Knowledge versus data in TTS: evaluation of a continuum of synthesis systems

Citation for published version:

Published In:
INTERSPEECH 2015, 16th Annual Conference of the International Speech Communication Association, Dresden, Germany, September 6-10, 2015

General rights
Copyright for the publications made accessible via the Edinburgh Research Explorer is retained by the author(s) and / or other copyright owners and it is a condition of accessing these publications that users recognise and abide by the legal requirements associated with these rights.

Take down policy
The University of Edinburgh has made every reasonable effort to ensure that Edinburgh Research Explorer content complies with UK legislation. If you believe that the public display of this file breaches copyright please contact openaccess@ed.ac.uk providing details, and we will remove access to the work immediately and investigate your claim.
Knowledge versus data in TTS: evaluation of a continuum of synthesis systems

Rosie Kay¹, Oliver Watts¹, Roberto Barra Chicote², Cassie Mayo¹

¹Centre for Speech Technology Research, University of Edinburgh, UK
²Speech Technology Group, Universidad Politécnica de Madrid, Spain
rosiekay@gmail.com, owatts@inf.ed.ac.uk, barra@die.upm.es, catherin@inf.ed.ac.uk

Abstract

Grapheme-based models have been proposed for both ASR and TTS as a way of circumventing the lack of expert-compiled pronunciation lexicons in under-resourced languages. It is a common observation that this should work well in languages employing orthographies with a transparent letter-to-phoneme relationship, such as Spanish. Our experience has shown, however, that there is still a significant difference in intelligibility between grapheme-based systems and conventional ones for this language. This paper explores the contribution of different levels of linguistic annotation to system intelligibility, and the trade-off between those levels and the quantity of data used for training. Ten systems spaced across these two continua of knowledge and data were subjectively evaluated for intelligibility.

Index Terms: text-to-speech, speech synthesis, under-resourced languages, letter-to-sound conversion, grapheme-based acoustic modelling

1. Introduction

Grapheme-based models have been proposed for both automatic speech recognition (ASR) [1, 2, 3] and text-to-speech (TTS) [4, 5, 6] as a way of circumventing the lack of expert-compiled pronunciation lexicons in under-resourced languages.

A primary goal of our recent work [7, 8, 9, 10] has been to produce freely available tools for building statistical parametric TTS systems with little or no expert supervision. The text-processing modules we have developed are designed to construct a TTS front-end which makes as few implicit assumptions about the target language as possible, and which can be configured with minimal effort and expert knowledge to suit arbitrary new target languages. A key point of these systems is that they operate directly on a Unicode representation of the surface forms of words. It is often observed in such work that this strategy works adequately in a language such as Spanish, which employs an orthography with an approximately one-to-one letter-to-phoneme mapping.

Our experience has shown, however, that there is still a significant difference in intelligibility between grapheme-based systems and conventional ones, even in this language. Specifically, we conducted a (hitherto unpublished) experiment using data from the Albayzín 2012 TTS Challenge; this experiment was run in order to obtain a more tightly-controlled comparison of our unsupervised letter-based system [9] and a more conventional system from the challenge than had been done in the official challenge evaluation. Tighter control took the form of using identical acoustic features and voice-building recipes for both systems which mean that it was now possible to attribute the difference in performance between systems purely to the difference in system front-ends. Also, the systems were evaluated for intelligibility (unlike in the official challenge evaluation); this was done by having human listeners transcribe speech produced by the systems and using the known text to compute word error rates (WER) of the resulting transcripts.

The difference in WERs between the unsupervised and topline systems was large (8.2% absolute: the system scores were 46.8% and 36.6% respectively) and found to be statistically significant (with $\alpha = 0.05$) using the bootstrap procedure of [11]. The work presented here is motivated by these findings. This paper explores the contribution of different levels of linguistic annotation to system intelligibility, and the trade-off between those levels and the quantity of data used for training. Ten systems spaced across these two continua of knowledge and data were subjectively evaluated for intelligibility. These evaluations are designed to test 2 hypotheses: firstly, that the system using most data and most linguistic knowledge will be most intelligible. Secondly, however, we hypothesise that additional training data can compensate for a lack of linguistic knowledge.

2. Systems built

2.1. Data

The Spanish part of the Tundra corpus [12] was used for training all voices. This data is from an audiobook recording of Don Quijote and consists of eight hours of utterance-aligned speech in total. To deal with the variability inherent in audiobook data the lightly-supervised data selection method described in [7] was used to remove the least neutral 20% of the data. A chapter of the resulting data (119 sentences) was then held out for testing, leaving just over five hours of data for training.

To test the effect of varying amounts of training data, and the interaction of this variation with the variation in amount of linguistic knowledge given to the system, a small 1 hour set and a large 5 hour set were prepared from the training data.

2.2. Front-end annotation

Two sets of annotation were prepared for the data: one naive one based on surface orthographic forms (letters) and one based on linguistic knowledge.

For the naive annotation, the audiobook text was converted to lowercase and non-ASCII characters were substituted with ASCII-safe replacements. Whitespace was stripped and punctuation was replaced with a silence marker. Context-dependent labels were prepared from this processed text, in which each letter is characterised by the identities of the letters occurring in a five-letter window surrounding it. No further features were used (in contrast to e.g. [7] where positional and vector space model features were used).

The knowledge-based annotation was obtained from a conventional Spanish front-end which uses rule-based mod-
Table 1: Five levels of the knowledge continuum chosen for evaluation. Each level from P+ to F makes use of knowledge at all previous levels, excluding L; letter-based knowledge is specific to the letter-based systems. Knowledge corresponds to the standard linguistic and prosodic contexts taken into account in an HTS system, as in [14].

<table>
<thead>
<tr>
<th>Level of continuum</th>
<th>Knowledge</th>
</tr>
</thead>
<tbody>
<tr>
<td>Letters (L)</td>
<td>Five-letter context window: the system knows what the current, preceding two and following two letters are.</td>
</tr>
<tr>
<td>Phonemes (P)</td>
<td>Five-phoneme context window: the system knows what the current, preceding two and following two phonemes are.</td>
</tr>
<tr>
<td>Phonemes plus phonological class information (P+)</td>
<td>Five-phoneme context window plus additional linguistic knowledge about phonological classes, as well as place and manner of articulation e.g. vowel height, vowel length, place of articulation of consonants.</td>
</tr>
<tr>
<td>Syllable (S)</td>
<td>Syllable features as in [14]</td>
</tr>
<tr>
<td>Full (F)</td>
<td>Utterance features in [14]</td>
</tr>
</tbody>
</table>

Table 2: Identifiers for 10 voices evaluated in the subjective listening test.

<table>
<thead>
<tr>
<th>Knowledge</th>
<th>1 hour</th>
<th>5 hours</th>
</tr>
</thead>
<tbody>
<tr>
<td>Letters</td>
<td>L_60</td>
<td>L_300</td>
</tr>
<tr>
<td>Phonemes</td>
<td>P_60</td>
<td>P_300</td>
</tr>
<tr>
<td>Articulatory information</td>
<td>P+_60</td>
<td>P+_300</td>
</tr>
<tr>
<td>Syllables</td>
<td>S_60</td>
<td>S_300</td>
</tr>
<tr>
<td>Full</td>
<td>F_60</td>
<td>F_300</td>
</tr>
</tbody>
</table>

3.1. Methodology

Table 2 lists the 10 voices subjectively evaluated. They represent all combinations of 5 levels along the knowledge continuum and 2 levels along the data continuum. The initial portion of the system identifiers indicates amount of knowledge used, as in Table 1. The final portion of the identifiers indicates the amount of data used, in minutes.

3.2. Results

Figure 1 shows average WER for all voices, across all listeners. The letter systems perform worse than all other levels of the knowledge continuum, and there is a clear effect of training set size: all the five-hour voices (except L_300) outperform all the one-hour voices. More interesting are the relative differences between levels of the knowledge continuum: the differences appear to be more pronounced in the one-hour voices compared to 4http://www.cs.columbia.edu/~hgs/audio/harvard.html
the five-hour voices. That is, additional linguistic knowledge appears to be less important when a large amount of training data is available, which supports the hypothesis that additional training data can compensate for a lack of linguistic knowledge.

Results do not support the hypothesis that F_300 would be the most intelligible. That the F voice was not the most intelligible voice is replicated at both levels of the data continuum, with tight error bars in all cases, suggesting it is not just an anomaly. In both cases, syllable, phrase and utterance level-knowledge appears to deteriorate system intelligibility. One possible explanation for this is that the decision tree clustering may have split the data too much, leading to sparsity problems. Another possibility is that there are issues with the front-end, and specifically its prediction modules for labelling the training data.

Finally, all WERs (except for the letter systems) are quite low. Due to the use of non-SUS, the sentences may have been too predictable, leading to a flooring effect. However, the sizeable gap between letter and phoneme systems suggests this cannot entirely be the case: if the sentences were entirely predictable, similarly low WERs would be expected.

Table 3 shows the results of paired Wilcoxon signed-rank tests on all voices within each level of the data continuum. This supports the hypothesis that there are greater differences between the one-hour voices than the five-hour voices, as almost all of the one-hour voices are significantly different from each other. For the five-hour voices, all voices are significantly different from the letter system, but there is only one other significant difference (between F and P+).

3.3. Post-hoc analysis: types of errors

In total, 2692 response utterances contained errors, with between one and six errors per error-containing-utterance. It was not possible to conduct a full analysis of all errors for all voices. Instead, three one-hour voices were chosen and 200 errors from each, selected at random, were examined in detail, to investigate whether the types of errors differ across voices. Explored in more detail were L_60 (the least intelligible voice), P+_60 (the best of the one-hour voices) and F_60 (the expected best one-hour voice). Three error categories were defined: replaced sound (one phoneme replaced, e.g. los → las), replaced word (more than one phoneme replaced e.g. claro → blando), and omitted word (participant made no attempt to type the word). Only three insertion errors were seen throughout the entire analysis so this is disregarded as an error category for current purposes.

The percentage of each type of error for each voice is listed in Table 4. Most of the errors for L_60 are omissions of words altogether, rather than an incorrect guess at a word. Conversely, far fewer of the errors for the P+ and F voices are omissions: it is more likely that a word is detected, but one or more phonemes are incorrectly heard. This suggests a less serious failing on the system’s part, particularly where only one phoneme is incorrect. Indeed, it seems reasonable to think of types of errors in a hierarchy, with omitted word errors being of the highest magnitude, and replaced sound errors of the lowest. It is clear, then, that the letter-based voice does worse, not only in overall WER, but also in that errors tend to be more serious: when synthesis fails, it is often so bad that the listener does not attempt to guess what is heard.

One such omission comes up repeatedly for L_60: 14% of omission errors are of the word ‘y’ (meaning ‘and’), and there are many other instances of monosyllabic words such as ‘hay’ ‘un’ and ‘la’ being omitted by listeners. Listening to synthesised utterances in which ‘y’ was omitted, there is silence where ‘y’ should be (see Figure 2). This strongly suggests the system failed to learn a good model for the letter ‘y’ (which alternates between a vowel and a consonant, depending on context: here, a vowel). This is one example where a phoneme-based system has a clear advantage, as such alternations do not have to be learned, but are already known, assuming correctly labelled data.

Table 4: Types of error for each voice, based on samples of 200 errors selected at random per voice.
4. Conclusion

Our results suggest that a linguistically naive Spanish system can achieve a reasonable level of intelligibility, particularly when sufficient data is available. A WER of 5.2% for L_{300} is encouraging, although perhaps could have been expected to be slightly higher, given the good letter-to-phoneme correspondence in Spanish.

Although the L systems were the least intelligible, additional training data improves the situation markedly: the performance gap between L_{300} and P_{300} was smaller than that of L_{60} and P_{60}. That L_{300} is much more intelligible than L_{60} suggests the larger voice has done a better job at learning alternations, as many of the errors for L_{60} were due to problems with these. Examining some synthesized utterances suggested these were still causing some problems for L_{300}, but the situation was much improved compared to the one-hour system. With five times more data, the errors are reduced by approximately a third, thus, one can tentatively consider how much data might be needed before the performance begins to match that of a phoneme-based voice, if ever.

The question of whether additional training data can compensate for a loss of linguistic knowledge certainly seems to have been confirmed: the differences between levels of the knowledge continuum were less prominent for the five-hour voices than for the one-hour voices, with fewer significant differences between different five-hour voices. This closing of performance gap when more data is used strongly suggests a larger training set can compensate for less linguistic knowledge. This is probably the most significant finding in light of the ongoing research into improving synthesis with limited resources, as training data is easier and cheaper to source than expert linguistic knowledge. It is highly encouraging that with a large enough training set, intelligible synthesis can be achieved by a linguistically naive system: this dramatically reduces the bottleneck to providing TTS in new languages. Adding low level phonological features significantly improves intelligibility over a letter-based voice, but in the absence of the necessary resources to be able to develop a phoneme set for a new language and build the necessary text-processing modules, it is promising that simply adding more hours of training data can compensate to a certain extent.

5. Acknowledgements

The research leading to these results has received funding from the European Community’s Seventh Framework Programme (FP7/2007-2013) under grant agreement No 287678 (Simple4All).

6. References


