Integrating Language Generation with Speech Synthesis in a Concept to Speech System

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Abstract

Concept To Speech (CTS) systems are closely related to two other types of systems: Natural Language Generation (NLG) and Speech Synthesis (SS). In this paper, we propose a new architecture for a CTS system. A Speech Integrating Markup Language (SIML) is designed as an general interface for integrating NLG and SS. We also present a CTS system for a multimedia presentation generation application. We discuss how to extend the current CTS system based on the new architecture. Currently, only limited semantic, syntactic and prosodic features are covered in our prototype system.

1 Introduction

Currently, there are two ways to develop a Concept-To-Speech (CTS) system. The first is to design a monolithic CTS system for a specific application. This design involves a specific NLG module and an SS module, often developed for the application, where discourse, semantic and syntactic information produced by the NLG module can be used directly by CTS algorithms to determine system specific parameters for a Text-To-Speech system, or phonological parameters for a vocal tract model (e.g., (Young and Fallside, 1979)). One advantage of this design is its efficiency, but features from the two systems are usually so intertwined that the interface of the CTS algorithms are system dependent. Another design is to keep NLG and SS as independent as possible, thus allowing reuse of the current NLG tools and TTS systems for other applications. The typical design is equivalent to "NLG plus Text-To-Speech(TTS)" where the common interface between NLG and TTS is plain text. One advantage of this is its simplicity and adaptability. No change is necessary for existing NLG tools and TTS systems, but it suffers from a serious problem in that it loses useful information. All discourse, semantic and syntactic information is lost when the internal representation of NLG is converted to the text output and clearly this could be useful in determining prosody.

In this paper, we want to maintain the autonomy of NLG and SS so that they are reusable for different applications, yet flexible enough to easily integrate without losing useful information. We propose a new architecture in which the common interface is not plain text, but a Speech Integrating Markup Language (SIML). We show how this architecture can be used in a multimedia presentation application where a prototype SIML was designed for this purpose.

2 Related Work

Recently, people have become more interested in developing CTS algorithms to improve the quality of synthesized speech. In (Prevost, 1995) and (Steedman, 1996), theme, theme and contrast are used as important knowledge sources in determining accentual patterns. In (Davis and Hirschberg, 1988), given/new and topic structure are used to control intonational variation. Other CTS related research includes (Young and Fallside, 1979) and (Danlos et al., 1986). Most of the CTS systems developed to date have a closely integrated architecture. Because of this, CTS algorithms which map information from NLG to TTS parameters are system dependent.

There is some related research in developing markup languages for TTS and speech transcription. The Speech Synthesis Markup Language (SSML) (Isard, 1995) is used as an interface for TTS. The motivation behind SSML is to overcome the difficulty that different TTS systems require different input format. No additional information is provided as input to TTS, but SSML provides a straightforward representation of existing prosodic features. This
representation is too simple for the purpose of integrating NLG and SS for CTS. There is almost no discourse, semantic or syntactic information in their representation, yet these are features one would expect as output from NLG and which should influence the prosody of speech.

The Text Encoding Initiative (TEI) (Sperberg-McQueen and Burnard, 1993) provides a general guideline for transcribing spoken language using Standard Generalized Markup Language (SGML). SGML is an international standard for encoding electronic document for data interchange. Integrating two components in CTS is a specific SGML application. Therefore, it can't be addressed directly in SGML. But the design of SIML can be guided by TEI standards.

3 System Architecture

The main new feature of the architecture (see Fig. 1) is the introduction of SIML. The system has three major components: the NLG component, the SIML To Prosody Component (STP) and the TTS component. Each can be designed and implemented independently. The NLG-SIML component first converts the input concepts into grammatical sentences with associated discourse, semantic, and syntactic information. Then the SIML converter transforms the system specific NLG representation into standard SIML format. The STP component computes the prosodic features based on the discourse, semantic and syntactic information encoded in the SIML format. The STP component has three modules: the SIML parser, the STP algorithms and the SIML generator. First the SIML parser analyzes the information in SIML. The STP algorithms predict prosodic parameters based on the information derived from the markup language. Then the SIML generator encodes the prosodic features in SIML format. The TTS component first extracts the prosodic parameters from the SIML representation and translates it into a specific, system dependent TTS input. In this way various NLG tools, STP algorithms and TTS can be integrated through the standard interfaces, SIML.

4 MAGIC CTS system

Our CTS system is a component of the MAGIC system (Multimedia Abstract Generation for Intensive Care) (Dalal et al., 1996) (Pan and McKeown, 1996). MAGIC's goal is to provide a temporally coordinated multimedia presentation of data in an online medical database. The graphics and speech generators communicate through a media coordina-

In a semantic representation, a clause is defined by process type, participant and circumstance. Process type could be simple, as in the example, or composite (e.g., using conjunction). Each participant or circumstance may consist of a head and one or more pre-modifiers or qualifiers. Words and phrases are used to realize each semantic unit.

The surface realizer maps the lexicalized, semantic representation to its corresponding syntactic struc-
After linearizing the syntactic structure, which usually is the last step in a written language generation system, the internal semantic and syntactic structure as well as the words of the sentence are used as a rich and reliable knowledge source for speech synthesis.

**CTS Algorithms in MAGIC**

Due to the synchronization requirements, we are specifically interested in two features: pause and speaking rate. We want to increase or decrease the length of pauses or the speaking rate in such a way that speech actions begin and end at the same time as corresponding graphical actions. Even a small drift can be noticed by human eyes and cause uncomfortable visual effects. In MAGIC, only pause and speaking rate are set by our CTS algorithms; all other prosodic features are set by the default values predicted by AT&T Bell Labs’ TTS system.

Currently, we use a simple strategy in adjusting the speaking rate. We define the relative speaking rate as the ratio of the real speaking rate to the default speaking rate. Through experiments, we determined that the relative speaking rate can vary from 0.5 to 1 without significantly affecting the speech quality. In the future, we plan to develop an algorithm where the adjustable range is not uniform everywhere but decided by the underlying discourse, semantic and syntactic structures.

In the following, we give more detail on the CTS algorithm which is used to predict prosodic phrase boundaries. It provides a reliable indication on where pauses can be inserted and how long the pause could be.

We use semantic structures to derive the prosodic phrase boundaries. In our algorithm, we first identify the basic semantic unit (BSU), which is the smallest, complete information unit in the semantic structure. Then we define the closeness measurement between two adjacent BSUs. If two adjacent BSUs are loosely connected, then we have reason to believe that it won’t hurt the intelligibility significantly if we speak them separately. Therefore, semantic closeness is an important knowledge source for prosodic phrase boundary prediction. Other factors which also affect the placement of prosodic phrase boundary are breath length, and the distance to the end of the utterance.

A Basic Semantic Unit (BSU) is a leaf node in a semantic hierarchy. In the semantic hierarchy (see Fig. 2), the BSU is indicated by dark blocks.

We define the closeness between two adjacent BSUs as the level of the lowest common ancestor in the semantic hierarchy. If a node has only one child, then both parent and the child are considered at the same level. The closeness indicates the semantic distance of two adjacent BSUs. 1 means they are semantically far apart, while higher numbers indicate they are semantically close.

Breath length is defined as the typical number of words a human can speak comfortably without breathing. The value used in the algorithm is learned automatically from a corpus. The distance from the current place to the end of an utterance is simply defined by the number of words.

Now we have 3 factors working together determining the prosodic phrase boundary. Basically, there won’t be any prosodic phrase boundary within a BSU. For each place between two adjacent BSUs, we measure the possibility of inserting a prosodic phrase boundary using the combination of the 3 factors:

1. The larger the closeness measurement, the less the possibility of a boundary.
2. The closer the current breath length to the comfortable breath length, the more the possibility of a boundary.

3. The closer the current place to the end of the utterance, the less the possibility of a boundary.

4. The above factors are weighted, using a learning algorithm we trained automatically on a small corpus (40 sentences).

The result is encouraging. When we test this on the set provided in (Bachenko and Fitzpatrick, 1990), we get a 90% accuracy for primary phrase boundary and we get an 82% accuracy for the utterances in (Gee and Grosjean, 1983). We did not formally measure the algorithm for secondary phrase boundaries, because we only consider inserting pauses at primary phrase boundary.

**TTS in MAGIC**

Basically, we treat **TTS** as a black box in **MAGIC**. We use the escape sequence of **TTS** to override the **TTS** default value.

**5 Extensions to MAGIC CTS Based on the New Architecture**

The current **MAGIC** **CTS** uses **CTS** algorithms that are closely integrated with both the **NLG** tools and **TTS**. This will make it difficult to experiment with new tools, requiring changes in all the input and output format for the **CTS** algorithms. In the spirit of developing a portable language generation system such as **FUP**/**SURGE**, we are working on a portable spoken language generation system by using the new architecture.

**Extension 1: Design SIML for MAGIC**

In order to extend the current **CTS**, we must define a prototype **SIML**. As a first step, we have designed a prototype **SIML** that covers the information needed for **CTS** in the multimedia context. For our **CTS** algorithms, only semantic and syntactic structure are used in predicting prosodic phrase boundary and are represented in the **SIML**. Speaking rate and pause are also included in **SIML**.

We first describe how this information is represented in **SIML**, giving examples showing how to use **SIML** to tag pauses, speaking rate, semantic and syntactic structure. Then part of the formal Document Type Definition (**DTD**) of the prototype **SIML** is presented, providing a grammar for **SIML**. See (Sperberg-McQueen and Burnard, 1993) for more information about **SGML** and **DTD**.

**Example 1:** Using **SIML** to tag speaking rate and pauses:

```
<phrase dur=5 durunit=ms>Ms. Jones</phrase>
```

and

```
</phrase>
```

is the front and end tag of a phrase. **Rate** is an attribute associated with **phrase**. **Rate** indicates the speaking rate of the phrase. **Pause** is a tag with...
two associated attributes: dur and durunit. They indicate the length of the pause.

Example 2: using SIML to tag semantic structure:

```xml
<clause>
  <participant role=carrier>
    <np sem>
      The patient
    </np sem>
  </participant>
  <proc type=ascriptive lex=be>
    is
  </proc>
  <participant role=attribute>
    <adjp sem>
      hypertensive
    </adjp sem>
  </participant>
</clause>
```

Example 3: using SIML to tag syntactic structure:

```xml
<sentence>
  <np art=art>
    The patient
  </np>
  <vp verb=is>
    hypertensive
  </vp>
</sentence>
```

Part of the formal definition of SIML, using DTD:

```xml
<!-- DTD specifying speaking rate and pause -->
<!DOCTYPE utterance.pro [ !ATTLIST u.pro rate NUMBER ] !ATTLIST phrase rate NUMBER ] !ATTLIST pause rate NUMBER ] !ATTLIST u.pro dur NUMBER ] !ATTLIST phrase durunit NUMBER ] "ms">
```

In the above DTD specification, three elements and their associated attributes are defined:

- `u.pro` and its attribute, `rate`;
- `phrase` and its attribute, `rate`;
- `pause` and its attributes, `dur` and `durunit`.

The following is the element definition for `u.pro`:

```xml
<!ELEMENT u.pro [((#PCDATA | phrase | pause)*])>
```

Extension 2: Design the STP component

The STP component is the core part in the architecture and deserves more explanation. There are three tasks for this component: parsing of the input SIML, generation of prosodic parameters from the information produced by NLG, and transformation of the parameters into the SIML format. The SIML parsing is straightforward. It can be done either by developing an SIML specific parser for better efficiency or by using an SGML parser (there are several which are publicly available). The output of this component is the semantic and syntactic information extracted from SIML. Generation of prosodic parameters must be done using a set of CTS algorithms; we need to change the input and output of our existing CTS algorithms and make it system independent. Since the performance of these algorithms directly affects the quality of the synthesized speech, much effort is required to develop good CTS algorithms. The good news is that the proposed design ensures that the markup to prosody algorithms are system independent. Therefore, they can be reused in other applications. The output of the STP algorithms then converts to the SIML format by the SIML generator. The procedure is straightforward and it can be done very efficiently.

6 Generalize SIML

Since the current prototype SIML is designed specifically for multimedia application, it includes very limited semantic, syntactic and prosodic information. Thus, it is currently too primitive to be used as a standard interface for other CTS applications. For the future, we must include other forms information that are needed for speech synthesis and that can be generated by an NLG system. Some types of knowledge that we have identified include:

1. Discourse information (e.g. discourse structure, focus, rhetoric relations etc.), semantic structure and its associated features (such as in the prototype SIML), and syntactic structure.

2. Pragmatic information such as speaker-hearer goals, hearer background, hearer type, speaker
3. Morphology information, such as root, prefix, suffix.

4. Speech features, such as pronunciation, prosodic features, temporal information (such as duration, start, end), and non-lexical features (such as click, cough).

7 Conclusion and Future work

In this paper, a new CTS architecture is presented. The key idea is to integrate current NLG and TTS systems in a standard way so that the CTS system developed is able to use any existing NLG tools, STP algorithms and TTS systems and benefit from the information available from NLG. A Speech Integrating Markup Language is designed for this purpose.

In the future, we will extend our STP algorithms, to predict an adjustable range of speaking rate and stress placement based on discourse, semantic and syntactic information. As a result, we need to extend our SIML so that new information can be incorporated easily.

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