Speak_Pseudocode2c A_framework Report

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Currently, we are surrounded by a world dependent on AI-based speech assistants which are built to make many trivial tasks in our lives easy. For most of us, the ultimate luxury would be an assistant who always listens for your call, anticipates your every need, and takes action when necessary. That luxury is now available thanks to artificial intelligence assistants, aka voice assistants. Even these voice assistants are implemented through millions of lines of code. Today, coding is being taught from primary levels at school. According to the new education policy, programming has been introduced as a compulsory course for students from junior classes.

At the moment, there is very less support provided to help these students understand the programming concepts better. With the surge in the number of programmers and advancements in technology, the increasing competition among students can be very intimidating for a child. Also, the c code given in textbooks is empirically observed to be not very easy for students to comprehend at the school level when compared to natural language pseudocode. Editing huge lines of unformatted code can sometimes be frustrating even for a professional programmer. Therefore, our goal is to use a voice-based digital assistant to automate the process of writing and editing code by providing our voice input.

Our aim for this project is to make these children familiar with the coding environment so that they do not shy away from programming and they have an all time accessible tool to help them in their programming journey. This tool will take a user's voice input as pseudocode and convert it into a c program, given that the user gives the correct pseudocode as voice input.

Speak Pseudocode2c: A framework to convert customized pseudocode to c code. Today, coding is introduced in the early stages of school to prepare them better in the programming world. But this alone is not enough for students to get a good grip on programming. Introducing coding at an early age comes with its own challenges. These challenges include getting familiar with a totally different language when they are also coping with other subjects in their school curriculum, not having a proper platform to help them with coding i.e not getting enough practical experience with coding. Our tool will help a student with the experimental aspect of coding. Writing and implementing huge lines of code can be a great deal of hassle for a coder. Doing everything manually can be a real pain sometimes. This tool can help programmers with editing their code just by providing their voice input. Manual labor will be decreased to a great extent.

We all use speech recognition in our daily lives. However, the problem with these is that sometimes they don't respond due to some fault in the input taken or some misunderstanding in the meaning of speech. Many studies are going on to minimize the errors. These errors can be caused due to incorrect meaning, substituted words or some missing phrases in the input taken.

This study proposes the method to address the problem of choosing between different speech models. It combines 2 stages using Microsoft API and Google API model voice.

Recent research shows that the most popular operating systems like Android and Microsoft so Google and Microsoft API are used in this research.

Windows Speech API is used widely across the globe for speech to text conversion. All the microsoft windows systems use this speech API integrated inside windows Cortana. This application interface can be adapted as a toolbox which allows users to implement speech recognition applications.

Microsoft has improved its engines with deep neural networks and hidden markov models. This enables long speech recognition.

The main advantage of Google Cloud API is the ability to support real time speech to text conversion. It also has multiple language support. It is implemented using deep neural networks. The error rate has reduced from 23% in 2013 to 8% in 2015. Speech recognition is generally measured using Word Error Rate(WER). Some popular cloud speech APIs have the following Word Error Rate:

- Google API: 9%
- Microsoft API: 18%
- CMU Sphinx: 37%

Some studies have shown that when the audio is recorded using a microphone in the form of stereo, the error rate drastically reduces to 3%.

The speech is generally taken input through a microphone in .wav format and has a frequency of 16,000Hz. The input speech file is transformed to characters and phrases, called phonemes, using speech to text engines.

There is a detection value defined between 0 to 1 where 1 means detectable speech and 0 is undetectable. If the value is less than 0.5, then the speech will be passed through Google API and then the detection value is recalculated.

The mechanism behind this is that Microsoft API will add words and retrieve variables to compare words and speech. Whereas Google API will recode the speech and transform to WAV 1,600Hz using a wave library. Later on, the speech recognition happens via Google cloud speech API and it returns JSON string

Each speech recognition API uses different processing algorithms so it is best to combine their abilities in order to get the best out of them. For the testing stage, Word Error Rate was calculated. The results are as follows:

- Google API: 11%
- Microsoft API: 14%
- Multi Pipeline Combination: 6%

The recognition accuracy for 70dB sound are:

- Microsoft API: 65%
- Google API: 82%
- Multi Pipeline Combination: 86%

Pseudocode is written in natural language along with mathematical expressions which are useful for describing the source code. Pseudocode eases the process of understanding the written code for programmers irrespective of the programming language. However, writing pseudocode for a particular source code is not an easy task as it takes extra time. The problem which the authors are trying to solve is to generate pseudocode from given source code automatically and to achieve this goal, they have built a neural network with the help of deep learning using Long Short-Term Memory architecture.

The input to code2pseudocode model is source code written in python language and the output is the natural language pseudocode with mathematical notations in it. Each line of code is tokenized i.e each statement is considered as a sequence of tokens. For example:

$$os.remove(fname) \longrightarrow os, ., remove, (, fname,)$$

The encoder and decoder are crucial components of the code2psuedocode model. Both of the components are Recurrent Neural Network (RNN) which has a number of Long Short-Term memory (LSTM) units. LSTM is a special neuron for reciting long-term dependencies. It has an internal state variable that is transferred from one cell to the next and is changed using operation Gates. It is intelligent enough to know how long to keep old records, what to recall and forget, and how to make connections between old memory and new knowledge.

For translation of long statements, instead of translating reading the complete sentence at once they translated one phrase at a time towards the completion of the sentence. An attention layer is applied to the model to decide the contribution of each token in the input chain, so that when forecasting a new output token, previous token contribution accounts for the prediction outcomes.

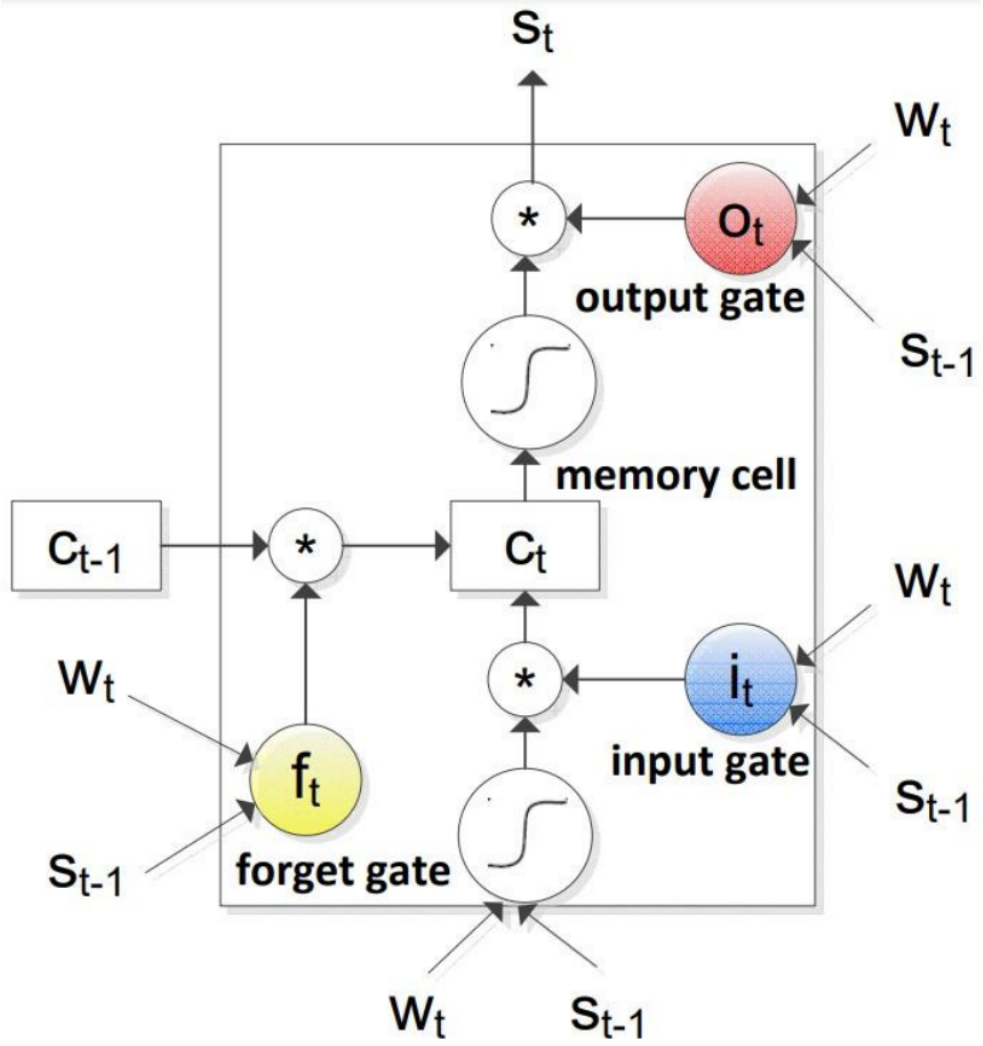


Figure 3.2(a): The internal structure of an LSTM unit

The authors use a technique called teacher-forcing technique. The Ideology behind teacher-forcing technique is: Consider a math exam question with several sections, of which the answer to part (a) is necessary for the equation in part (b), the answer to part (b) is required for part (c), and so on. To get the correct answer, we must pay extra attention to these questions because if part (a) is incorrect, all subsequent parts will be incorrect as well, despite the fact that the formulas and equations are correct. Similarly, Teacher Forcing takes care of this in the following manner: When we get an answer for part (a), a teacher can equate it to the right one, record the

score for part (a), and remind us what the correct answer is so we can use it for part (b). This is a snippet of their sample result which they have added in their paper.

Python Code	Pseudo-Code
Manually-Generated Pseudo-Code	
`def npgettext(context, singular, plural, number)`	# Define the function "ngettext" with 4 arguments: context, singular, plural, and number.
`del self._expire_info[key]`	# Delete the value under key "key" of "self._expire_info".
`return self.render()`	# Call the method "self.render", return the result.
`yield representative, True`	# Yield representative and boolean True.
Code2Pseudocode Encoder-Decoder	
`def npgettext(context, singular, plural, number)`	# Define the function "npgettext" with 4 arguments: context, singular, plural, and number.
`del self._expire_info[key]`	# Delete the value under the key "key" of the "self._expire_info" dictionary.
`return self.render()`	# Call the method "self.render", return the result.
`yield representative, True`	# return is boolean True.

Figure 3.2(b): Sample output from CODE2PSEUDOCODE model

For benchmarking and comparing their result with other techniques, they used the same dataset as used by other authors implementing different techniques. The sample dataset which is used is retrieved from the code base of Django which is a framework for web application. They hired an engineer to extract it's source code which is written in python and create pseudocode for that particular source code. They obtained 18,800 pairs of code along with pseudocode for their dataset. To measure the quality of generated pseudocode they used a scale called Bilingual Evaluation Understudy (BLEU). BLEU is a technique to evaluate the quality of text when text is converted from one machine translated language to another natural language.

Pseudo-Code Generator	BLEU%
Code2Pseudocode	54.78
T2SMT	54.08
NoAtt	43.55
PBMT	25.17
SimpleRNN	06.45

Table 3.2(a): Percentage values of BLEU score for our CODE2PSEUDOCODE and the two techniques proposed in: Phrase-Based Machine translation (PBMT) and Tree-To-String Machine Translation (T2SMT).

The problem which we are trying to solve is the reverse of the problem defined above i.e converting pseudocode to source code but our approach is completely different because we are trying to build a framework which links various predefined functionalities/applications to achieve the goal. For taking pseudocode as input, we are giving facility to user to speak the pseudocode

(close to natural language) instead of typing it and the voice input will be displayed on the monitor by using the Google Cloud's Speech-To-Text API and that pseudocode will be processed and automatically converted into source code using our custom defined mapping library.

In communication in language innovation applications, for example, voice-controlled brilliant partners like the Amazon Echo or Google Home, named element acknowledgment (NER) is a basic job. Its aim is to find instances of named entities in text.
As new expressions are gotten from the framework over the long run, NER models are retrained at ordinary spans. Word-level representations are used in current models, or they are combined with character-level representations.. Shortcomings of such models −

1. Large memory requirements and longer training time.
2. Combination with other subword units is ignored leading to low accuracy.
3. Out-of-vocabulary (OOV) words can pose a problem.

The neural solution used in this paper is based on subword units, such as letters, phonemes, and bytes. We take in portrayals from every one of the three subword units for each word in an expression.

The model learns a low-dimensional representation from each subword unit (character-, phoneme-, and byte-level) for each word in an utterance, which is then concatenated and fed into a bidi- rectional LSTM-CRF model.. LSTM implementation used by the model−

$$
\begin{aligned}
i_t &= \sigma(W_{xi}x_t + W_{hi}h_{t-1} + b_i) \\
c_t &= (1 - i_t) \odot c_{t-1} + \\
&\quad i_t \odot tanh(W_{xc}x_t + W_{hc}h_{t-1} + b_c) \\
o_t &= \sigma(W_{xo}x_t + W_{ho}h_{t-1} + W_{co}c_t + b_o) \\
h_t &= o_t \odot tanh(c_t),
\end{aligned}
$$

Figure 3.3(c): LSTM implementation that was adopted by Lample

where W 's are shared weight grids, b's are the inclinations, is the component astute sigmoid capacity, xt addresses the token at position t, ht is covered up state at time t.
A softmax layer is utilized on top of the yield of the bidirectional LSTM organization to figure a likelihood appropriation of yield labels for a given token.

1. To examine the yield of individual subword units and different blends of subword units, both with and without the utilization of word-level embeddings.
2. Compare the output of subword-only models, dedicated word-level embedding models, and models that combine both.
3. Look at whether subword units can help with out-of-vocabulary terms.

A wide genuine world dataset that incorporates American English (EN), German (DE), French (FR), and Spanish (ES) (ES). The data depends on client demands for voice-enacted gadgets that were physically deciphered and clarified with named elements.

1. Different subword units provide the same results for different languages when used individually.

2. For all languages, results improve when many subword units are combined, and for all but French, the best results are obtained when all of the subword units are combined.
3. The best combination for French is characters and bytes.
4. Additive gains are acknowledged by using several subword units compared to using only one.

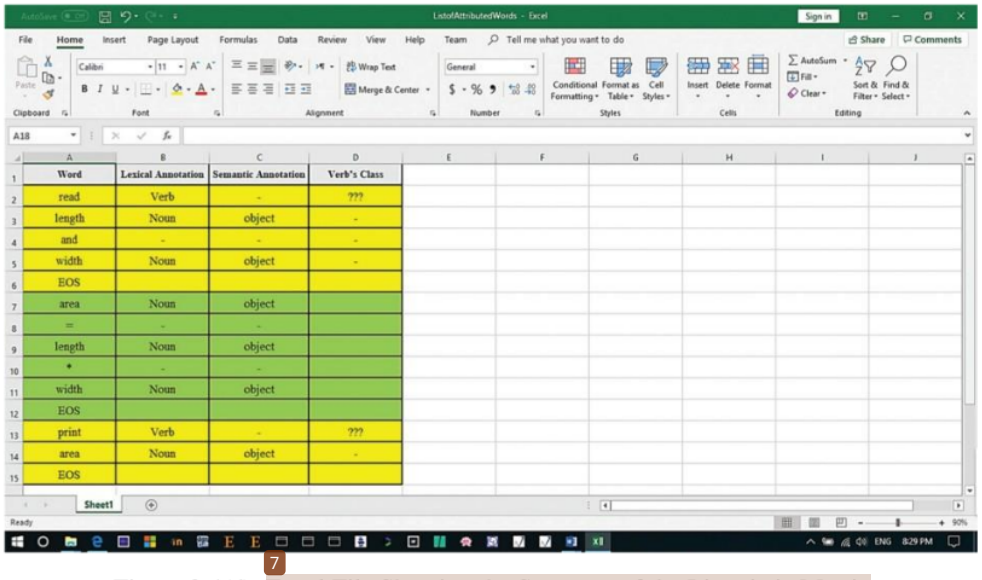
1. With larger training data, performance of models trained solely on subword units approaches that of models with dedicated word-level embeddings (91.35 vs 93.92 F1 for EN), but with smaller vocabulary size, performance of models trained solely on subword units falls short of that of models with dedicated word-level embeddings.
2. In the presence of out-of-vocabulary terms, the subword-based model outperformed the corresponding word-level model (34.9 vs 34.8), whereas the combined model achieved 37.1 F1, suggesting that subword units are useful.
3. Combining subword units improves performance.

The paper talks about a few techniques to enhance and make possible what the author calls the process to computerize the structure of a programming language code from pseudocode, which is seen here as an interpretation interaction for a characteristic language text, as pseudocode is an arranged book in normal English language.

The first tool it talks about is an Automatic Code Generator, abbreviated as ACG, which uses programs to generate source code which humans would otherwise have to write. They can be either passive or active. Passive code generators generate code that needs some modification by the user before running, whereas active code generators are embedded in the software development process.

The paper also talks about the use of Natural Language Processing (NLP) techniques to format the input in various ways to take multiple inputs/ slangs but generate the same source code which the user wants. This helps in clubbing various actions and making the natural language used in the approach to be more easy and not a fixed or particular pseudo code language.

The author proposes the “CodeComposer System”, which is a Machine Translation tool that uses Semantic Role Labelling(SRL) and NLP techniques to identify and convert pseudocode to mapped C # code.



Word	Lexical Annotation	Semantic Annotation	Verb's Class
read	Verb	-	???
length	Noun	object	-
and	-	-	-
width	Noun	object	-
EOS			
area	Noun	object	
=	-	-	
length	Noun	object	
+	-	-	
width	Noun	object	
EOS			
print	Verb	-	???
area	Noun	object	-
EOS			

Figure 3.4(d): Excel File Showing the Structure of the Linguistic Matrix

It uses the semantic nature of the pseudocode’s language to find out the verbs and maps them into actions and the nouns and maps them into variables or mathematical equations to process on the system.

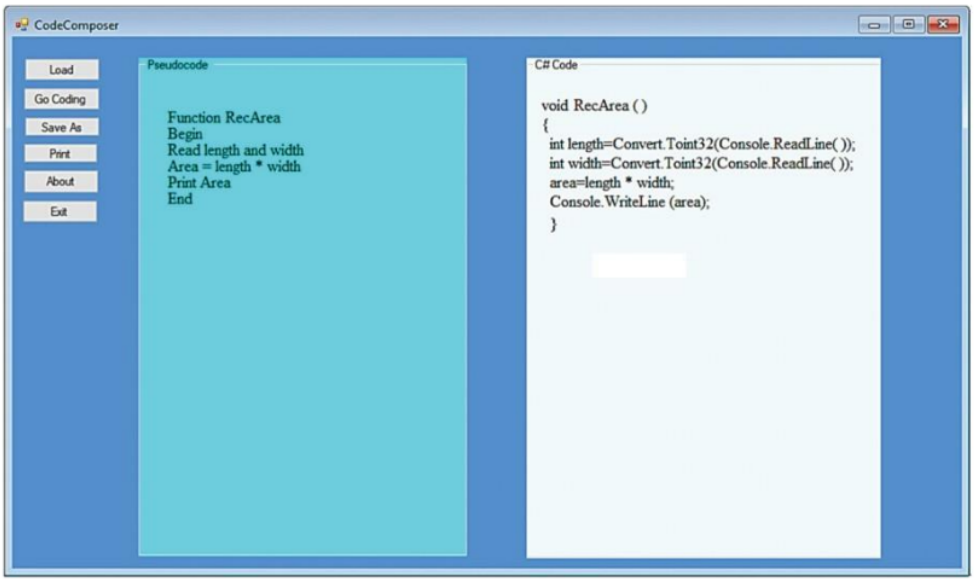


Figure 3.4(e): The Output of Testing a Pseudocode Using CodeComposer

An assessment of the exactness of CodeComposer utilizing a binomial method shows that it has an accuracy of 88%, a review of 91%, and a F-proportion of 89%.

The paper introduces the approach of trying to convert and edit pseudo code to source code in multiple languages by using XML (eXtensible Markup Language) and the features of DOM(document object model) by encapsulating features of each line of code in some tags eg: $< var >, < var - type >, < var - name >, < var - value >$ etc.

DOM is utilized as an article arranged information interface. DOM is an assortment of articles, which characterizes the document structure. What's more, developers can peruse, traverse, change, add and erase on XML archives through the control of these items. DOM can guarantee the right employments of punctuation and design, and improve on the procedure on the record, and furnish a decent blend with the information base change. DOM can make a tree of hubs (alluded to nodetree) in light of the data of XML archives. Also, we can utilize these interfaces and the tree design to get to the data of these XML records.

```
<?xml version="1.0" encoding= "gb2312 "?>
 <code>
  <var>
      <varname> temp</varname>
      <vartype>int</vartype>
  </var>
  <array >
       <arrayname> str</arrayname>
       <arraytype>String</arraytype>
      <arraysize>4</arrzysize>
  </array>
 </code>
```

Expressed by DOM, as shown in Figure 1:

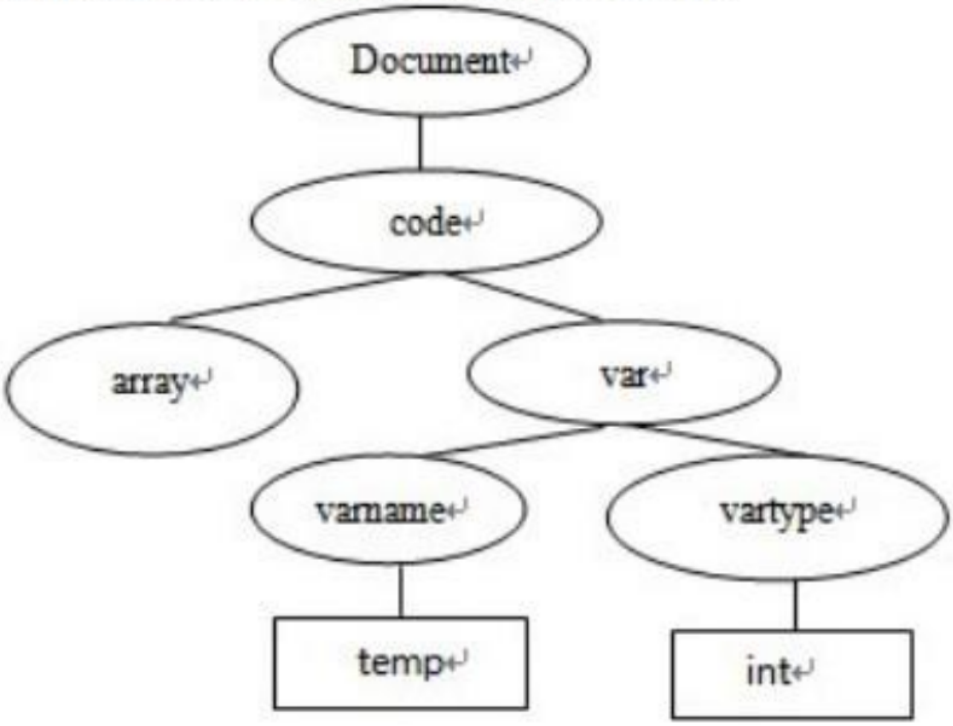


Figure 3.5(f): Description of XML document by DOM model

This model has the benefit of a systematic approach and the use of this converted document to be written in any language the user wishes and not just c code. The author mentions the use of *< cyc >*, *< while >*, *< for >*, *< do >* and various tags for loops, and for blocks where even nesting is important, it introduces the <nesting> tag which can encapsulate various statements.

We are surrounded by a world dependent on AI-based speech assistants which are built to make many trivial tasks in our lives easy. Even these voice assistants are implemented through millions of lines of code.

According to the new education policy, programming has been introduced as a compulsory course for students from junior classes. At the moment, there is very less support provided to help these students understand the programming concepts better. The c code given in textbooks is empirically observed to be not very easy for students to comprehend at the school level when compared to natural language pseudocode.

Editing huge lines of unformatted code can sometimes be frustrating even for a professional programmer. Therefore, our goal is to use a voice-based digital assistant to automate the process of writing and editing code by providing our voice input.

The scope entails providing a framework for students and beginners to get familiarized with the concept of programming and not overwhelmed by the huge lines of code with hard rules. We plan on doing so by retrieving text data from streamed voice input given by user and vice-versa. Once we acquire this text, we convert raw text input to processed natural language formatted output. The natural language formatted output can also be seen as a pseudocode which we then use for the final stage i.e. mapping customised pseudocode to finished c code and well-defined algorithms.

The user can edit and execute the code in a similar fashion and be independent of the rules of typing for the language and focus on the algorithms more. The tool can be helpful for better understanding and can target developers and students at all levels in the future.

Speak Pseudocode2c is a commercial product which will work as a cloud based application. It consists of three phases:-

Retrieving the text from cloud: The user accesses the microphone present at client end. This

input will be streamed and preprocessed at cloud platform (server end) and the user will get the output as text, with minimal errors and maximum accuracy, on the console window.

Formatting text into Tags: The important keyword is picked from received text will be translated into a defined formatted structure which will make it easy to map pseudo-code to source code.

Mapping the Tags to source code: The tags will act as an intermediate model between pseudo-code and source code (c program) which helps in mapping of the same.

With the speak pseudocode2c the user only has to speak the basic pseudocode and the well-defined structured and formatted c program will be shown on the display screen.
The user doesn't have to press any key neither for writing the basic c program nor for compiling and executing the program. The only workload on the user is to speak the pseudocode.
Using Google Text-To-Speech API, available on Google Cloud Platform which is a cloud service provider, makes it easy to convert speech to text with high accuracy.

- The user needs a system which has at least a basic text editor and support for gcc compiler and python3.7. Using an IDE is a big plus.
- Microphone: For receiving the speech input the system should have an in-built microphone or external microphone which capture sound in good quality.
- Speakers: After the text is generated from voice input the framework will use Speech-to-Text and ask the user for confirmation whether the text is right or wrong by saying "yes" or "no". For this there is a requirement for good quality speakers.

- The framework can work on 32-bit and 64-bit Linux, Windows, as well as on Mac OS.
- The machine should have a minimum of 4GB RAM and 256 GB secondary memory.
- A good internet requirement is a key requirement for this framework to work.

- GCC compiler need to be installed on the host machine.
- Python3.7 needs to be installed.
- The basic text editor for displaying the program on the screen.
- Google Cloud python library needs to be installed in order to run the framework.

- The user has good internetconnectivity.
- Microphone and Speakers are working fine at the host machine.
- Errors from the cloud service provider.
- The client has all the software requirements installed on the machine.
- If the machine does not have enough hardware resources available for the application then the users might have to allocate them with other applications.

- Failure of microphone/speaker may stop the program.
- Unknown keywords picked up from the speech or unclear speech due to background noise.
- Incomplete or erroneous installation of the gcc compiler on the host machine.
- Different python3 version installed on the host machine.

The possible risk which might occur are: −

- Retrieving voice input from the cloud with subpar accuracies.
- Formatting the text after successfully retrieving it from the cloud.

- Creating a custom natural language-based dictionary from pseudocode for c language.
- Real-time updation of c code in the editor.
- Crashing of program because of different versions of python3 and gcc compiler.

As our goal is to convert spoken pseudocode to text and then map it to code, the main functional requirements include a user interface being a code editor where the user will be able to make changes to the code by using their voice. The program should be updated in real-time and visible to the user in the code editor as the user speaks. This might include some latency in the beginning as we are using Google Cloud Speech engine.
Our goals will include to implement this framework with minimal latency. This will be an interactive framework which confirms with the user whether to make any changes or to compile the final code and act accordingly.

- The framework will be accessible to the user through a software GUI built with native components from the system itself.
- We are planning on using a voice based system so the text interaction will be a choice for the user to opt from. For better readability, we'll go with textbook fonts like Arial, Cambria, Times New Roman.
- The layout will be available currently only for systems and not mobile. The layout will be horizontal with one half of the screen for the user and the other half for the code to be showcased for easy editing.
- The standard functions will include : language, editor, execute program, stop recognizing, licensing and help.
- The relative timing of inputs will depend on the user's voice input and his internet connection to the cloud. The only delay on the code side which will have to be optimised for being sub-second is the conversion of pseudocode to finished c code.

- The programmable functions will be the input microphone, language spoken and to display recognized text or not.
- The error messages will be displayed on the screen as dialog boxes with minimal UI and also be spoken out to the user for enhanced accessibility.

- Microphone : The device types that can be supported are the default installed microphone inputs with properly installed drivers. The microphone will communicate through system permissions and no default protocol is needed.
- Speaker : A default output speaker is required although is optional in the case the user doesn't want any voice output throughout the program. The speaker will communicate through system permissions and no default protocol is needed.

- Name and Description: GUI implemented by Tkinter
- Version / Release Number: 1.0
- Operating Systems: Across all platforms
- Tools and libraries: Tkinter in python

- Name and Description: Google Cloud Speech to Text
- Version / Release Number:
- Operating Systems: Windows 10, MacOS, Linux
- Tools and libraries: Text to Speech and Speech to Text

- Name and Description: gcc compiler
- Version / Release Number: gcc version 9.3.0 or above
- Operating Systems: Windows 10, MacOS, Linux

- Protocol: HTTP/HTTPS
- Client port: <any>
- API port: 80/443
- Allowed methods: GET, POST, PUT, DELETE, UPDATE

This communication between the client and the cloud platform is essential and the only communication for the whole framework. This communication is responsible for the speech to text conversion and the return methods will provide us with the text for further processing.

- The reliability of the framework will be dependent on google cloud and user defined libraries for speech to text conversion.
- It can be accessed by anyone around the world.
- It is an on-demand service, hence requires no prior installations.
- It is operable only if the user has a stable internet connection.
- The tool will be compatible with all operating systems thus making it more flexible to use.

- The framework provides built-in protection for all user files as it is reliable on google cloud.

- No authentication is required to use the product, only licensing of the software.
- Confidentiality and data integrity is provided as per google cloud services.
- A user's files cannot be accessed by other users accessing the tool.

This document is the high level design document for the “Speak Pseudocode2c”. The purpose of this High Level Design (HLD) Document is to add the necessary detail to the current project description to represent a suitable model for coding. This document is also intended to help detect contradictions prior to coding, and can be used as a reference manual for how the modules interact at a high level.

The HLD documentation presents the structure of the framework, such as the master class diagram, system architecture, High Level Design, State Diagram (flow) and deployment model. The HLD uses non-technical to mildly-technical terms which should be understandable to the administrators of the system. The goal is to display the code on screen via. speaking. The abstract is that the users will speak the natural language pseudocode in the microphone and corresponding to that pseudocode it’s respective c language code will be generated. Mapping pseudocode to source code will be done automatically via the framework.

This document focuses on the general principles of design and building the framework for speak Pseudocode2c systems and explain the high level concepts of different layers in the framework. This will introduce the methodology used for the evaluation and further refinement of concepts, as well as acts as a guidance for the process of identifying the necessary technologies required and used.

According to the new education policy by the government, coding is now mandatory at school level, the support for it is negligible. Speak Pseudocode2c framework will boost and enhance the coding experience for the students. The design concepts for Speak Pseudocode2c framework will have to take into account aspects related to installation, updation and maintenance. The

framework uses the Cloud services for processing the voice input. The advantages of it is as follows: −

- Processing Speed: The client machine has minimal processing to do because most of the processing will be done on the cloud server. The voice input will be streamed as it will be broken into chunks of data.
- Accuracy: Using the Cloud API such as Speech-To-Text provides a large corpus of words as well as support for different languages, which increases our output space for the words. As Google cloud Speech process engine is very superior to other cloud vendors, therefore the accuracy is very high (more than 90%).
- Security/Privacy: As the voice input is sent to the Google Cloud via. a json format key, which is unique for every user provided the google service account, hence providing security is again the responsibility of the Google cloud.
- Availability: Google cloud is known for the it's 24x7 availability. The framework just needs an active internet connection to connect to the cloud for sending and receiving the data in the required format.

These all qualities indicate and cover all the basic requirements which are necessary for any product that is launched in the market. Therefore, our approach for choosing the cloud to convert Speech-To-Text is better and more reliable than other methods in building Speak Pseudocode2c framework.

Our team tried different methods to convert Speech-To-Text which are as follows: Offline Mode:

- Neural engine
- Mozilla deep speech
- CMU Sphinx

The accuracy of converting Speech-To-Text offline was very low (approx. less than 50%). Most of the words aren't even getting recognized and processing time was high as all the processing is happening on the client machine. Accuracy is the primary why we opted for the cloud approach.
Online Mode(Cloud):

- IBM Watson
- Amazon AWS
- Google Cloud
- Google Speech-to-Text (without cloud's API)

The low results of offline Speech-To-Text made us look for other options. Using cloud API for achieving the goal solved our problem in hand but different cloud vendors have different accuracy, therefore we compared different API as mentioned above. It turns out Google Cloud provides the most accuracy and speed for converting Speech-To-Text. The comparison is shown in Fig 1. For us the accuracy for Google Cloud Speech-To-Text was 93% and word error rate was less than 10%.
Therefore, using Google Speech-To-Text seems to be a good option and it's advantages are already described.

Comparison Table

	Amazon Transcribe	Google Cloud Speech-to-Text	IBM Watson Speech-to-Text	Nuance Dragon NaturallySpeaking
Languages	2	120	7	43
Multiple Speaker Identification	✓	✓	✓	✓
Audio Sample Rate	8kHz to 48 Kh	16 kHz	16 kHz	16 kHz
Custom Vocabulary	✓	✓	✓	✓
Timestamps	✓	✓	✓	✓
Confidence Score	✓	✓	✓	✓
Profanity Filter	✓	✓	✓	✓
Price	$0.00056 per second	$0.006 per 15 seconds	$0.02 per minute	From $150 lifetime deal

Figure 5.2(a): Comparison of Different Cloud Vendor

- Currently, we are using the Google Cloud free tier which restricts us from using their service for any commercial applications. We have to abide by their terms of service for the same.
- Google Cloud free tier has a limitation for Speech-to-Text API which we are using in the development phase of the application, it is free for 60 minutes/month afterwards it will use the credits. Compared to other cloud vendors' Speech-To-Text Google provides very less free availability of this API.
- The hardware requirements for the system are minimal, although there is a requirement for a microphone and speaker with a working internet connection.

- The user is familiar with the pseudocode of at least one programming language.
- User has a working internet connection.
- The host machine has a gcc compiler installed for execution of the c programs along with python3 and google cloud python library which is necessary for running the framework.
- There is no problem with User Microphone and Speakers.
- The Background noise is minimum where the client is using the application.

- Logical user groups : Students in preliminary classes with coding as a subject, beginners in the world of coding, developers.
- Application component : Google Cloud API, Google Speech to Text API, Google Text to Speech API, Formatting identified text into pseudocode, mapping pseudocode to executable c code
- Data Component : Mapping dictionaries, verb corpus for NLP processing

- Interfacing systems : Google Cloud API and mapping algorithm with mainframe integration and UI.

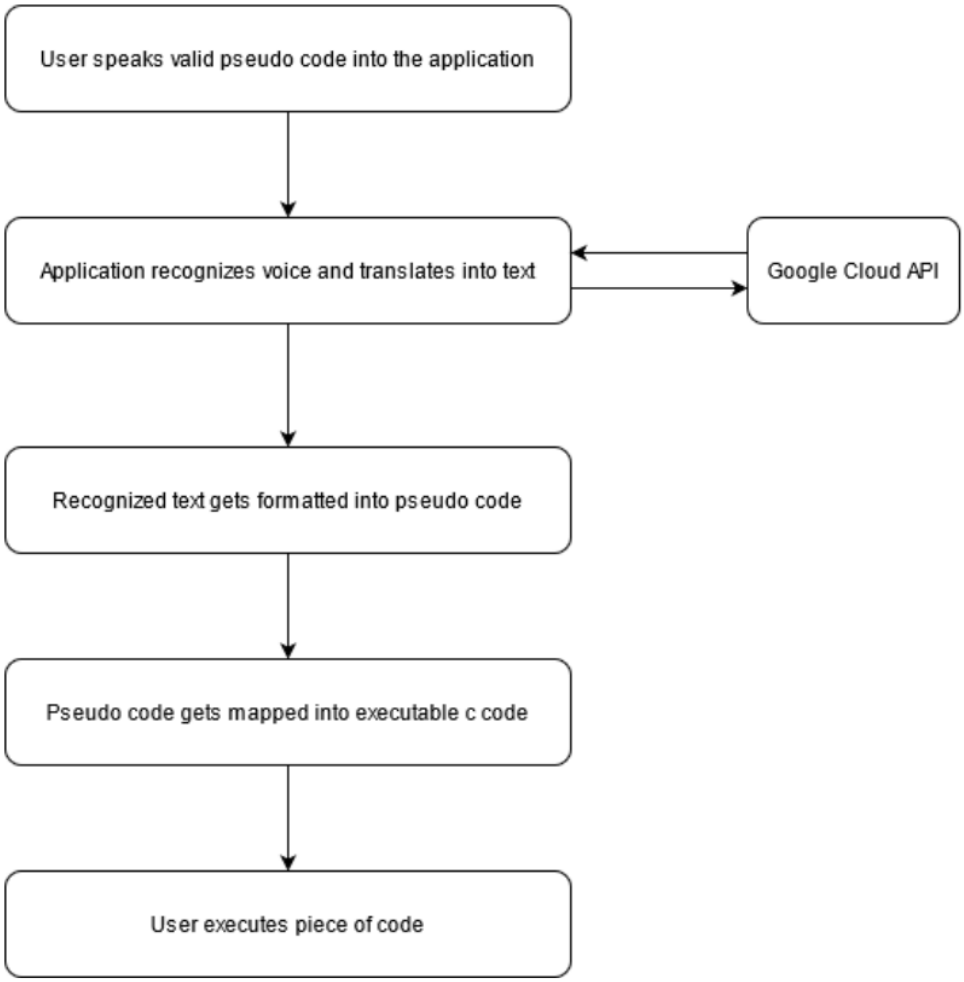


Figure 5.3(b): Flow Chart for High Level Design

The main classes in the class diagrams include client/user, Google Cloud Platform, the terminal on which the program is run and the GUI of the program.

Initially the client has to register using their Google account to the framework. The client then starts speaking to the program. They speak the general pseudocode. The spoken pseudocode is sent to the Google Cloud Platform which returns the formatted pseudocode text. This is done

through API call along with many optimizations for voice to text conversion. The pseudocode text is returned to the terminal on which the program is run. Further, the pseudocode text is mapped to the C code. Finally, a GUI is created which contains the pseudocode and the mapped C code as output.

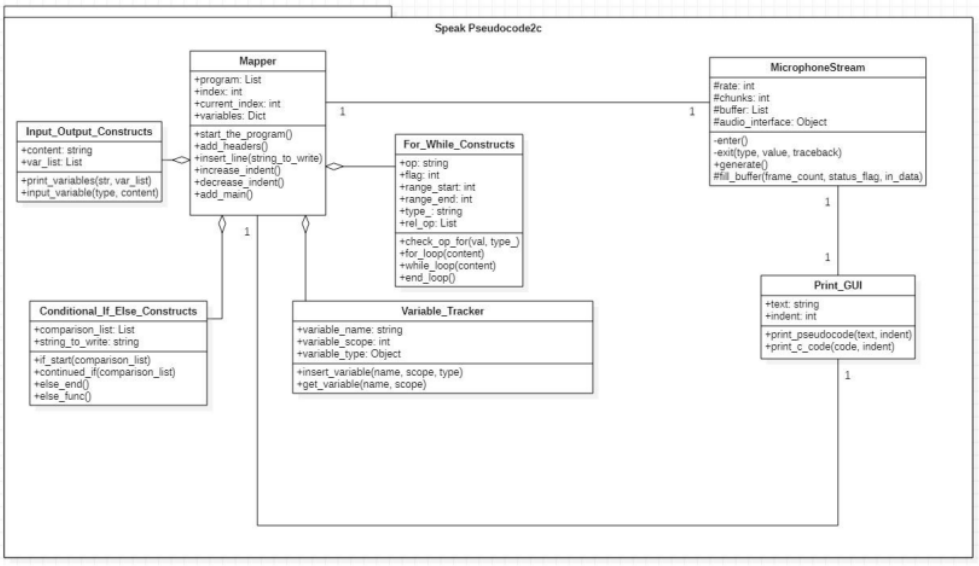


Figure 5.3(c): Master Class Diagram

This project has three main components:

1. Speech to Text using Google Cloud API
2. Mapping Pseudocode to C code
3. Text to Speech Above three components are reused in this project. These components can also be reused in future work or integrated in other projects.

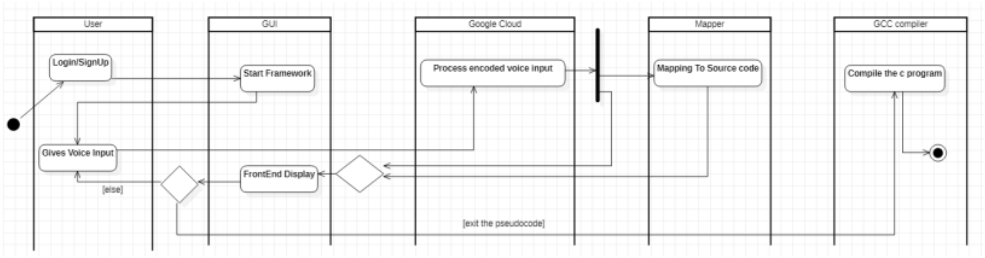


Figure 5.3(d): Activity Diagram

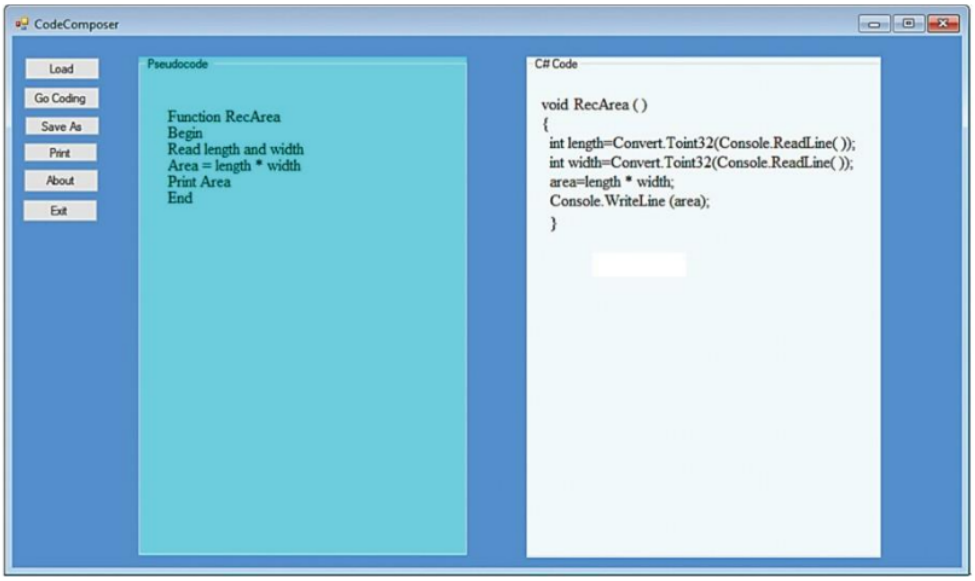


Figure 5.3(e): Activity Diagram

The graphical user interface will have two text widgets in 30% to 70% ratio. The left side being 30% of the screen containing the pseudocode spoken to the program and the right side being 70% will contain the C code. These widgets will be updating their contents in real time as the

user speaks. The goal is to reduce the latency to minimum possibility and increase the accuracy of pseudocode spoken.

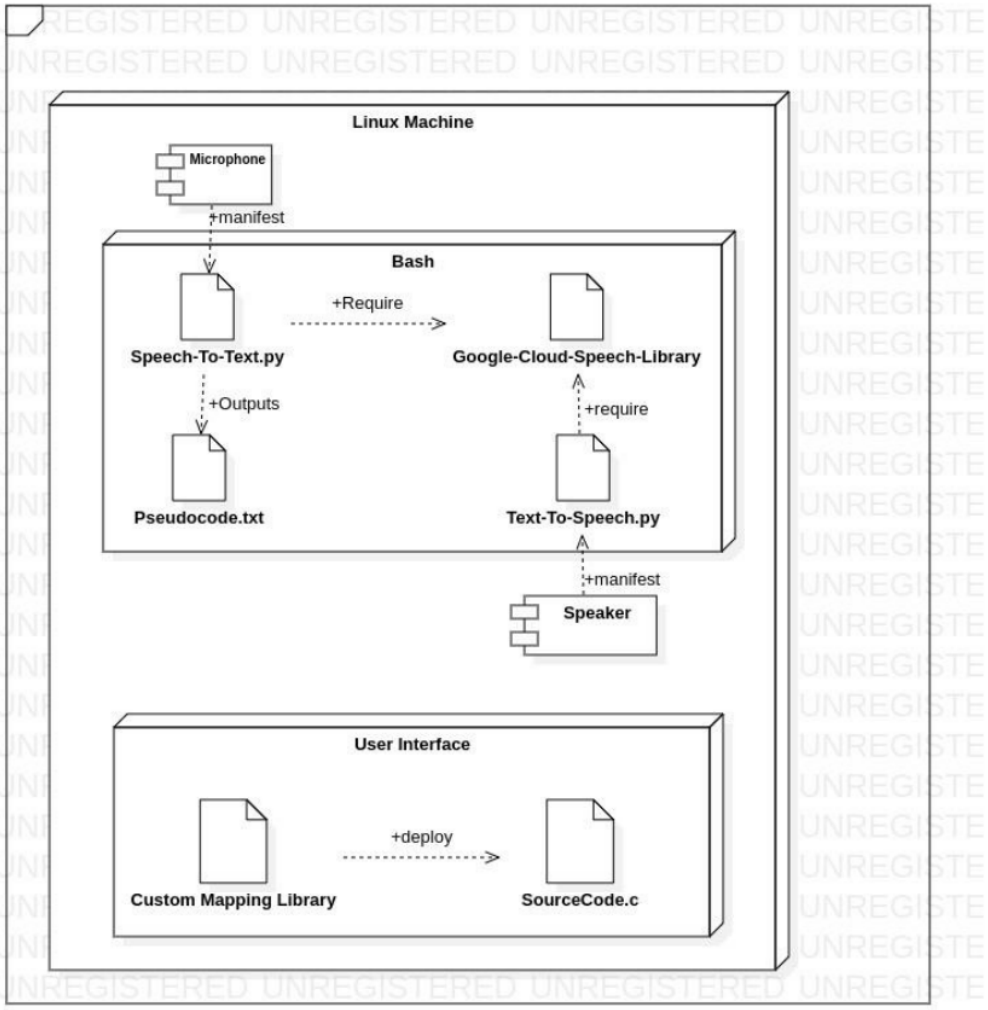


Figure 5.3(f): Deployment Diagram

The aid planned for the system currently is the feedback provided by the system in two modes :

- Text

- Speech

We plan on providing both options for the user to configure the application and use whichever is convenient for them. The text feedback will be provided on the same pane where the user speaks and the system identifies the same. The speech feedback will just be the translation to voice for the same text. The speech option will be togglable. This help option is supposed to provide the user the correct or accurate way to use the application without any errors happening in the application. These may be feedback for the name of the function input or as simple as asking for the type of variable the user is declaring.

Also, there will be a simple documentation on basics of pseudocode which will be defined by our system but not strict by the word. The user can learn basic pseudocode language to use the application with the utmost ease and this documentation will be in the Help section of the menu bar so the user can easily spot it.

A simple user manual with all the features of the application and how to use them will be given as a tutorial in the beginning of a new registration/installation so that the user doesn't feel alien to the interface.

The system is novel as it uses user speech and directly translates it into machine translatable code which can make the process easier and faster. Current research defines many systems with the ability to convert pseudocode to machine code with a certain degree of accuracy. We intend on increasing the accuracy with the added benefit of speech so that the feature is actually usable for the user with ease.

There is no commercial application which tends to do what we plan on achieving with this product. The innovativeness depends on our product mapping the pseudocode to c code with high accuracy of speech and ignoring meaningless commands.

Our product highly depends on the ability to use the information provided to us by the user. The speech to text engine will be highly dependent on this as well as the pseudocode to c code engine will be dependent.
The performance of the system is extremely crucial as the user will be preferring it because of the fast performance of the system and for this the speech to text engine which will be extrapolated from Google Cloud Engine will have to be extremely efficient for it to function properly.

Security will play a vital role in the system as the credentials of the GCP engine should not be publicly released which can result in monetary damages to the developers. Also, the code by the user should in no way be released to the public without the permission of the user as that could result in copy and infringement of original work which we in no way intend.

The system needs to be very reliable in terms of the converted code and the speech being accurately translated by the GCP engine. If the code converted is not interpreted in the correct way it could ruin the whole workflow of the student/developer.

5.3.1.2 Maintainability

We need to ensure maintainability of the system as by the coming day new algorithms and bugs will be addressed on the system which will have the need to be constantly updated on the product so that the user can ensure a smooth experience while coding their application.

Portability not being crucial will be inbuilt into the system as it depends on the cloud and has its own interface in the places needed ensuring the user having a similar experience on any platform where the software is installed.

From the start of building the application we have kept a modern approach integrating cloud and UI components within the application so that the need of legacy to modernization in the future remains minimal and easy for the developers to integrate.

Everything built into the application should be built as a component providing high cohesion and low coupling so that reusability of the components in various places is maximum.

As discussed previously also, the application compatibility is crucial and built into the application as use of cloud and its own UI so that it is usable on any machine having python installed with certain modules.

The utilization maximization and efficiency is crucial to the system and we are constantly trying to optimize the workflow so that we can achieve the maximum results with minimal computation both by the user and developer end to reduce costs and efforts.

This document focuses on the general principles of design and building the framework for Speak Pseudocode2c systems and explains the low-level concepts of different layers in the framework. This will introduce the methodology used for the evaluation and further refinement of concepts, as well as act as a guide for the implementation of the modules in reference.

This document is the low-level design document for the “Speak Pseudocode2c”. The purpose of this Low-Level Design (LLD) Document is to add the necessary detail to the current project description to represent a suitable model for coding. This document is also intended to help detect implementation specifics and can be used as a reference manual for how the modules interact at a low level.

The LLD documentation presents the structure of the framework, such as the master class diagram and Design Description. The LLD uses technical terms which should be understandable to the administrators of the system. The goal is to display the code on screen via speaking. The abstract is that the users will speak the natural language pseudocode in the microphone and corresponding to that pseudocode its respective c language code will be generated. Mapping pseudocode to source code will be done automatically via the framework.

1. Currently, we are using the Google Cloud free tier which restricts us from using their service for any commercial applications. We have to abide by their terms of service for the same.

2. Google Cloud free tier has a limitation for Speech-to-Text API which we are using in the development phase of the application, it is free for 60 minutes/month afterward it will use the credits. Compared to other cloud vendors' Speech-To-Text Google provides very little free availability of this API.
3. The hardware requirements for the system are minimal, although there is a requirement for a microphone and speaker with a working internet connection.

4. The user is familiar with the pseudo code format.
5. The user has a working internet connection.
6. The host machine has a GCC compiler installed for the execution of the c programs along with python3.
7. and google cloud python library which is necessary for running the framework.
8. There is no problem with the User Microphone and Speakers.
9. The Background noise is minimum where the client is using the application.

We have decided to follow the function-oriented design approach where the system is comprised of many smaller sub-systems known as functions. These functions are capable of performing significant tasks in the system. The system is considered as the top view of all functions.
Function-oriented design inherits some properties of structured design where divide and conquer methodology is used.
This design mechanism divides the whole system into smaller functions, which provides means of abstraction by concealing the information and their operation. These functional modules can share information among themselves by means of information passing and using information available globally.

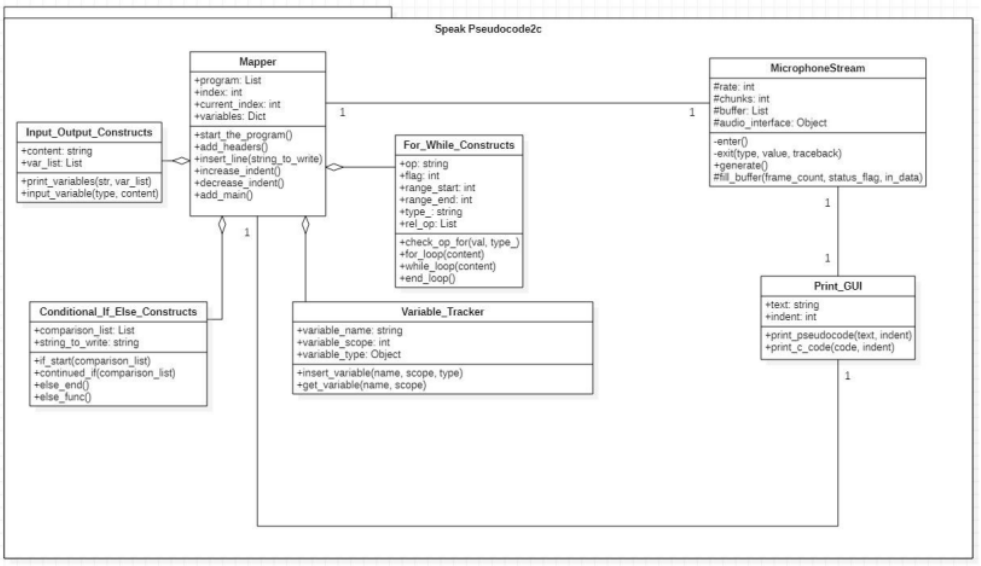


Figure 6.3(a): Master Class Diagram

This module takes in the voice input from the user and sends it to the Google Cloud Server with valid credentials. First off, Google Cloud validates those credentials using a JSON Key and then the voice input is converted into a list of valid outputs. Then from the list of potential outputs, we chose the one which has the highest confidence according to the Google training model and then we process that output further.

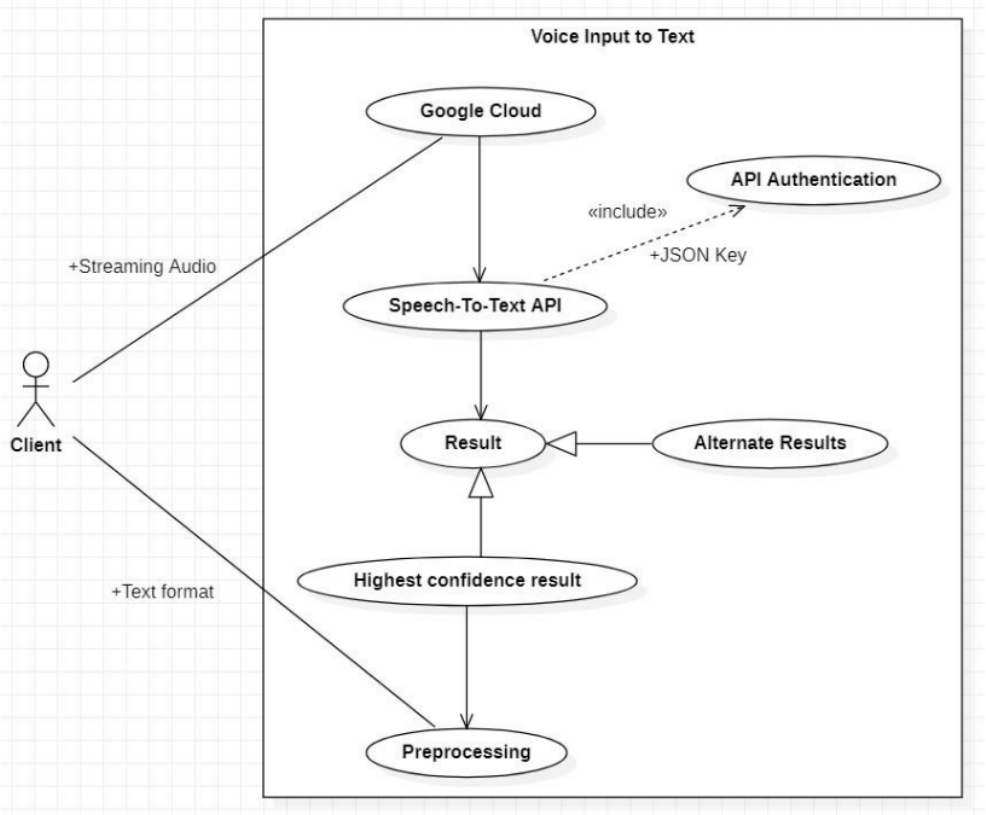


Figure 6.3(b): Use Case Diagram

This class is responsible for mapping generated text to c code. It consists of methods that have mapping for constructs like input and output, for loop, while loop, and if-else.

Data Name	Data Type	Access Modifiers	Initial Value	Description
program	List	private	NULL	To store the converted source code
index	int	private	0	Used in insert_line
current_indent	int	private	0	To keep track of indent
variable_obj	Variable() class	public	Variable()	To keep track of variable
nlp_obj	NLP() class	public	NLP()	To support NLP

Table 6.3(b): Data Members in Mapper class

6.3.4.1 start_the_program

- **Purpose** - call functions add_headers and add_main.
- **Parameter** - mapper object

6.3.4.2 add_headers

- **Purpose** - insert the initial headers required.
- **Parameter** - mapper object
- **Input** - mapper object
- **Output** - add generated code to mapper object.

6.3.4.3 add_main

- **Purpose** - add the main function for the program.
- **Parameter** - mapper object
- **Input** - mapper object
- **Output** - add generated code to mapper object.

6.3.4.4 declare_variable

- **Purpose** - provides mapping for declaring a variable for pseudocode format - declare <variable name> <variable type> .
- **Parameter** - mapper object, generated text from speech input.

- **Input** - generated text from speech input.
- **Output** - add generated code to mapper object.

6.3.4.5 initialize_variable

- **Purpose** - provides mapping for initializing a variable for pseudocode format -initialize <variable name> = <variable value> .
- **Parameter** - mapper object, generated text from speech input.
- **Input** - generated text from speech input.
- **Output** - add generated code to mapper object.
- **Exception** - ValueError

6.3.4.6 input_variable

- **Purpose** - provides mapping to input a variable for pseudocode formats -
    1. input <variable name> <variable type>
    2. input <variable names> <variable types>
- **Parameter** - mapper object, list of generated text from speech input.
- **Input** - list of generated text from speech input.
- **Output** - add generated code to mapper object.

6.3.4.7 assign_variable

- **Purpose** - provides mapping for assigning a variable for pseudocode format - <variable result> = <variable 1> <operator> <variable 2>
- **Parameter** - mapper object, list of generated text from speech input.
- **Input** - list of generated text from speech input.
- **Output** - add generated code to mapper object.
- **Exception** - VariableNotDeclared

6.3.4.8 print_variables

- **Purpose** - provides mapping for printing a string or a variable . It handles the following formats -
    1. print variable <variable name>
    2. print <string>
- **Parameter** - mapper object, list of generated text from speech input.
- **Input** - list of generated text from speech input.
- **Output** - add generated code to mapper object.

6.3.4.9 continued_if

- **Purpose** - provides mapping for normal and nested if-else statements . It handles the following formats -
    1. if <variable1> <operator> <variable2>
    2. else if <variable1> <operator> <variable2>
    3. else
- **Parameter** - mapper object, list of generated text from speech input.
- **Input** - list of generated text from speech input.
- **Output** - add generated code to mapper object.

6.3.4.10 while_loop

- **Purpose** - provides mapping for while statements . It handles the following formats -
    1. while <variable>
    2. while <variable> <operator> <variable>
- **Parameter** - mapper object, list of generated text from speech input.
- **Input** - list of generated text from speech input.
- **Output** - add generated code to mapper object.

- **Exceptions** - VariableNotDeclared

6.3.4.11 for_loop

- **Purpose** - provides mapping of all for statements . It handles the following formats -
    1. for iterator [anything] (optional start_point) till end_point(char or int) (optional increment/decrement by int).
    2. for iterator in range from alphanumeric till alphanumeric increment by integer.
    3. for iterator in range alphanumeric till alphanumeric.
    4. for iterator in range till alphanumeric.
- **Parameter** - mapper object, list of generated text from speech input.
- **Input** - list of generated text from speech input.
- **Output** - add generated code to mapper object.
- **Exceptions** - VariableNotDeclared

6.3.4.12 end_func

- **Purpose** - To handle ending of constructs like while, for and if.
- **Parameter** - mapper object
- **Input** - mapper object
- **Output** - add closing braces to mapper object.

6.3.4.13 get_program_list

- **Purpose** - Return the contents of the program in mapper object.
- **Parameter** - mapper object
- **Input** - mapper object
- **Output** - return program

6.3.4.14 comment

- **Purpose** - Enable user narration by commenting lines which are not intended to be part of the code.
- **Parameter** - mapper object, list of generated text from speech input.
- **Input** - list of generated text from speech input.
- **Output** - add generated code to mapper object.

6.3.4.15 break_stmt

- **Purpose** - Insert break statement wherever required.
- **Parameter** - mapper object.
- **Input** - mapper object.
- **Output** - add break statement to mapper object.

6.3.4.16 continue_stmt

- **Purpose** - Insert continue statement wherever required.
- **Parameter** - mapper object.
- **Input** - mapper object.
- **Output** - add continue statement to mapper object.

6.3.4.17 process_input

- **Purpose** - process the speech input and send it to the appropriate function for further conversion.
- **Parameter** - mapper object, speech input in string format.
- **Input** - mapper object, speech input in string format.
- **Output** - return program from mapper object.

This class is used to implement the graphical user interface (gui) for the framework. The class contains two vertical split text boxes which run parallely with the help of threads. The left text box is used to interact with Google Speech to text, it takes speech input customized pseudocode and converts it into text. The right text box displays the c language source code. It takes the pseudocode in text format and passes it to the Mapper class.

Data Name	Data Type	Access Modifiers	Initial Value	Description
path	Path() class	public	os.getcwd()	Get current working dir
alive	bool	public	True	Check if GUI is running
is_save	int	public	0	To check if .c is saved
font14	Tuple	private	(Times New Roman)	Set the font in GUI

Table 6.3(c): Data members in Pseudocode2c class

6.3.5.1 callback

- **Purpose** - This function contains the piece of code which will run after the thread terminates, usually, garbage cleaning.
- **Input** - class object.
- **Output** - Closes the Tkinter.

6.3.5.2 run

- **Purpose** - Thread starts running from this point. The code written under this will be executed first.
- **Input** - class object
- **Output** - Build various widgets (text box, frame, and button) of the framework .

6.3.5.3 save_code

- **Purpose** - When the save button is clicked then it executes the code under this function. It stores the text content in the right text box and writes it into the .c file.
- **Input** - class object.
- **Output** - Output - .c source program file.

6.3.5.4 compile_program

- **Purpose** - When the compile button is clicked then it executes the code under present in the right text box, but first the text needs to be stored in the .c file.
- **Input** - class object.
- **Output** - Run .c program and display result on the terminal.

6.3.5.5 remove_junk

- **Purpose** - When the undo button is clicked then it deletes the last number of lines written in the right text box i.e the conversion of pseudocode to source code for a particular line.
- **Input** - class object and count of lines written.
- **Output** - deletes the previously written lines.

6.3.5.6 exit_code

- **Purpose** - When the exit button is clicked then it executes the code under this function. It destroys all the widget created by the run function.
- **Input** - class object.
- **Output** - destroy all the widgets.

6.3.5.7 insert_lhs

- **Purpose** - It listens to the output of Google speech-To-Text and writes the text format pseudocode in the text box.
- **Input** - class object and text to be written in the left text box.

- **Output** - writes the pseudocode in left text box.

6.3.5.8 insert_rhs

- **Purpose** - It passes the output of Google speech-To-Text to the Mapper class which returns the c language source code and writes the source code in the right text box.
- **Input** - class object and text to be written in the right text box.
- **Output** - writes the c source code in right text box.

6.3.5.9 show_alert

- **Purpose** - On occurrence of any error or exception, it prompts an alert box showing the type of exception occurred and alerts users about the mistake committed.
- **Input** - class object and text to be written in the alert box.
- **Output** - Prompt the alert box if any error or exception occurs in the c program.

This class opens a recording stream as a generator yielding the audio chunks. It receives the audio data, encodes it and sends it to Google cloud for processing. After processing, it receives the text output from Google Speech-To-Text API and outputs it to the graphical user input (gui).

Data Name	Data Type	Access Modifiers	Initial Value	Description
rate	int	private	16000	Rate at which audio is sent
chunk	int	private	1600	Creating small packets of audio
buff	queue.Queue()	private	Queue()	Store the encoded audio
closed	Boolean	public	False	Check if audio stream is open

Table 6.3(d): Table to show data members in Mapper class.

6.3.6.1 __enter__

- **Purpose** - It creates a thread-safe buffer of audio data and runs the audio stream asynchronously to fill the buffer object. This is necessary so that the input device's buffer doesn't overflow while the calling thread makes network requests, etc.
- **Input** - class object
- **Output** - class object and start listening to the audio and also, starts writing data to the buffer.

6.3.6.2 __exit__

- **Purpose** - Signal the generator to terminate so that the client's streaming_recognize method will not block the process termination.
- **Input** - class object, value and traceback
- **Output** - Closes the audio listener and clear the buffer.

6.3.6.3 _fill_buffer

- **Purpose** - Continuously collect data from the audio stream, into the buffer.
- **Input** - class object, in_data, frame_count, time_info, and status_flag.
- **Output** - Writes data to the buffer.

6.3.6.4 generator

- **Purpose** - Use a blocking get() to ensure there's at least one chunk of data, and stop iteration if the chunk is None, indicating the end of the audio stream.
- **Input** - class object
- **Output** - Yields the data in binary raw format.

6.3.6.5 listen_print_loop

- **Purpose** - Iterates through server responses and prints them. The response passed is a generator that will block until a response is provided by the server. Each response may

contain multiple results, and each result may contain multiple alternatives. Here we print only the transcription for the top alternative of the top result.

- **Input** - responses from Google Speech-To-Text
- **Output** - Prints the customized pseudocode on the gui left text box.

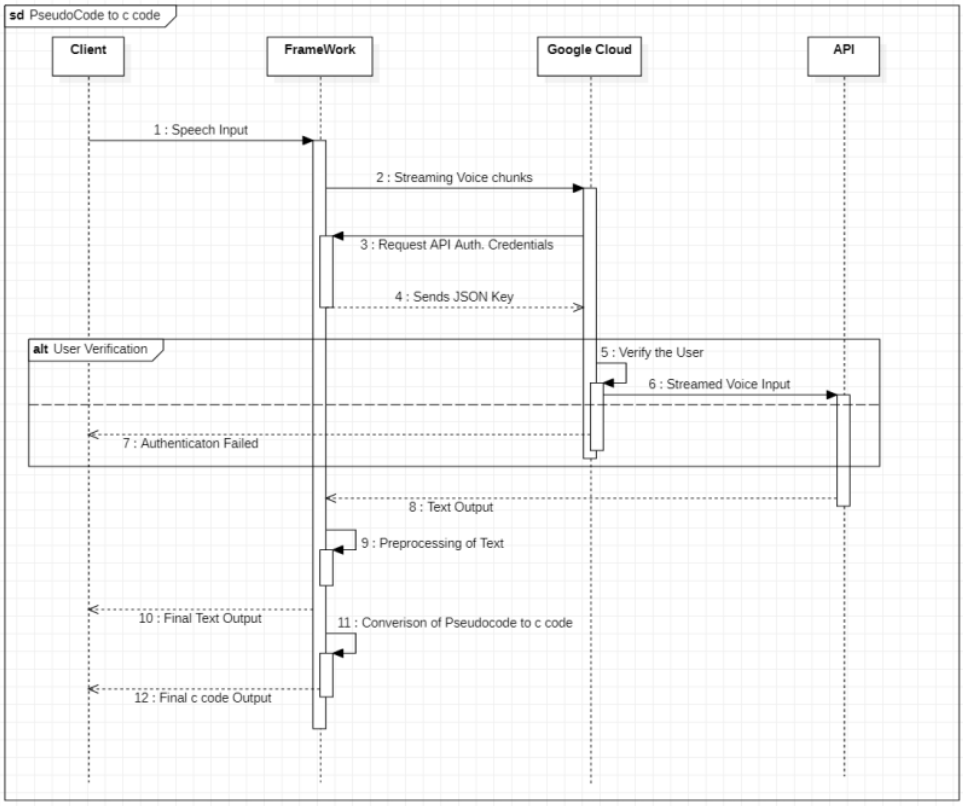


Figure 6.4(c): Sequence Diagram

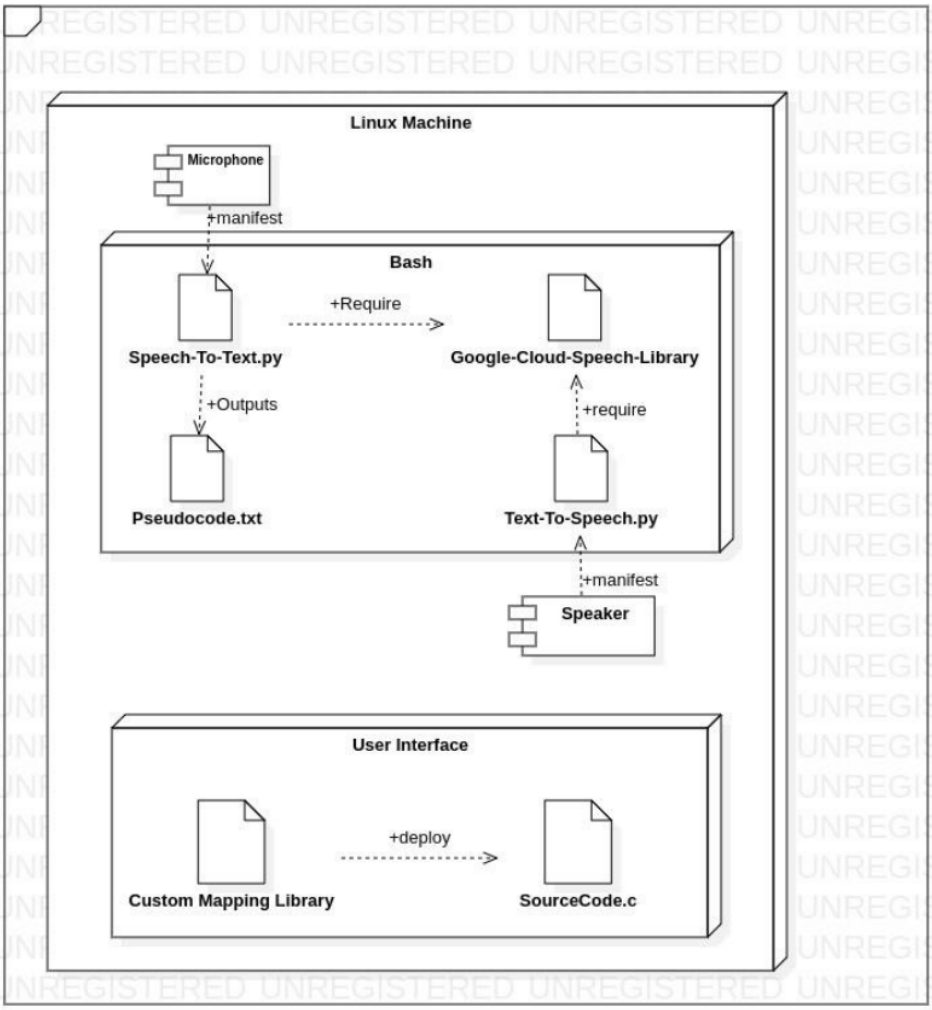


Figure 6.5(d): Deployment Diagram

We first import the google cloud speech to a text library which is used to send the audio data in the form of chunks. The authentication of the user is done automatically when the library is imported as it looks for a JSON file which is generated for a service account and the path to that file needs to be in the environment variable.

```
from google.cloud import speech_v1p1beta1 as speech
```

Since we are sending data in the form of chunks, therefore we need to set the frequency and language of the chunks. en-IN is used to recognize English-India which improves the accuracy of the output. The audio file is encoded with LINEAR16 (16-bit linear pulse-code modulation (PCM) encoding) codec. An audio encoding refers to the manner in which audio data is stored and transmitted. The output needs to be stored somewhere in order to do preprocessing on it. To do that a filename transcript.txt is used and the output is flushed from terminal to that file for further preprocessing.

```
RATE_Of_Frequency = 44000
CHUNK_Date = int(RATE_of_Frequency / 10) #440 language_used = ”en-IN” # language code
file2 = open(”transcript.txt”, ”w”) file2.close()
config =speech.RecognitionConfig(encoding=speech.RecognitionConfig.AudioEncoding.LINEAR16,
↪ sample_rate_hertz=RATE, language_code=language_code,speech_contexts=[{”phrases”:
↪ corr_list, ”boost”: 20.0}])
```

As there is no limit on the duration for the user to speak, therefore the microphone needs to keep on listening. In order to do that we keep the mic open and when it receives the word exit or quit in the content stream, it stops the program automatically. *MicrophoneStream* is a

class which contains modules/functions that acts as a generator for audio chunks by opening a recording stream. The API in this program supports only 1-channel (mono) audio. To fill the buffer object, the audio stream runs asynchronously. This is needed to prevent the input device's buffer from overflowing as the calling thread makes network requests. This is the structure of the *MicrophoneStream* class:

```
class MicrophoneStream(object):
def____init___(self, rate,chunk):
def____enter___(self):
def____exit___(self, type, value, traceback):
def _fill_buffer(self, in_data, frame_count, time_info, status_flags): def
generator(self):

```

To print the data which is received from the server we implemented the function called listen_print_loop which takes the responses as a function argument. The function iterates through server responses and prints them. The responses passed is a generator that will block until a response is provided by the server. Each response may contain multiple results, and each result may contain multiple alternatives. We print only the transcription for the top alternative of the top result. The pseudo-code of listen_print_loop is as follows: procedure listen_print_loop(responses):

```
set variable num_chars_printed to 0
start for loop on responses using iterator as response
if response.results is None //iterate over other responses
    continue
set variable transcript to result.alternatives[0].transcript if
    result is not final
        write the data + "\r" on console flush
        the buffer
    else
        write data on console
        if exit or quit in transcript stop
            the loop
        set variable num_chars_printed to 0


```

Print extra space Display interim results, but with a carriage return at the end of the line, so

subsequent lines will overwrite them. If the previous result was longer than this one, we need to print some extra spaces to overwrite the previous result.

The implementation of listen_print_loop is as follows:

```
def listen_print_loop(responses):
        num_chars_printed = 0 for
        response in responses:
        if not response.results:
                continue
# The ‘results‘ list is consecutive. For streaming, we only care about the first result being considered,
    ‹→ since once it’s ‘is_final‘, it moves on to considering the next utterance. result
        = response.results[0]
        if not result.alternatives: continue
# Display the transcription of the top alternative transcript =
        result.alternatives[0].transcript
# Display interim results, but with a carriage return at the end of the line, so subsequent lines will
    ‹→ overwrite them. If
the previous result was longer than this one, we need to print some extra spaces to overwrite the
    ‹→ previous result
    overwrite_chars= ” ” * (num_chars_printed-len(transcript)) if not
        result.is_final:
                sys.stdout.write(transcript + overwrite_chars + ”\r”)
                sys.stdout.flush()
                num_chars_printed = len(transcript)
        else:
                print(transcript + overwrite_chars)
                if re.search(r”\b(exit|quit)\b”, transcript, re.I): print(”Exiting..”)
                        break num_chars_printed =
                0
```

We wrote our custom defined function to convert the numeric words to digit. For example:

```
one:1, two:2, five thousand six hundred seventy:5670
```

By doing this it will increase the readability of the pseudocode and it will help us in the later stage of the implementation where we have to map our custom pseudocode to the source code

in c language. We have taken this a step further and also converted the basic mathematical symbols in words to actual symbols. For example:

```
Plus : +      Minus : -      Modulo : %      is equal to : =
```

For this task we could have used the python libraries like *wordtodigits* or words2number but there was a problem with this. When speaking a word which can not be converted to a digit (normal english words) then it immediately stops listening to the server and returns an error which we don't want. Further these libraries do not map the mathematical symbols from words to actual symbols and in order to do that along with these libraries, it was increasing the latency which is again a downside. Therefore, using a custom defined function we can fill two needs with one deed.

It keeps track of variable's type and scope throughout the program for easy access to other modules and exceptions like Variable Already Declared and Variable Not Declared scenarios. It is a Singleton Class.

The module receives distinct types of inputs from the user i.e. declare, initialize, input (scanf) and output (printf). Uses an escape keyword for variables in output called "variable" followed by variable name.

Pseudocode:-

```
declare number integer assign
number = 1
print value of number is variable number
```

Output:-

```
int number;
number = 1;
printf("value of number is %d", number);
```

Maps all the if, else, and else if statements by the user to the conditional statement structure in c. Has support for giving multiple relational operators in the condition.

Pseudocode:-

```
if number less than ten print
number less than ten else
print number not less than ten end if
```

Output:-

```
if(number<10)
{
	printf(”number less than 10”);
}
else
{
	printf(”number not less than 10”);
}
```

Takes in multiple inputs from the user for all different types of implementations possible. Can have empty declarations and can handle increment and decrement of iterators easily. It handles declaration of the iterator if needed, or uses an iterator declared in the previous scope. For example:-

Pseudocode:-

```
for i in range from 1 till number increment by 1. end for
```

Output:-

```
for(int i = 1; i <= number; i++)
{
}
```

Maps all the while statements spoken by the user and handles any variable not declared errors by some default declaration of the variables used in the while loop. Also has support for multiple relational operators. Can use the break and continue statements to exit from the loop for some condition. For example:-

Pseudocode:-

```
while i < 10 end
while
```

Output:-

```
int i = 0; while(i
< 10)
{
}

```

The foremost thing which drives our results is the accuracy of the speech-to-text engine which is a third party engine. According to cxtoday [6], Google cloud speech-to-text scored a 79% accuracy in 2020. Keeping that in mind, due to the nature of the stubs approach and static mapping, the accuracy for the framework for converting speech to source code will also be 79%. Therefore, the accuracy of the conversion, given that the user does not speak anything which is alien to the program and the conversion made by the engine is accurate, will be 100%.

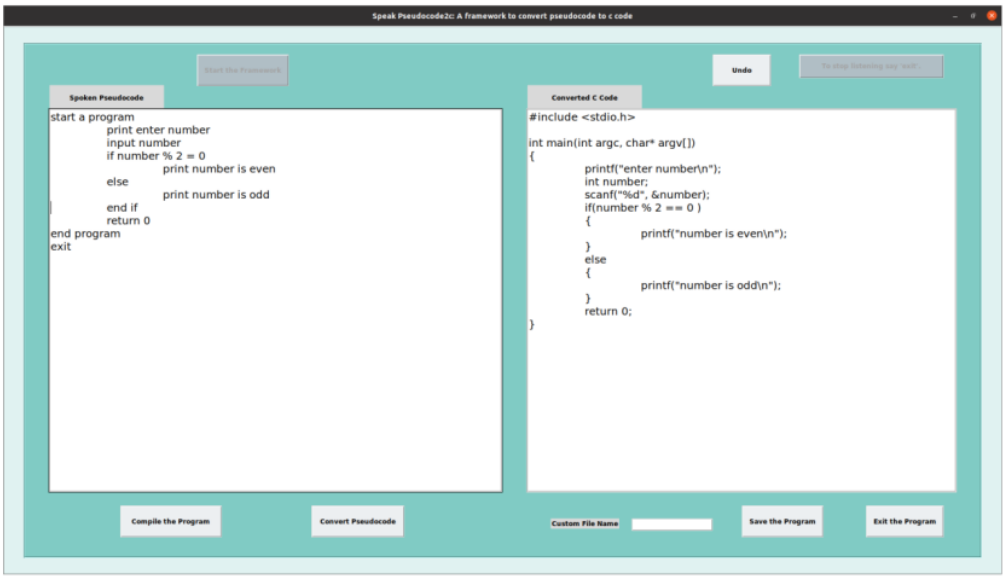


Figure 8.0(a): This is the framework, currently converted for a program to print odd and even numbers. The left side represents the pseudocode spoken and the right side represents the converted source code in c

The framework is currently tested on a set of 30 programs chosen for beginner levels i.e. class 6^{th}, 7^{th}, and 8^{th}, 10 programs each. The accuracy of conversion on the problem set was 100% with no error. This accuracy comes with some assumptions that the user is following a certain set of rules, like giving variables meaningful names which can be spoken and recognised easily. The framework also includes tools for undoing wrong translations made by speech or otherwise, which makes it more robust to errors. Supporting keywords for compiling and saving the program are also included in the GUI and as a verbal command.

- The framework successfully converts natural language pseudocode to formatted c source code with 100% accuracy.
- User can *Compile* and *Save* the generated program through the voice input.
- Added functionality for a narration/commenting system, convert written pseudocode button is there to directly convert the whole pseudocode to c code at one go and an undo button for a robust framework and handling errors in speech.
- The framework is beneficial to use especially for beginners who want to learn the C language. Speak Pseudocode2c can be accessed by the users through a well formed GUI which provides real time line to line updation of voice input.
- The user can save the generated program with any desired file name. Even if no file name is given, it will be saved with a default name and can be accessed by the user later in time if needed.
- When the framework is started, a set of rules are displayed to brief the user about the guidelines and limitations of the framework for smooth usage throughout.

- Functionality can be extended to include support for strings and other data types.
- Presently, the framework supports C language only. The framework can be extended for other languages as well.
- The framework can be modified to make the process completely offline as translating and retrieving data can add latency and doing so will remove the internet dependency as well.
- Functionality which provides text to speech conversion can be added. This could be helpful for the user as they can rectify if speech to text translation is correct or not.

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