

# SMaTTS: Standard Malay Text to Speech System

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**Abstract**—This paper presents a rule-based text-to-speech (TTS) Synthesis System for Standard Malay, namely SMaTTS. The proposed system using sinusoidal method and some pre-recorded wave files in generating speech for the system. The use of phone database significantly decreases the amount of computer memory space used, thus making the system very light and embeddable. The overall system was comprised of two phases the Natural Language Processing (NLP) that consisted of the high-level processing of text analysis, phonetic analysis, text normalization and morphophonemic module. The module was designed specially for SM to overcome few problems in defining the rules for SM orthography system before it can be passed to the DSP module. The second phase is the Digital Signal Processing (DSP) which operated on the low-level process of the speech waveform generation. A developed an intelligible and adequately natural sounding formant-based speech synthesis system with a light and user-friendly Graphical User Interface (GUI) is introduced. A Standard Malay Language (SM) phoneme set and an inclusive set of phone database have been constructed carefully for this phone-based speech synthesizer. By applying the generative phonology, a comprehensive letter-to-sound (LTS) rules and a pronunciation lexicon have been invented for SMaTTS. As for the evaluation tests, a set of Diagnostic Rhyme Test (DRT) word list was compiled and several experiments have been performed to evaluate the quality of the synthesized speech by analyzing the Mean Opinion Score (MOS) obtained. The overall performance of the system as well as the room for improvements was thoroughly discussed.

**Keywords**—Natural Language Processing, Text-To-Speech (TTS), Diphone, source filter, low-/ high- level synthesis.

## I. INTRODUCTION

**S**PEECH is the act of producing voice via variation of the air pressure that is emitted by the articulatory system (Dutoit, 1997). Whilst, speech synthesizer is the artificial production of human speech where a text-to-speech synthesizer should be able to automatically convert any text into speech by encoding the text into signals carrying linguistic information before it is converted into an acoustic waveform using machine. Major purpose of Text-to-Speech Synthesis Systems is to transform a given linguistic representation, say a chain of phonetic symbols into artificial, machine-generated speech with information on phrasing, intonation and stress by means of an appropriate synthesis method. For a Malay text-to-speech synthesizer, the text written in Malay Language is introduced into the computer by an operator as electronic text. This Malay speech synthesis is an ongoing process, as new and

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existing data sets are continuously accessed for many different experimental speech perception and generation processing tasks. A Text-to-Speech synthesizer would involve grapheme-to-phoneme transcription of input sentences. Graphemes are the letters in a words dictionary listing whilst phoneme is the smallest unit of speech that differentiates one word from another. To convert a grapheme to phoneme, a TTS system would involve the Natural Language module to analyze the text in term of the phonetizer, syntactic analyzer, lemmatizer and prosody generator. The processed sentence would be passed to Digital Signal Processor which generates the corresponding speech signal as shown in the figure above.

Thus, it is easier to divide the module for text-to-speech synthesizing system into two phases, the Natural Language Processing (NLP) module and the Digital Signal Processing (DSP) module. The previous convert written text into readable form with information of the phonetic transcription, intonation and pitch, and the duration of the speech. The latter implies the process of converting the information received from NLP module into natural-sounding speech. A general functional diagram of text-to-speech could be described as in the following figure:

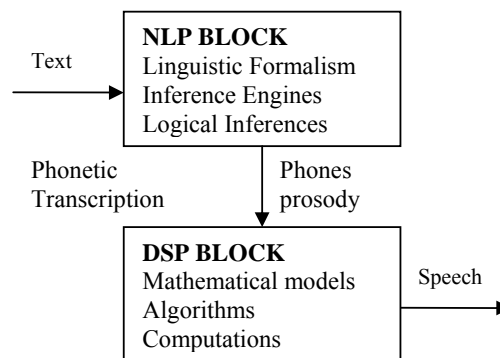


Fig. 1 General Functional Diagram of TTS system

## II. MODERN TECHNIQUES AVAILABLE FOR SPEECH SYNTHESIZER

The choice of the technique generally depends on the language, platform used and the purpose of the system itself. Although it is almost impossible to approximate a human natural speech, it is important to make sure that the synthesized speech is of sufficient quality in such, to ensure an adequate and understandable reading. This section is the survey of the common techniques that are currently used in this field in Digital Signal Processing (DSP) module.

Generally, the methods available are divided into three types as follows:

#### A. Format Synthesis

Rule based or formant synthesis offers freedom in speech production and provides much flexibility in producing even meaningless words compared to the other synthesizer methods. However, the quality of the output speech is low since the sounds are rather synthetic and is under the acceptable level of human hearing.

#### B. Articulatory Synthesis

This synthesis tries to model main articulators and vocal cords in human speech production. Although this method might give the most satisfying natural sounding speech theoretically, it is however the most complex method with the highest computational load due to the fact that human anatomy is very complex and flexible.

#### C. Concatenative Synthesis

The models used in concatenative synthesizers can be classified into two groups: the production models where each physical and acoustical property are modeled by mathematical scheme such as in the Linear Predicting Coding (LPC) model, and the pure DSP models where the signal processing methods applied for synthesis such as in Pitch Synchronous Overlaps and Add (PSOLA) models. Although concatenated TTS synthesizer system is inflexible and has very limited knowledge of the data handled (Dutoit, 1997), this is the simplest synthesis method in producing the most natural sounding speech. Concatenative synthesizers which uses certain length of prerecorded samples from speech database (or also known as lexicon), is the most commonly used technique nowadays since it produces acceptable quality of speech and perhaps is the simplest way in producing intelligible and natural sounding synthetic speech.

The main part for this type of synthesis is choosing a unit for concatenation purpose. This selection will affect the overall performance and quality of the output speech where the longer length of a segmental unit implies a higher naturalness, less concatenation point and better control of coarticulation parameter. However, the amount of required units and memories will increase dramatically as the number of units needed to be concatenated and stored will increase (Lemmetty, 2003). While on the other hand, the selection of a shorter segmental unit might effectively overcome these problems yet the sample collecting and labeling procedures are more complex with higher distortion at the concatenation points. Nowadays, the units used in a system might consist of either syllables, demisyllables, phonemes, diphones or even triphone and words.

The most recent technique offers variable length unit selection (Yi, 1998) such as syllables and phones for high quality speech synthesis. The unit selection algorithm is similar as for searching the best state sequence via Viterbi algorithm in Hidden Markov Model. However, despite a promising higher quality of TTS system, the results are unpredictable and inconsistent. If the selection algorithm

failed to find the best suitable target unit, some prosodic modification would be carried out to the selected unit and this would contribute to the degradation of speech quality (Yi, 1998).

#### D. Sinusoidal Models

A sinusoidal model, also known as McAulay/Quatieri model, was developed for speech analysis and synthesis by McAulay and Quatieri in 1986 (Dutoit, 1999; Bozkurt, 2001). The basis of this technique is through the assumption that speech signal can be represented by the sum of sine waveforms (Lemmetty, 1999) with time varying amplitudes and frequencies where the speech signal  $s(n)$  is represented by  $L$  number of sinusoids while amplitude and phase each represented by  $A_i(n)$  and  $\theta_i(n)$ .

$$s(n) = \sum_{l=1}^L A_l \cos(w_l n + \theta_l)$$

The sinusoidal analysis/synthesis system is shown in Fig. 2. This method is claimed to be more applicable for singing voice synthesis with LYRICOS which is said to be the best synthesis system (Lemmetty, 1999) developed for this purpose.

Even though this model might represent periodic signals such as vowel and voiced consonants very well, the unvoiced sounds are poorly modeled. The modifications of the basic model include Hybrid/Sinusoidal Noise and ABS/OLA, a combination of sinusoidal model with analysis by synthesis/overlap where signal is expressed as a sum of overlapped short-time signals represented as a sum of sinusoidal (Stylianou et al, 1997).

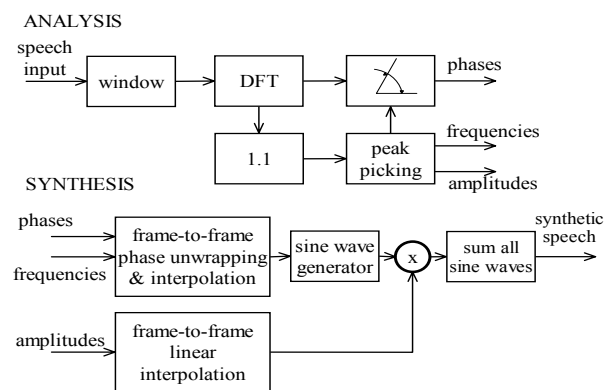


Fig. 2 Sinusoidal analysis / synthesis system

#### E. Hybris System

Various experiments have been made to improve the performance and speech quality of concatenated based synthesizer while several systems even allow modification of the synthesized speech itself. Combination of time domain and frequency domain might be a good choice as they complement each other's deficiencies.

In fact, formant synthesizers allow good control over fundamental frequency (pitch and duration) and produce flexible but rather synthetic sounds while time domain

synthesizers produce more natural sounding speech but synthesized speech faces distortion and discontinuities at some segment boundaries. This type of combination is called hybrid system and the basic idea of the system is shown in Fig. 3. One of the most used hybrid models is Harmonic Noise model (HNM) which is a combination of sinusoidal modeling and LPC framework.

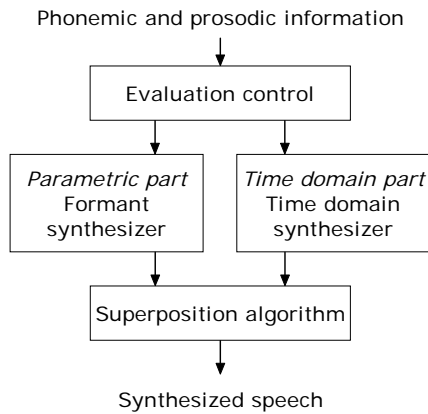


Fig. 3 Basic idea of hybrid synthesis system

Besides all the methods discussed in this paper, a lot of techniques are available for determining the control parameters (duration, pitch, gain and fundamental frequency) for a speech synthesizer. Hidden Markov Models (HMM) and Neural Networks (NN) based methods, for example, are some of the methods commonly and successfully used within speech synthesis.

F. Harmonic and Noise Models (HNM)

HNM is a hybrid model which represent the speech signal in twofold. The first phase is the deterministic part where the signal is decomposed as sum of related sinusoids that vary slowly in amplitudes and frequencies, and secondly the stochastic noise component is for other than described by the harmonic components where the residual signal is obtained by subtraction of the sinusoidal components from the original speech (Dutoit, 1997). Basically, the working principal is similar to that of PSOLA method (Dutoit, 1997; Lemmetty, 1999) where synthesis is done by overlapping and adding pitch synchronous segments but it does not require pitch marks (Stylianou et al, 1997) to be determined as in PSOLA method. The lower band of a voiced speech segment is modeled by deterministic component while the upper band is modeled by an AR model and modulated by time-domain amplitude envelope. The whole spectrum in the unvoiced speech segment is modeled by stochastic noise (Stylianou et al, 1997; Benjamin, 1999).

The deterministic component  $h(t)$  is

$$h(t) = \sum_{k=1}^{K(t)} A_k(t) \exp(jk\theta(t)) \quad \text{with} \quad \theta(t) = \int_{-\infty}^t w_0(l) dl$$

and  $A_k(t)$  is the component amplitude and phase at time  $t$  of  $k$ -th harmonic,  $w_0(t)$  is the time varying fundamental

frequency while  $K(t)$  is the time-varying number of pitch-harmonic.

Stochastic part which models the upper band is generated as white Gaussian noise,  $b(t)$  is filtered by a time-varying all-pole filter,  $F(t, z)$  and the time domain structured by an energy envelope function,  $w(t)$  to allow time-stretching factors without generating undesired periodicity in unvoiced sounds (Stylianou et al, 1997).

$$n(t) = w(t)[F(t, Z) * b(t)]$$

The analysis and synthesis scheme is shown in Fig. 4 and 5 accordingly.

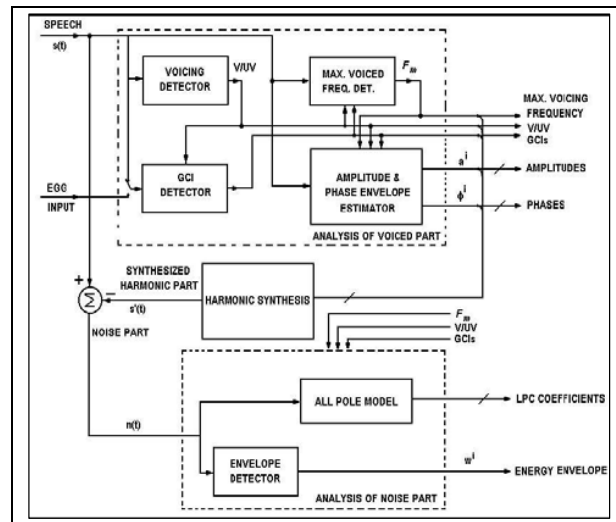


Fig. 4 Analysis of speech using HNM

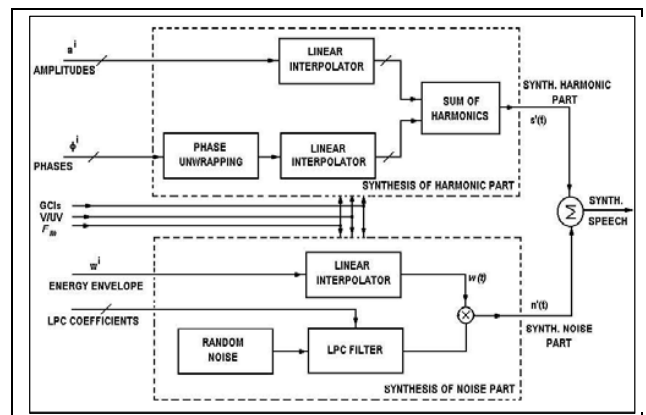


Fig. 5 Synthesis of speech using HNM

Since HNM assumes that speech signal is composed from two different parts, its use for TTS system yield to a straightforward timescale and pitch-scale modifications (Stylianou et al, 1997) with much simpler way for smoothing the boundaries, thus producing a more natural-sounding synthesized speech. Several tests carried out by researchers

(Stylianou et al, 1997) using natural prosody have shown that the quality of synthetic speech is proved to be the best to approximate the quality of the original sentences without distortion problems if compared to other methods where new voices can also be easily integrated into TTS system since this method does not require glottal closure instants.

### III. HISTORY OF MALAY SPEECH SYNTHESIZER AND TECHNIQUES USED

Most of the Malay speech synthesizers proposed are generally small scale projects or based on other open source or commercial speech synthesizer. There are many TTS systems available nowadays but only few can be considered as able to give great contributions to the world of Malay speech synthesizer. This section will discuss only the most significant systems or proposals in term of theoretical frameworks, NLP module, DSP module or even the system itself.

The first and the most successful Malaysia text-to-speech software were thought to be FASIH which was launched by Mimos Bhd last year. It is thought to be the leader of Malay TTS engine due to the ability of the system to produce an unrestricted vocabulary of Standard Malay with natural sounding speech. The first version of Fasih was successfully commercialized and used in training software, QuickDo. Other applications included e-mail reading, language-based training software and other typical voice service applications. This diphone based concatenative TTS system uses time domain MBROLA as its speech synthesizer where MBROLA itself was inspired by MBR-PSOLA algorithm. The diphone database which is specially adapted to the requirements of the synthesizer was obtained via hybrid Harmonic/Stochastic analysis-synthesis of the database, resulted in the flexibility of parametric speech models while keeping the computational simplified.

Another attempt to build a Malay speech synthesizer by adapting MBROLA algorithm was made by Nur-Hana Samsudin and Tanya Enya Kong (Samsudin and Kong, 2004) from University Science of Malaysia. The system used four syllable structure of consonant-vowel clusters (CV), vowel-consonant clusters (VC), consonant- vowel –vowel clusters (CVC) and vowel (V) cluster with a few sub-models proposed for exception such as loan words pronunciation. The database used prerecorded syllable segment from a native Malay speaker to avoid phonological problem derived from the use of Speech Application Programming Interface (SAPI) due to the fact that only American English phonological representation is used in the interface, hence yield to the sound of Malay Language being very foreign and awkward. However, segment discontinuation and distortion at the boundaries cannot be avoided since the database was built without prosody modification.

Another Malay TTS synthesizer was established by Yousif A. El-Imam and Zuraidah Md. Don. They proposed a system based on unit-selection methods with four synthesis units, namely, CV, VC, vowel- consonant- vowel clusters (VCV) and consonant- consonant clusters (CC). Each of the synthesis units contains 162, 162, 972, and 729 clusters respectively. All the input text would first tagged with this CV rules before

the syllable segmentation that is used to process text utterance can be obtained.

The system which adapted from a previously developed synthesizer for Standard Arabic language also proposed a general linguistic analysis and phonological aspect of Standard Malay and loan words from Arabic language that can as well be implemented to the NLP module of our Malay TTS synthesizer system. A lexicon containing all the special properties such as abbreviations, acronyms, and special symbols will divide the user input into two fields, the orthography of the item and its pronunciation of the words or the representative word sequence. The database would be scanned for the first matching entry.

In *Say It!* system [4], the segmenting technique is to select the longest phoneme sequence and compare the selected sequence in the available syllable database. If matches occur, the sequence will be taken out and consider as a syllable unit. Else, the last phoneme in the done again with the reduced phonemes sequence. The process will be repeated until the match is found in the database. This technique does provide a simple implementation and produced quick result but the parsing could also be segmented wrongly.

### IV. THE STANDARD MALAY LANGUAGE AND PHONOLOGY SYSTEM

The Standard Malay Language or Bahasa Baku (the word Baku comes from a Javanese word which means true and correct) was made upon agreement made by Malaysia, Indonesia and Brunei, is Bahasa Riau. This implies that the spelling, words, phrasing, grammar, pronunciation, punctuation, sentences, abbreviations, acronyms, capital letters, numbering and style of the language are already standardized. In this research, it is important to emphasize that the output of our Text-To-Speech system will only be in spoken Standard Malay.

Standard Malay is written in 26 Latin alphabets consists of six vowels, nineteen primary consonant, native consonant sounds and eight secondary consonants (consonants borrowed from other languages). The details of the phonological rules for this system would be discussed in detail in the following chapter. The vowels used in the correct spelling of the language are a, e, i, o and u. However, it is important to not that there are two different types of 'e' in Malay words; for instance, '*teman*' (friend) using 'e pepet' and '*senget*' which uses 'e taling'. This resulted in the distinction and disambiguation between the two 'e's. The 'e pepet' was originated from Sanskrit language adapted to Malay modern phonological system. In ancient Malay, the word '*sepuluh*' (ten) for example, is originated from the word *sapuluh*. Another example; the consonant b were actually v in Sanskrit language; in example *bulan* (moon) is originated from the word *vulan*. All the h in modern words were also deleted from the origin word; in example, *sahaya* becomes *saya* (means me) and *samuha* becomes *semua* (which means all). In addition, few consonants that are available nowadays are originated from some consonants or diphthongs from foreign language. Some other modification made to were as in the following table:

TABLE I  
EXAMPLES OF MALAY LANGUAGE

Ancient Malay	Standard Malay	Loan words	Malay words	Meaning
th	s, t	therapy	terapi	therapy
dh	d, z	dhaif, dhalim	daif, zalim	cruel
sh, ch	c	shitta	cita	ambition/dream
kh	k	sukha	suka	like/love
yi	i	nayik	naik	climb up/ go up
n	ny	vanak	banyak	plenty of
oo	u	mee- hoon	bihun	mee hoon (Chinese food)

#### A. Text-To-Phoneme Conversion

In most text-to-speech systems the ASCII representation of each input sentence is given as input to the text analysis module of the system. The input is analyzed in such a way as to:

- Reformat everything encountered (e.g., digits, abbreviations) into words and punctuation
- Parse the sentence to establish the syntactic structure
- Find the semantically determined locations of contrastive and emphatic stress
- Derive a phonemic representation from each word
- Assign a stress pattern to each word

#### B. Text Formatting

All TTS Systems have a preprocessing module for formatting the input text [1,2,3,4,5,6,7]. This module organizes the input sentences into manageable lists of words. It identifies numbers, abbreviations, acronyms and idiomatics and transforms them into full text when needed (i.e. \$35.61, 35.61, 2000, the year 1971, 10:15 p.m.). This is very practical for text-to-speech systems. Commercial systems, which must be prepared to deal with more exotic material such as embedded escape sequences and other nonalphabetic characters, have adopted two general strategies.

Klatt indicates that TTS Systems such as the Infovox SA-101 and the Prose-2000 provide the user with a set of logical switches which determine what to do with certain types of nonalphabetic strings. For example, "-" is translated to either "dash" or "minus" depending on the state of a switch. And adds, DECTalk ignores escape characters, and usually spells out words containing nonalphabetic characters [4].

#### 1. Letter-To-Phoneme Conversion

One issue in the preparation of rules and data structures for synthesis is how to best represent phonemes, allophones, stress, and syntactic symbols. Computers often require a representation that can be printed within the limitations of the ASCII character set. There is no agreement on either the set of phonetic symbols to be represented or the phonetic/ alphabetic correspondences in this situation.

In order to derive a phonemic representation of a word, letter-to-sound rules and exceptions dictionary are used. For such languages that by using affixes, many forms of a word can be obtained (i.e. *Turkish, Finnish*), an alternative to this method is to develop a large morpheme dictionary and try to decompose each input word into its constituent morphemes (i.e. *stem + affixes*).

In Bell Labs., they used a set of *letter-to-sound rules* that simply map sequences of graphemes into sequences of phonemes, along with possible diacritic information, such as stress placement [6].

#### 2. Prediction of Lexical Stress from Orthography

The newer systems not only base stress assignment on factors such as morphological structure and the distinction between strong and weak syllables, but also on presumed part of speech. Klatt emphasizes the importance of syntactic categorization which uses morphological decomposition, involved situations when the surface form did not contain a silent "e" (choking -> choke + ing), there had been consonant doubling (omitted -> omit + ed) or a final "y" had been modified (cities -> city + s). Some morphemes are pronounced differently depending on the stress pattern of the word and the nature of the other morphemes present (note the second "o" of "photo" is realized phonemically as /o,ə,a/ in "photo," "photograph," "photography," respectively) [4].

One of the advantages of a morpheme lexicon, aside from an ability to divide compound words properly, is that a set of 12 000 morphemes can represent well over 100 000 English words. Thus a very large vocabulary is achieved at moderate storage cost. However, the greatest advantage of the morpheme lexicon may turn out to be its ability to specify parts of speech information to a syntactic analyzer in order to improve the prosody of sentences.

#### 3. Syntactic Analysis

Furthermore, some pronunciation ambiguities can be resolved from syntactic information. For example, there are more than 50 noun/verb ambiguous words such as "permit" that are pronounced with stress on the first syllable if a noun, and with stress on the second syllable if a verb. The only way to pronounce these words correctly is to figure out the syntactic structure of an input sentence, including the location of the verbs. Thus it would be highly desirable to include a parser in a text-to-speech system.

While powerful parsing strategies exist, they tend to produce many alternative parses, even for sentences that seem simple and unambiguous. For example, "Time flies like an arrow" is multiply ambiguous at a *syntactic* level; a syntactic analysis system would require an immense store of world

knowledge (semantics/ pragmatics) to behave as we do and focus immediately on the only sensible structural interpretation of the sentence.

If a parts-of-speech categorization is not available for most words, the simplest parsing strategy would be to use function words such as prepositions, conjunctions, and articles to find obvious phrase boundaries, leaving the remaining boundaries undetected.

#### 4. Semantic Analysis

Semantic and pragmatic knowledge is needed to disambiguate sentences. Klatt gives an excellent example showing the ambiguity in a sentence such as “*She hit the old man with the umbrella*”. There may be a pseudopause (a slowing down of speaking rate and a fall-rise in pitch) between the words “*man*” and “*with*” if the woman held the umbrella, but not if the old man did. Similarly, a “*rocking chair*” will have the word “*chair*” stressed if the combination of adjective and noun has been associated by frequent use into a single compound-noun entity.

Emphasis or contrastive stress may be applied to an important word depending on the meaning: “*The OLD man sat in a rocker*” (not the younger man). Finally, words that have lost their importance in a dialog, either because of prior occurrence of the word or by anaphoric reference, should be

### V. DIGITAL SIGNAL PROCESSING MODULE

#### A. Phonemes-to-Speech Conversion

Each syllable of a word in a sentence can be assigned a strength or *stress* level. Differences in assigned stress make some syllables stand out from the others. The stress pattern has an effect on the durations of sounds and on the pitch changes over an utterance. The phonological component of the grammar converts phonemic representations and information about stress and boundary types into (1) a string of phonetic segments plus (2) a superimposed pattern of timing, intensity, and vocal cord vibrations which are known as sentence *prosody*.

For synthesis of natural-sounding speech, it is essential to control prosody, to ensure appropriate rhythm, tempo, accent, intonation and stress.

The *phonological component* of the grammar includes rules to make the substitutions of phonemes, either by replacing one symbol by another, or by changing the feature representation of a phoneme. In mapping phonemes into sound, traditional linguists recognize a second intermediate level of representation that has been termed the phonetic segment or *allophone*. For an extreme example, the phoneme /t/ may be replaced by one of six distinctly different allophones [6].

Many steps are required in order to convert a phoneme string - supplemented by lexical stress, syntactic, and semantic information - into an acoustic waveform. Most of the phonemes are realized in their canonical phonetic form. These canonical allophones might be modified by some rules involving stress, duration, and phonetic context. Next, each phonetic segment is assigned an inherent duration by table lookup, and a set of duration rules is applied to predict

changes to the duration of the segment as a function of sentential context. Stressed vowels are lengthened, as are the consonants that precede them in the same syllable. Then, a fundamental frequency (*f<sub>0</sub>*) contour is determined by rules that specify the locations and amplitudes of step and impulse commands that will be applied to a lowpass filter in order to generate a smooth *f<sub>0</sub>* contour as a function of time. The rises and falls set off syntactic units. Stress is also manifested in this rule system by causing an additional local rise on stressed vowels, using the impulse commands. The amount of rise is greatest for the first stressed vowel of a syntactic unit, and smaller thereafter.

#### 1. Speech Synthesis

Speech Synthesis Systems aim to produce speech that is both intelligible and natural. There are two types of synthesis [2,4,6],

1. Rule-Based Synthesis
2. Concatenative Synthesis

##### i. Rule-Based Synthesis

An important advantage of the language is an ability to refer to natural sets of phonemes through a distinctive feature notation, making rule statement simple, efficient, and easy to read. These rules are then compiled automatically into a synthesis-by-rule program. A number of languages (Swedish, Norwegian, American English, British English, Spanish, French, German, and Italian) have been synthesized using this system and the resulting system has been brought out as a product. Modern systems such as MITalk contain special set of rules for translating phonemics in to allophonic input [1].

##### ii. Concatenative Synthesis

In this system smaller units are recorded in order to synthesize the sound, since it is impossible to store the whole words. Turkish is very suitable for concatenative synthesis since generally the phonetical representations of the spells do not depend on the position in the word. Concatenative speech synthesis systems generate speech by concatenating and manipulating prerecorded units of speech. Three design decisions are particularly important: choice of units, storage of units, and concatenation method. Concatenating single phones, the first approach to concatenative synthesis, yields rather poor quality. Wolters claims that quality improves dramatically when diphones are used instead. Diphone units consist mainly of the transition between two phones  $p_1$ ,  $p_2$ . The unit boundaries are in the steady states of these phones in order to allow smooth concatenation. Wolters gives an example that, when synthesizing the word test /tɛst/, we have to concatenate the diphones #-t, t-ε, ε-s, s-t, and t-# [7]. Diphone synthesis is based on two assumptions:

1. The transition between the two phones is sufficient to model all necessary coarticulatory effects.
2. The spectra of the steady states of the phones are consistent enough across different instances to avoid grave spectral mismatches at the concatenation points.

Another category of approaches in concatenation synthesis, exemplified by Hunt & Black, derives the units from a phonetically balanced speech corpus, where each phone has been labelled with name, pitch, and other relevant information. For a given utterance, the synthesis algorithm now searches for a sequence of speech from the corpus that minimizes concatenation costs [7].

For minority languages, a completely corpus-based approach would be ideal for several reasons:

1. **Recording:** meaningful text is easier to read than nonsense words.
2. **Unit selection and concatenation:** It would not be necessary to write a separate unit selection algorithm, if the standard algorithm is flexible enough.

On the other hand, phonetically balanced texts that provide all necessary units are very hard to design.

In TCTS Laboratories, both methods described above, were tried. In the dictionary based solution the software stores the maximum of a phonological knowledge into a lexicon. It only stores morphemes in order to restrict the size and then it can generate the sound of inflected, derived and compound words by using these morphemes. In Rule based solution, most of the phonological competences of dictionaries are transferred into a set of letter-to-sound rules. An exceptions dictionary is also stored in order to keep the words that are pronounced in a particular way. Then they concluded that the method used gives better result according to the language [2].

Wolters makes a comparison between two methods and concludes that; rule-based synthesis allows considerable freedom; a high number of adjustable parameters make high-quality speech output possible. There are no discontinuities that result from concatenating two prerecorded speech units. But this freedom is also the biggest disadvantage of rule-based synthesis: setting the parameters and devising rule sets such that the resulting speech is both intelligible and natural is very difficult, even more so for articulatory than for acoustic rule-based synthesis, because as yet, we know very little about the mechanisms of speech production. On the other hand, a concatenative approach only needs a reliable auditory analysis of the language's phonetics, a good concatenation algorithm, a patient speaker, and enough time for segmenting the units. A thorough phonetic description is available for almost all languages that have been described linguistically, because phonetics is easiest to analyse for a trained field worker. Concatenative synthesis also has the added advantage of sounding more natural than a rule-based voice after the same amount of work, since the units that are manipulated are original speech recordings with the right "parameter" values already built in [7].

### B. Sound Database

The sound database contains the set of elementary sounds (speech units). Concatenating elementary sounds we can generate a sound signal corresponding to any text.

According to Ferencz the speech units can be chosen between: words, sentences, morphemes, syllables, phonemes, demisyllables, etc., according to the requirements of the

application. Using words and sentences as basic units (having them recorded with intonation and articulation) we can obtain high quality speech but for restrained domains (for example portable dictionaries). Morphemes are alternative units which can be used [3]. The English language contains, for example, 12.000 morphemes (like book, ed, have, s).

If we want to have an unrestricted vocabulary the storage space becomes too big, then the idea of recording all the words becomes inefficient. The solution is to use as speech units some more elementary sounds like phonemes. But here we meet the disadvantage that a phoneme corresponds to an infinite - but specified - class of temporal or frequencial variants. Ferencz indicates that the physical features of a phoneme vary from one speaker to another, and even if the speaker is the same there can be changes depending on the speaker's state, the place of the phoneme and of the accent in the word, the intonation and the accent in the phrase, the pronouncing duration, etc. [3].

Using phonemes as basic units we need interpolation at the transition from one phoneme to another because the vocal tract does not change shape abruptly, gliding smoothly from one articulation position to another. The effect of this transition must be incorporated into the algorithm by inserting sets of interpolated parameters between neighboring phonemes. This works well with slow transition as in case of vowels, but in the case of consonants the transition is too fast and the acoustic effect is lost. To overcome this problem Ferencz suggests usage of diphones or demisyllables [3].

If we want a trade-off between the storage space and the production of an intelligible speech we can use the diphones as database elements. A diphone is a sound consisting of the two neighboring halves of two adjacent phonemes. Then a diphone starts in the middle of the first phoneme and ends up in the middle of the second. Almost any combination of two phonemes could make up a diphone, so the number of diphones in a language is at most equal to the square of the number of phonemes in that language. In the case of synthesis with diphones the sound database will consist of all the diphones in the language.

A diphone database is used for a TTS System prepared in Romanian Language [3]. For Turkish, morpheme based database is ideal since Turkish words constructed from unite of morphemes.

### C. Prosody and Sentence-Level Phonetic Recording

A sentence cannot be synthesized by simply stringing together a sequence of phonemes or words. It is very important to get the timing, intonation, and allophonic detail correct in order that a sentence sounds intelligible and moderately natural.

A pure tone can be characterized by prosody, which is an important aspect of speech. It expresses linguistic information such as sentence type and phrasing as well as paralinguistic information such as emotion. It is characterized by three parameters: fundamental frequency, duration, and intensity. Prosodic phonology examines the grammar of prosody and the relation between prosodic units and segments. Prosodic units are characterized by a variety of phonetic markers and form part of hierarchical schema. According to Klatt, the two most

important prosodic parameters are pitch and duration [4]. Furthermore, stress is expressed mainly by prosodic correlates, and very important to model.

The following two sections take up the parameters that characterizes the prosody in detail [2,4,5,7].

1. **Duration rules:** In reading a long sentence, speakers will normally break the sentence up into several phrases, each of which can be said to *stand alone* as an intonational unit. If punctuation is used liberally so that there are relatively few words between the commas, semicolons or periods, then a reasonable guess at an appropriate phrasing would be simply to break the sentence at the punctuation marks though this is not always appropriate. Psychological and semantic variables influence the average speaking rate and determine durational increments due to emphasis or contrastive stress.
2. **Fundamental frequency rules:** Many phenomenological observations have been collected about pitch motions in English sentences, and hypotheses have been generated concerning their relations to linguistic constructs known as intonation and stress. The intonation pattern is defined to be the pitch pattern over time that, for example, distinguishes statement from question or imperative, and that marks the continuation rise between clauses for an utterance of more than one clause. The stress pattern on syllables can distinguish words such as "insert" from "ins'ert" even though the two words have identical segmental phonemes.
3. **Intonation:** There are many different methods for describing the fundamental frequency rules contour of an utterance. The British school (Cruttenden 1997) uses a contour based description: a pitch contour is a sequence of rises and falls. The American school, on the other hand, describes intonation as a sequence of pitch targets. There are two levels of targets and tones: high (H, maximum) and low (L, minimum). These targets mark either a pitch accent, that is an extremum in the pitch contour, or a prosodic boundary (boundary tone). Both approaches are on a phonological level: they are used to describe the rough pitch contour of an utterance and have to be transformed into phonetic descriptions. Describing and synthesizing intonation is a difficult task. For minority languages, time constraints will only permit to model a few typical pitch contours, and set global parameters such as base line and declination.
4. **Stress:** Stress is an important information carrier. *Word stress* determines which syllable in a word is stressed; *phrase stress* determines the words in a phrase that receive stress. Stress can be signaled by all prosodic parameters, pitch, intensity, and duration, as well as by segment quality. Stressed syllables tend to be longer than unstressed ones, and they are usually further marked by a local maximum or minimum in the fundamental frequency contour.

#### D. Allophone Selection

Words are lexically represented by phonemes and stress symbols. Allophone selection is then an important aspect of the sentence generation process.

Heffner shows that the part of the problem of speaking naturally concerns the phonetic form of function words. Words such as "for," "to," "him" often take on the reduced forms [fɜ], [tə], and [ɪm], but not in all phonetic environments [4].

Klatt claims this area of allophonic detail and prosodic specification is one of the weaker aspects of rule systems, and contributes significantly to the perception of unnaturalness attributed to synthetic speech. Incremental improvements that are made to these rules on the basis of comparisons between rule output and natural speech cannot help but lead to improved performance of text-to-speech systems [4].

#### VI. EVALUATION OF TEXT-TO-SPEECH SYSTEMS

In order to evaluate Text-to-speech systems Klatt specifies some criterias. This evaluation and comparison is done with respect to intelligibility, naturalness, and suitability for particular applications. One can measure the intelligibility of individual phonemes, words, or words in sentence context and one can even estimate listening comprehension and cognitive load [4].

**Intelligibility of isolated words:** The measurement of intelligibility can be performed in many different ways. Since consonants have been more difficult to synthesize than vowels, the modified rhyme test is often used, in which the listener selects among six familiar words that differ only by an initial consonant or a final consonant. This is not a very severe test of system performance since the response alternatives may exclude a confusion that would be made if a blank answer sheet were used, but the test does facilitate rapid presentation to naive subjects and automatic scoring of answer sheets.

**Intelligibility of words in sentences:** In comparison with words spoken in isolation, words in sentences undergo significant coarticulation across word boundaries, phonetic simplifications, reduction of unstressed syllables, and prosodic modifications that, among other things, shorten nonfinal syllables and modify the fundamental frequency contour. In order to evaluate the ability of text-to-speech systems to realize these transformations, tests of word intelligibility in sentence frames have been devised.

**Reading comprehension:** Since synthetic speech is less intelligible than natural speech, what happens when one tries to understand long paragraphs? Do listeners miss important information? Is a listener so preoccupied with decoding individual words that the message is quickly forgotten? In an attempt to answer these questions, Pisoni and Hunnicutt included a standard reading comprehension task in their evaluations. Half the subjects read the paragraphs by eye, while the other half listened to a text-to-speech system. In a later experiment, comparison was made with a human voice reading the paragraphs. Studies have shown that there is a



wide range of performance between text-to-speech systems in terms of segmental intelligibility. Measured in terms of error rate, a system with a 3% error rate is twice as good as one with a 6% error rate, at least in terms of the average time interval between misperceptions in running text [4].

**Naturalness:** Naturalness is a multi-dimensional subjective attribute that is not easy to quantify. Any of a large number of possible deficiencies can cause synthetic speech to sound unnatural to varying degrees. Fortunately, systems can be compared for relative subjective naturalness with a high degree of inter-subject and test-retest agreement. A standard procedure is to play pairs of test sentences synthesized by each system to be compared, and obtain judgments of preference. As long as the sentences being compared are the same, and the sentences are played without a long wait in between, valid data can be obtained.

**Suitability for a particular application:** Text-to-speech devices are being introduced in a wide range of applications. These devices are not good enough to fully replace a human, but they are likely to be well received by the general public if they are part of an application that offers a new service, or provides direct access to information stored on a computer, or permits easier or cheaper access to a present service because more telephone lines can be handled at a given cost.

## VII. CONCLUSION

As indicated in the above paragraphs, speech synthesis will be studied continuously, aiming more natural and intelligible speech. It is quite certain that TTS technology will create new speech output applications associated with the improvement of speech quality. We also have to consider variabilities resulting from human factors, such as speaking purpose, utterance situation and the speaker's mental states. These paralinguistic factors cause changes in speaking styles reflected in a change of both voice quality and prosody. The investigation of these variations will contribute to elaborate synthetic speech quality and widen its application fields.

In the papers that we have made researches, we realized that synthetic conversion of a written text is easy. But the important part of these systems is to make more natural conversion, which means reading a text like a human with stressing, intonations and durations. Furthermore, the structure of the TTS Systems differs by the language, which the system prepared for.

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