the restriction of language possibilities the comparison can also be made

This API speech to text may not surprise you because of the the already implemented x-webkit-speech attribute introduced in Chrome 11 that is very important in our consideration. The main differences is that the Web Speech to text API allows you to see results in real time and utilize a grammar with different language with their set of eyes and instructions. You can read more about this attribute, by taking a look at How to Use HTML5 Speech Input Fields.

Now that you should have a good overview of what this speech to text technology is and what it can do, let's examine its properties and methods.

II METHODOLOGY AND ALGORITHM

To instantiate a speech recognizer, to start the speech to text conversion, use the function speechRecognition() as shown below:

```
var recognizer =newspeechRecognition();
```

This object exposes the following methods:

- onstart: Sets a callback for that is fired when the recognition service has begun to listen to the audio which is very important aspects with the intention of recognizing.
- onresult: Sets a callback that is fired when the speech recognizer returns a which is in the form of text . The event must use the SpeechRecognitionEvent interface.
- onerror: Sets a callback that is fired when a speech recognition error occurs. If any error occurs .It shud be recover event must use the SpeechRecognitionError interface.
- onend: Sets a callback that is fired when the service has disconnected and stops. The event must always be generated when the session ends, no matter what the reason that is very important point in this technology

In addition to these methods, we can configure the speech recognition object using the following properties:

- continuous: Sets the rule and instruction for the type of the recognition (one-shot or continuous). If its value is set to true we have a continuous recognition, otherwise the process ends as soon as the user stops talking. By default it's set to false that this help in the time management
- lang: Specifies the recognition language. By default it corresponds to the browser language which gives the use of different languages with their attributes
- interimResults: Specifies if we want interim results about the technology used. If its value is set to truewe'll have access to interim results that we can show to the users to provide feedback. If the value is false, we'll obtain the results only after the recognition ends. By default it's set to false.

As the argument to the result event handler, we receive an object of type SpeechRecognitionEvent. The latter contains the following data:

- results[i]: An array containing the results of the speech recognition. Each array element corresponds to a recognized word.
- resultIndex: The current recognition result index.

- results[i].isFinal: A Boolean that indicates if the the result is final or interim.
- results[i][j]: A 2D array containing alternative recognized words. The first element is the most probable recognized word.
- results[i][j].transcript: The text representation of the recognized word(s).
- results[i][j].confidence: The probability of the result being correct. The value ranges from 0 to 1.

III CHROME BROWSER COMPATIBILITY

The previous section pointed out that the proposal for the Web Speech API was made in late 2012. So far, the only browser that supports this API is Chrome, starting in version 25 with a very limited subset of the specification. Additionally, Chrome supports this API using the webkit prefix. Therefore, creating a speech recognition object, looks like this in Chrome:

```
var recognizer =newwebkitSpeechRecognition();
```

IV RESULT AND DISCUSSION


C.G.P.E.T ONLINE FORM FILLUP
Step-by-step procedure to fill CG PET Application Form 2019
Given below is the procedure to fill and submit the application form of CG PET 2019:
Enter the required details: Candidates will have to enter the following details:
Candidate's Name (As in 10th mark sheet)
Father's Name Mr./shri
Mother's Name Mrs./Shrimati
Date of Birth DD/MM/YY
Roll Number of Class 10
Gender- M/F
Category (Caste) General/SC/ST/OBC
CG Domicile YES/NO
Physically Handicapped (PH)YES/NO
Fee
Height (cms)
Weight (Kgs)
Religion
Are you a J&K migrant?
Two Identification Marks

Figure 4.1 SPEECH TO TEXT ONINE FORM FILLUP

V CONCLUSION AND FUTURE SCOPE

This paper introduced the speech to text online form fillup using html and explained how it can help improve user experience, especially for those with disabilities. The implementation of this speech to text online form fillupis at a very early stage, with only Chrome offering a limited set of features. The potential of this speech to text online form fillup is incredible, so keep an eye on its evolution. In future the database and different online form fillup application with advanced feature will be developed.

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