A school intercom system that filters inappropriate phrases and plays emails as announcements in the voice of the sender.

Final Report

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Submitted as partial fulfilment of the requirements of Project EPR402
in the Department of Electrical, Electronic and Computer Engineering
University of Pretoria

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Study leader: Mr. A. Oloo

Part 1. Preamble

This report describes the work I did in designing an intercom system that receives emails and plays the text out as speech, using speech theory and digital signal processing.

Project proposal and technical documentation
This main report contains an unaltered copy of the approved Project Proposal (as Part 2 of the report).

Technical documentation appears in Part 4 (Appendix).

All the code that I developed appears as a separate submission on the AMS.

Project history
This project adapted the formant synthesizer developed by Klatt (1980). The properties of speech acoustics were obtained through the use of Praat recording software and Klatt parameters. Software for using an email application programming interface (API) was taken from PythonCode (2022). The databases used for parts of speech (POS) tagging and phoneme transcription was provided by the NLTK database and the CMU institution. The rest of the work reported on here, is entirely my own.

Language editing
This document has been language edited by a knowledgeable person. By submitting this document in its present form, I declare that this is the written material that I wish to be examined on.

My language editor was ___mrs. Noeroenniesa Ramedies___ .

__________________________ __________________________
Language editor signature *Date*

Declaration

I, ___Ishaque Ramedies_____ understand what plagiarism is and have carefully studied the plagiarism policy of the University. I hereby declare that all the work described in this report is my own, except where explicitly indicated otherwise. Although I may have discussed the design and investigation with my study leader, fellow students or consulted various books, articles or the internet, the design/investigative work is my own. I have mastered the design and I have made all the required calculations in my lab book (and/or they are reflected in this report) to authenticate this. I am not presenting a complete solution of someone else.

Wherever I have used information from other sources, I have given credit by proper and complete referencing of the source material so that it can be clearly discerned what is my own work and what was quoted from other sources. I acknowledge that failure to comply with the instructions regarding referencing will regarded as plagiarism. If there is any doubt about the authenticity of my work, I am willing to attend an oral ancillary examination/evaluation about the work.

I certify that the Project Proposal appearing as the Introduction section of the report is a verbatim copy of the approved Project Proposal.

______________________ __________________

I. Ramedies Date

TABLE OF CONTENTS

LIST OF ABBREVIATIONS

LA	Linguistic analysis
NLP	Natural language processing
DSP	Digital signal processing
TTS	Text-to-speech
STT	Speech-to-text
API	Application programming interface
POS	Part(s) of Speech

Part 2. Project definition: approved Project Proposal

This section contains the problem identification in the form of the complete approved Project Proposal, unaltered from the final approved version that appears on the AMS.

For use by the Project lecturer	Approved		Revision required	
Feedback			Approved	

To be completed by the student						
PROJECT PROPOSAL 2022				Project no	A05	Revision no 2
Title	Surname	Initials	Student no	Study leader (title, initials, surname)		
Mr	**Ramedies**	**I**	16023405	Mr A Oloo		
Project title						
A school intercom system that filters inappropriate phrases and plays emails as announcements in the voice of the sender.						

Language editor name	Language editor signature
N. Ramedies	
Student declaration I understand what plagiarism is and that I have to complete my project on my own.	Study leader declaration This is a clear and unambiguous description of what is required in this project. Approved for submission (Yes/No)
Student signature	Study leader signature and date

1. Project description

What is your project about? What does your system have to do? What is the problem to be solved?

The problem addressed in this project, is the lack of physical contact between teachers and students in schools, due to traveling between classrooms after periods or possibly teacher unavailability. Teachers that make announcements and students that hear them, may not able to be in the same physical location. Employing an individual, to collate announcements would not be an efficient solution to this problem. Thus, the problem addressed in this project is to develop a school intercom system that will make announcements on behalf of the teacher, eliminating the need for announcers or any human intervention.

The concept is to have a system that receives a text-based email from a teacher, then announces this email to the students, in the voice of the teacher that sent the email. This may be accomplished by integrating natural language processing techniques, speech synthesis, and digital signal processing into a single intercom system. The system should also account for any inappropriate phrases received from the email and should be able to play the announcement over an integrated speaker, so that multiple students in a classroom are able to hear it.

2. Technical challenges in this project

Describe the technical challenges that are *beyond* those encountered up to the end of third year and in other final year modules.

2.1 Primary *design* challenges

A main design challenge would be accounting for multiple user's voices. Another is hardware allocation of the system resources. The overall system should be designed to work in real time, so all training of artificial intelligence systems, if used, will have to be conducted, prior to final product compilation. The linguistic analysis technique used will have an effect on the processing time of the overall implementation, and each technique has benefits over others. Selection of a specific technique, or development thereof, will be seen as a design challenge.

2.2 Primary *implementation* challenges

A system implementation challenge, is managing waveforms and assigning it to analyzed text, correctly. Testing results of synthesized audio, when compared to natural language audio, might be difficult to achieve, unless extra functional components are developed. It also might be challenging to develop bandpass filters with very narrow passbands, depending on the support that the development platform provides. Another implementation challenge would be within the natural language processing and linguistics analysis algorithms. The English language contains many rules and it might prove challenging to combine multiple rules without making design errors within the algorithms.

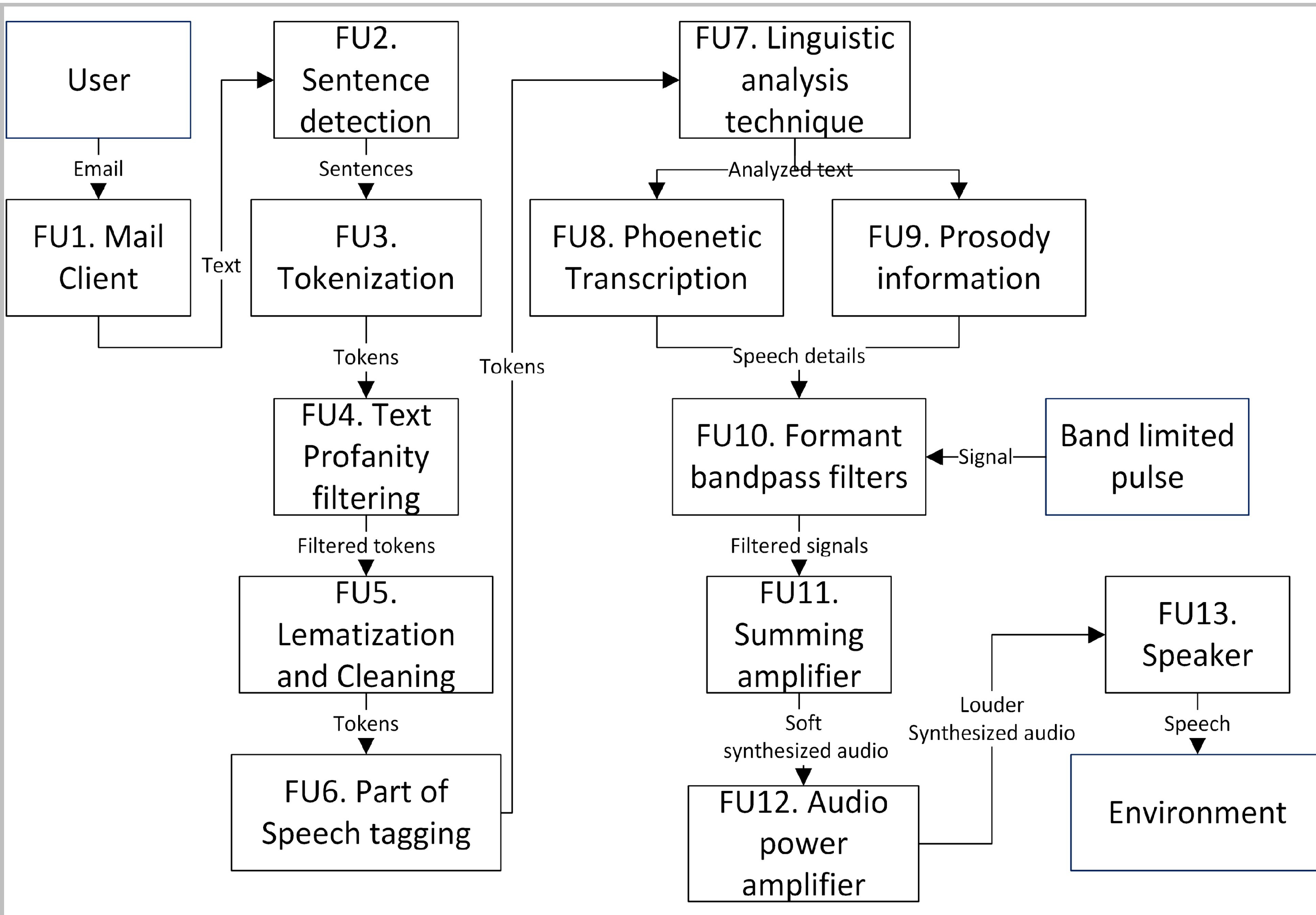
3. Functional analysis

3.1 Functional description

Describe the design in terms of system functions as shown on the functional block diagram in section 3.2. This description should be in *narrative format*.

The user input will be an email text announcement that is sent to the system via the internet. FU1 is a mail client that will receive these emails. The text is extracted from the email whereafter, natural language processing (NLP) and linguistics analysis is performed on the text from FU2 to FU7. The process involves detecting sentences (FU2), and creating tokens from those sentences (FU3). These tokens are run through a profanity filter (FU4) that removes all inappropriate phrases contained within a set of tokens, by checking the tokens against a database of profane words. The output of the profanity filter will be a set of tokens with some marked to be inappropriate, if applicable. The set of marked tokens will then go through a lemmatization and cleaning process, in FU5, where sentences are correctly formatted. Each token is then tagged with its part of speech (FU6), relevant to the whole sentence. A linguistics analysis technique, such as a rule-based technique, is then performed on the output, in FU7, to determine the meaning of the tagged sentences. The analyzed text is sent to FU8, a phonetic transcription function, that will append phonemes to every part of the word, representing what it will sound like. The analyzed text is simultaneously sent to a prosody generator (FU9), which determines the prosody information, i.e., the pitch, loudness, and duration of the words. The information generated from the combination of FU8 and FU9 are then used in FU10 and FU11, that make up a waveform generator or synthesizer. The synthesizer artificially creates a sound signal from the retrieved prosody information using formant synthesis. Formant synthesis combines 5 different frequencies (formants), of a user's voice, utilizing digital bandpass filters (FU10), into a single voice signal, by using a summing amplifier (FU11). Depending on the strength of the synthesized signal, it may be amplified through an audio power amplifier (FU12). The synthesized speech would then be output as an announcement through FU13, a speaker. The output should be heard in the classroom environment by students.

3.2 Functional block diagram



4. System requirements and specifications

These are the core requirements of the system or product (the mission-critical requirements) in table format IN ORDER OF IMPORTANCE. Requirement 1 is the most fundamental requirement.

	Requirement 1: the fundamental functional and performance requirement of your project	Requirement 2	Requirement 3
1. <u>Core mission requirements of the system or product.</u> Focus on requirements that are core to solving the engineering problem. These will reflect the solution to the problem.	The announcement has to be broadcast in a synthetic voice that sounds similar to the sender's voice, in terms of the pitch, loudness, length, and strength.	The system should be able to convert an arbitrary text input to speech information in natural language.	The message that is announced, should be filtered of all explicit inappropriate phrases. The system should consist of a comparative database with which to deal with all profanity.
2. What is the <u>target specification</u> (in *measurable* terms) to be met in order to achieve this requirement?	The fundamental frequencies, amplitude, duration, and amplitude dynamics of the formants in a speaker's voice should be band limited. Formant peaks should not differentiate more than 10% of the original voice.	A linguistics analysis module produces word error rates (WER) of 40% or below, after processing text. Processing for short non-complex sentences comprising of around 15-20 words should provide speech information to the synthesizer within 2 seconds.	The accuracy of a large sample set, should be at 100%, to ensure that the profanity filter is operable for explicit profanity.
3. <u>Motivation</u>: *how* or *why* will meeting the specification given in point 2 above *solve the problem*? (Motivate the *specific* target specification selected)	Adults can differentiate a 1-2% change in 4000Hz tones. At lower voice frequencies reaching maximum ranges of 155Hz - 255Hz, people have less sensitive hearing. A 5-10% change in this metric will yield a similar sounding synthetic voice.	Developed STT transcribers of human voices yield WERs between 15 to 40%. It is expected that synthesized speech will lie within this range. Performance measure is a non-deterministic polynomial (NP) problem, so a cubic bound complexity is assumed.	A 100% profanity filtering margin ensures that evident profanity is removed. This relies on the assumption that teachers are responsible in what they announce.
4. How will you <u>demonstrate at the examination</u> that this requirement (point 1 above) and specification (point 2 above) has been met?	Recordings of multiple announcers' voices will be played, in conjunction with the synthesized speech for auditory comparison. Waveforms of the speech will also be displayed and may be compared.	Text data will be fed into the system, wherein a linguistics analysis function will provide a synthesizer with speech information. A STT transcriber will interpret the perceived announcement after synthesis. WER and execution time will be displayed.	Text data sets, containing profanity, will be input into a profanity filter. The filter will identify explicit profanity and the corresponding specification may be observed.
5. <u>Your own design</u> <u>contribution:</u> what are the aspects that *you will design and implement yourself* to meet the requirement in point 2? If none, *remove this requirement.*	A formant text-to-speech (TTS) synthesizer, consisting of DSP techniques, will be designed and implemented. A prosody generation function will be implemented.	A linguistics analysis module, will be implemented to generate speech information. Natural language processing functionality will be implemented to reduce possibility of error for the linguistics analysis module.	A text-based profanity filter will be developed and implemented.
6. What are the aspects <u>to be taken off the shelf</u> to meet this requirement? If none, indicate "none"	A microcontroller board, capable of digital signal processing, will be purchased. Software to record user audio will be used. A speaker will be used for audio playback of the processed signal.	A microcontroller board with adequate processor speed. A speech-to-text (STT) software platform to evaluate the system will be used. An audio recorder will be used to store system output. A database of English words will be used.	None.

System requirements and specifications page 2			
	Requirement 4	Requirement 5	Requirement 6
1. Core mission requirements of the system or product. Focus on requirements that are core to solving the engineering problem. These will reflect the solution to the problem.	A mail system should receive emails from a user.		
2. What is the target specification (in *measurable* terms) to be met in order to achieve this requirement?	The text data that is sent by the user in the email, should be correctly reflected by the receiving end of the system, with no changes made to the input.		
3. Motivation: *how* or *why* will meeting the specification given in point 2 above *solve the problem*? (Motivate the *specific* target specification selected)	To ensure that the mail system receives emails correctly, it must prohibit the receiving end from making any changes to the user's input.		
4. How will you demonstrate at the examination that this requirement (point 1 above) and specification (point 2 above) has been met?	A display screen should indicate notification upon receiving the email. The output of the text should be displayed and confirmed by the user.		
5. Your own design contribution: what are the aspects that *you will design and implement yourself* to meet the requirement in point 2? If none, *remove this requirement.*	Implementation of an email text display feature.		
6. What are the aspects to be taken off the shelf to meet this requirement? If none, indicate "none"	A Wi-Fi module, capable of Internet connection, and an LCD screen, for debugging and display purposes, will be purchased. An email library will be used for the implementation of the mail client.		

System requirements and specifications page 3			
	Requirement 7	Requirement 8	Requirement 9
1. Core mission requirements of the system or product. Focus on requirements that are core to solving the engineering problem. These will reflect the solution to the problem.			
2. What is the target specification (in *measurable* terms) to be met in order to achieve this requirement?			
3. Motivation: *how* or *why* will meeting the specification given in point 2 above *solve the problem*? (Motivate the *specific* target specification selected)			
4. How will you demonstrate at the examination that this requirement (point 1 above) and specification (point 2 above) has been met?			
5. Your own design contribution: what are the aspects that *you will design and implement yourself* to meet the requirement in point 2? If none, *remove this requirement.*			
6. What are the aspects to be taken off the shelf to meet this requirement? If none, indicate "none"			

System requirements and specifications page 4			
	Requirement 10	Requirement 11	Requirement 12
1. <u>Core mission requirements of the system or product.</u> Focus on requirements that are core to solving the engineering problem. These will reflect the solution to the problem.			
2. What is the <u>target specification (</u>in *measurable* terms) to be met in order to achieve this requirement?			
3. <u>Motivation</u>: *how* or *why* will meeting the specification given in point 2 above *solve the problem*? (Motivate the *specific* target specification selected)			
4. How will you <u>demonstrate</u> at the examination that this requirement (point 1 above) and specification (point 2 above) has been met?			
5. <u>Your own design</u> <u>contribution:</u> what are the aspects that *you will design and implement yourself* to meet the requirement in point 2? If none, *remove this requirement.*			
6. What are the aspects <u>to be</u> taken off the shelf to meet this requirement? If none, indicate "none"			

5. Field conditions

These are the REAL-WORLD CONDITIONS under which your project has to work and has to be demonstrated.

	Field condition 1	Field condition 2	Field condition 3
<u>Field condition requirement.</u> In which field conditions does the system have to operate? Indicate the one, two or three most important field conditions.	The intercom system has to be audible enough for a noisy indoor room environment. It should be operable from any location with internet access.		
<u>Field condition specification.</u> What is the specification (in measurable terms) for this field condition?	A standard noisy environment for a classroom of students ranges between 65-70 dB.		

6. Student tasks

6.1 Design and implementation tasks

List your primary design and implementation tasks in bullet list format (5-10 bullets). <u>These are *not* product requirements, but *your* tasks.</u>

- The intercom system must be designed and implemented.
- An appropriate linguistics analysis technique has to be selected.
- Audio data from multiple people will have to be collected.
- Large sentence text data sets, with profanity, have to be collected and sorted.
- A physical cover for the system will have to be designed and built.
- The system should be integrated onto a PCB board.
- Simulated versions of the functions should be developed for debugging and demonstration purposes.

6.2 New knowledge to be acquired

Describe what the theoretical foundation to the project is, and which new knowledge you will acquire (*beyond* that covered in any other undergraduate modules).

- The student will be required to master theoretical background knowledge on Natural Language Processing and Speech Synthesis.
- The student will be required to acquire knowledge on Linguistics Analysis techniques.
- The student will need to learn how to work with audio data sets.
- The student will be required to learn about rulings of speech in the English language.
- The student will have to learn how various aspects of digital signals affect the auditory output of speech.

Part 3. Main report

1. Literature study

1.1. Background and context of the problem

Speech is the expression of thoughts and feelings by articulate sounds, and it is the most natural and convenient form of communication for a human. Speech technologies are those that are able to recognize, analyse and understand spoken speech. The two main forms of speech technology are speech recognition and speech synthesis technologies, where one form of technology uses input speech to perform text processing and the other produces speech from a text processing module. The project entails the latter, a text-to-speech (TTS) intercom system. TTS systems provide benefits to the population by assisting those with literacy difficulties, and reduced vision, among others. It goes without saying that the design and implementation of such a system will positively affect the people that use it. **Figure 1** illustrates the main components involved in a TTS system.

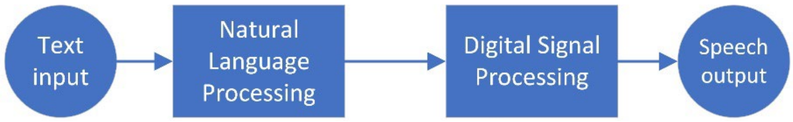


Figure 1 Block diagram of TTS system

To achieve the intercom system, there are a number of subsystems that have to be considered. These main subsystems are the natural language processing (NLP) unit, and speech synthesizer. Depending on the application of the TTS system, it may include a variety of other subsystems, such as a linguistics analysis (LA) module, profanity filter, and spell checker, to name a few. Implementing these subsystems require knowledge of digital signal processing (DSP) techniques, speech processing theory, optimization algorithms, search algorithms, and signal feature extraction. A literature study is conducted to summarise techniques and work related to the intercom system in order to make a decision on its design.

1.1.1. Natural language processing

Natural language processing is a fundamental, key component of text-to-speech synthesis. The front-end development of TTS systems involves NLP, with the objective of attaching prosody or sound information to the input text. Zhigang provides information on how the components involved in the front-end development work together, to provide the necessary output to the DSP unit [1]. The NLP unit consists of a pre-processing block, morphological analysis unit, contextual analysis unit, syntactic analysis unit, phonetization module, and prosody generation function. Reichel describes some generic NLP modules associated with TTS synthesis [2]. The modules are defined as text normalization, part of speech (POS) tagging, grapheme-to-phoneme conversion and word stress.

(a) Pre-processing

Pre-processing of text rectifies any error or abnormalities in text that is received as input. Text such as abbreviations, numbers, acronyms, and dates have to be converted into full text. This process is a form of text normalization, but what it really does is put all input text into a recognizable format for the computer to perform operations on. Pre-processing also breaks text up into sentences, by utilising classifiers to determine where sentences are split up. There are several steps involved in pre-processing, however, the type of system dictates which of the steps are necessary to be implemented.

(b) Morphological analysis and Contextual analysis

The morphological analysis and contextual analysis are usually associated together. This process involves taking the sentences from a pre-processing module and dividing it into a sequence of words. Morphological analysis involves this word segmentation also known as text normalization, and the contextual analysis determines the context of each segmented word or token in relation to the sentence. Whereas classifiers may be used to determine how to split up sentences, for text normalization in the English language, whitespaces often indicate the end of a word.

(c) Syntactic analysis

Syntactic analysis aims at predicting the prosody of the input sentence by parsing it. An important aspect of this module is performing POS tagging. This gives information about the grammar of the sentence by determining each word's part of speech. There are several tagging methods to consider, such as rule-based tagging, stochastic tagging, transformation-based tagging, and hidden Markov model (HMM) tagging [3].

(d) Phonetization

Phonetization involves the generation of phonetic symbols, or phonemes, for each word in the text. This is done by first constructing a lexicon, which is a dictionary of words with their pronunciations. The lexicon would serve as a look up table (LUT) for the words. For words or special characters that cannot be identified as part of the lexicon, it would be necessary to include an algorithm that would be able to generate the phone sequence for that word.

(e) Prosody generation

Prosody generation is the generation of prosodic information for the synthesizer, it would specify signal attributes like pitch curve information, duration, and pause information. This affects the naturalness and intelligibility of the speech output to a certain extent. Generating prosodic information is done if points of emphasis are identified, along with punctuation tokens, and phoneme indices in a lexicon.

1.1.2. Linguistics analysis

NLP consists of computational linguistics, which is the rule-based modelling of human language, with statistical, and machine learning models. In general, linguistics analysis approaches that should be avoided for this system implementation, are; rules-based and neural network implementations, as they are data-driven and may be time consuming. In the book; "*Theory and applications of digital speech processing*" [3], a better insight to linguistics analysis and the desired output is provided. It also provides knowledge on TTS systems and digital signal processing techniques, along with system design considerations. Linguistics analysis may thus be considered as the formatting approach taken for the NLP module.

1.1.3. Speech synthesis methods

Speech generation can be divided into two main categories, namely, rule-driven synthesis techniques, and data-driven synthesis techniques. The concept of rule-driven techniques is to generate speech output according to rules developed by simulating the articulation or acoustic process. Data-driven techniques are more reliant on recorded speech data or parameters derived from speech data. There are several techniques that fall under these categories. Rule driven techniques include the articulatory synthesis and formant synthesis methods, and data-driven techniques consist of the concatenative synthesis, unit selection synthesis, hidden Markov model (HMM) synthesis, and deep neural network (DNN) synthesis methods. These synthesis methods are described for comparative purposes.

(a) Articulatory synthesis

Articulatory synthesis is performed by collecting observations of physical human pronunciation. This synthesis requires rules for an articulatory model to be developed beforehand, so that the parameters of the model may be adjusted to change the synthesized sound. The parameters that are included in the model to be controlled are commonly the tongue tip position and height, tongue position and height, lip aperture, and lip protrusion. Due to the difficulty in simulating human pronunciation through these parameters, this synthesis method often leads to inaccuracies with the quality of sound.

(b) Formant synthesis

Formant synthesis simulates the acoustic process, instead of the physical pronunciation. There is therefore a larger emphasis on the sound of the output. Formants are the main frequencies that distinguish sounds from each other. There are usually 5 formants visible in a spectrum of an audio signal. In 1979, Dennis H. Klatt published a report on formant synthesis containing a realized speech synthesizer, in order to provide a flexible research tool, for studying aspects of speech [4]. The Klatt synthesizer model is based on *Acoustic Theory of Speech Production*, developed by Fant [6].

The paper demonstrates the result of using digital resonators in parallel or cascade configurations, while allowing the user to specify variable control parameter data. The paper also contains synthesizer design descriptions, motivations, computer requirements and strategies for imitating speech utterances. The paper thus provides a good baseline expectancy, when considering a formant synthesizer for speech output. Formant synthesizers have advantages of rule-driven synthesis techniques, and the sound quality of the synthesized speech is better than the articulatory synthesizer but does not sound natural.

(c) Concatenative synthesis

To establish speech that is more natural sounding, concatenative synthesis was developed in the 1990s. Concatenative synthesis is a data-driven synthesis technique, and as the name suggests, it uses various recorded voice units such as words, syllables, phones, and other recordings, and concatenates these units to produce speech. The majority of speech features remain unchanged when comparing it to natural speech, but the technique does rely on the speech corpus that is used to produce the output speech for good quality speech. This approach may be disadvantageous for systems with low memory requirements, and discontinuities around the speech units should be handled correctly for this approach to be stable. DSP techniques such as the pitch synchronous overlap and add (PSOLA) method was developed in 1986 to modify the speech units when concatenating them.

(d) Unit Selection synthesis

The unit selection synthesis is an improvement of concatenation synthesis and was first introduced in 1992 by Sagisaka [7]. In concatenation synthesis, when the speech signals were modified using the PSOLA method, the output speech would often sound more unnatural. Unit selection stores multiple instances of units, and selects the unit that matches prosodic features more accurately, so that less changes are made to the unit for more natural sounding output. The introduction of the model made speech more natural sounding, but still has the same problems of selecting error units and storing a large speech corpus. The approach is also not as flexible since the voice quality could not be improved.

(e) Hidden Markov model (HMM) synthesis

(f) Deep Neural Network Model synthesis

1.2. Application summary

summarises what has been learnt in the literature study and
describes how you applied what you learnt from literature (and perhaps expanded what has been reported on before in the literature).

2. Approach

Your approach to the design, as described in the study guide, goes here.

Commence this section on a new page

3. Design and implementation

3.1 Design summary

Required subheading

This section summarises the project tasks and how they were implemented (see table 1).

Deliverable or task	**Implementation**	**Completion of deliverable or task, and section in the report**
Design of a PCB for the main electronics	The PCB design was completed, using the PCBCAD package. This was done from first principles.	Completed Section 3.2 of the report
Development of optimisation routine	Optimization was completed in Matlab, but the Optimization Toolbox was used, and while some code was developed from first principles, numerical methods for optimization were taken off the shelf.	Incomplete Section 3.3 of the report
DC-DC converters had to be designed and implemented by the student.	The student completed the design and implementation.	Completed Section 4.2.8
The inverter had to be designed and implemented by the student.	The student did not complete the inverter. The design was completed and simulated, but the implementation in hardware did not work correctly.	Incomplete Section 4.2.8

Very important table in this section. The required content is described in the study guide. What

Table 1. Design summary.

This column will correspond to (i) deliverables mentioned in line 5 of the table of section 4 of the Project Proposal,
AND (ii) to the design and implementation tasks in section 6.1 of the project Proposal

Ensure that your table lists all of your technical design tasks. Don't including non-design tasks like “write report”, or “source components”). Please refer to the study guide.

3.2 Theoretical analysis and modelling

…

3.7 Hardware design

Student's own

Super important! The compulsory table mentioned in Appendix 4 of the study guide should appear here.

Commence this section on a new page

4. Results

4.1 Summary of results achieved

Example text appears in the table. Use your own text

Intended outcome	Actual outcome	Location in report
Core mission requirements and specifications		
The motor should be able to rotate at a speed that can propel the vehicle at at least 10 km/h.	The measured maximum speed was 12.6 km/h.	Section 4.2.6
The system should switch between the sources that supply the vehicle depending on energy demand. The battery should supply the vehicle up to 75% of its current rating. When exceeding this rating, the SC should take over the supply.	The switching does take place, and data gathered and stored on the vehicle's on-board memory during a live drive test shows that switching to the SC happened at a current of 2.3 A. The vehicle continued to drive on either source.	Section 4.2.2
The system must be able to determine the placement of each runner in the race with an error of at most 0.05 seconds.	Runners could consistently be placed within 0.1 ms, as determined from photographs.	Section 4.2.7
Bit error rate (BER) should be low. BER should be below 1E-6.	The BER was measured as 10 bit errors in 1000 bits.	Section 4.2.1
Delivered power should be adequate for the load. 2 kW should be delivered to the load.	The system could not deliver the required power into the load. The system could deliver 800 Watts into the load before overheating.	Section 4.2.3
Field condition requirements and specifications		
The system should supply power and actual environmental conditions (sunshine or rain; day or night)	The system was never tested under rainy conditions. The system could not supply power under any conditions other than bright sunlight.	Section 4.2.6
The system should use actual real-time data, corrupted by noise, arriving over a noisy wireless link.	The system could work error-free for at least one hour under these actual field conditions.	Section 4.2.1

Table 2. Summary of results achieved.

This column will correspond to (i) core mission requirements of the product, (ii) the corresponding target specifications, and

4.2 Qualification tests

Qualification test 1: measurement of closed-loop controlled vehicle speed

Objectives of the test or experiment
Equipment used
Test setup and experimental parameters
Steps followed in the test or experiment
Results or measurements
Observations
Statistical analysis

Example text – Use your own text

Qualification test 2: test of switching between battery and supercapacitor

Objectives of the test or experiment
Equipment used
Test setup and experimental parameters
Steps followed in the test or experiment
Results or measurements
Observations
Statistical analysis

Commence this section on a new page

5. Discussion

See the study guide – this section is very important. You need to show that you can stand back and be critical of your own work.

The worst possible thing that you can write here is "everything works perfectly".

There is no perfect design, and you as (aspiring) engineer

5.1 Interpretation of results

These (5.1 tot 5.5) are required headings. You may add more of your own

5.2 Critical evaluation of the design

5.3 Design ergonomics

5.4 Health, safety and environmental impact

5.5 Social and legal impact of the design

Commence this section on a new page

6. Conclusion

6.1 Summary of the work completed

Example text

This report describes work carried out on the design of a voice communication system, with the objective of using very low power to achieve reliable wireless communication over long distances.

A literature survey was completed on modern communication system design. The hardware and software for a low power communication system was then designed from first principles. At the core of the system is an existing DSP board, and all additional hardware were designed and implemented. A Matlab program was developed to simulate the system, as well as C code for the DSP, and assembly language code for a PIC processor that resides on the hardware that was designed. The system was implemented and several field tests were carried out. A voice communications channel was set up between the University and my home in Midrand. The main result is shown in the BER graph in section 3.3 of the report.

6.2 Summary of observations and findings

See example text in study guide

6.3 Contribution

See example text in study guide

6.4 Future work

See example text in study guide

Here you need to be extremely honest about what you achieved, and did not achieve.

Conclusions need to be technical and MAY NOT relate to your personal experience (e.g. "I learnt a lot" would be a good example of what NOT to write)

7. References

[1] Y. Zhigang, “An overview of speech synthesis technology,” in *Eighth International Conference on Instrumentation and Measurement, Computer, Communication and Control*, Beijing, China, 2018.

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Part 4. Appendix: technical documentation

1.1.1.

Commence this main section
on a new page.

1.1.2. HARDWARE part of the project
1.1.3.

Record 1. System block diagram

Record 2. Systems level description of the design

Required headings.
Please see the study guide.

1.1.4. Record 3. Complete circuit diagrams and description

1.1.5. Record 4. Hardware acceptance test procedure

1.1.6. Record 5. User guide

SOFTWARE part of the project

1.1.7. Record 6. Software process flow diagrams

1.1.8. Record 7. Explanation of software modules

1.1.9. Record 8. Complete source code

Required text

Complete code has been submitted separately on the AMS.

1.1.10. Record 9. Software acceptance test procedure

Record 10. Software user guide

EXPERIMENTAL DATA

1.1.11. Record 11. Experimental data