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Synthesis of Child Speech with HMM Adaptation and Voice Conversion

Oliver Watts, Junichi Yamagishi, Simon King, Senior Member, IEEE, and Kay Berkling, Senior Member, IEEE.

Abstract—The synthesis of child speech presents challenges both in the collection of data and in the building of a synthesiser from that data. We chose to build a statistical parametric synthesiser using the HMM-based system HTS, as this technique has previously been shown to perform well for limited amounts of data, and for data collected under imperfect conditions. Six different configurations of the synthesiser were compared, using both speaker-dependent and speaker-adaptive modelling techniques, and using varying amounts of data. For comparison with HMM adaptation, techniques from voice conversion were used to transform existing synthesisers to the characteristics of the target speaker. Speaker-adaptive voices generally outperformed child speaker-dependent voices in the evaluation. HMM adaptation outperformed voice conversion style techniques when using the full target speaker corpus; with fewer adaptation data, however, no significant listener preference for either HMM adaptation or voice conversion methods was found.

Index Terms—Speech synthesis, Hidden Markov models, Children.

I. INTRODUCTION

The synthesis of child speech presents special difficulties for data-driven speech synthesis systems due to the type of child speech corpus typically available. Unit selection speech synthesis (e.g. [3]), which has come to be the dominant approach to data-driven speech synthesis over the last decade, produces waveforms for arbitrary novel utterances by directly reusing existing sections of waveform from a database. One of the major strengths of this approach is that in ideal conditions, natural waveforms are reused directly with no need for manipulation of spectrum or fundamental frequency and the degradation of speech quality that this manipulation entails. However, some of the major drawbacks of unit selection stem from the same source: if the database is imperfect, this will have a direct impact on the quality of the speech synthesised. Imperfections include speaker inconsistency, background noise and poor phonetic coverage, all problems typically associated with available child speech data.

Statistical parametric approaches to speech synthesis (such as HMM-based speech synthesis) have grown in popularity over the last few years [4]. These approaches have been shown to be less sensitive than unit selection to imperfect training data [5]. In HMM-based speech synthesis, use of parameter sharing techniques allows the synthesis of models for speech units unseen in the training corpus; this contrasts with the corresponding strategy that must be used in unit selection where the system must select a substitute unit, typically on the basis of heuristics. Unlike unit selection synthesis, already-trained HMM-based systems can be adapted to the voice characteristics of a target speaker with small amounts of adaptation data. The fact that it is possible to use HMMs that have been trained on cleanly recorded data, rich in phonetic contexts, as the basis for adaptation means that high-quality speech can be synthesised even when the adaptation data is noisy and sparse.

Adaptation has been used to impose various types of characteristics onto existing statistical parametric synthesisers, for example, characteristics associated with dialect [6] and speaking style [7]. In [1] we applied adaptation techniques (among others) to the creation of what (to our knowledge) is the first data-driven synthesiser of child speech. We present this work here with fuller analysis and with a comparison between HMM adaptation techniques and techniques from voice conversion for the transformation of an existing synthesiser to a child speaker.

This paper is organised in the following way. Section II describes a corpus of child speech data collected especially for this research and compares its suitability for use in training speech synthesisers with that of a purpose-built speech synthesis corpus. Section III describes the systems built with the child speaker as target speaker, and Section IV reports evaluation of these systems. We conclude the paper by briefly summarising our findings in Section V.

II. CHILD SPEECH DATA

The training of data-driven synthesis systems for child voices presents difficulties due to the type of data which it is generally feasible to collect from child speakers. Data-driven speech synthesisers are conventionally trained on corpora that are phonetically balanced, consistently read, and cleanly recorded. A good example of such a corpus is the CMU Arctic database, designed and recorded by several adult speakers specifically for the purpose of speech synthesis [8]. The type of child speech typically available, on the other hand, more closely resembles ‘found’ data in that it does not give good
coverage of the phonetic/prosodic units of the language, is inconsistently read, and is imperfectly recorded. An example of such a child speech corpus is the one collected and prepared in the work described here. A comparison of this corpus with one part of the CMU Arctic database (the data of speaker SLT, which will henceforth be called as SLT) will be made in order to highlight the problems inherent in the sort of child speech data corpus typically available. We begin with a brief overview of the collection of what will be referred to as the Child corpus before moving on to make the comparison with the more conventional database.

A. Child Database: Overview

The North American-accented English speech of a 7-year-old tri-lingual (Spanish, English, German) female was collected using a headset microphone in an informal setting at the home of one of the authors over the course of several months. Consequently, the recordings contain considerable non-speech sounds, including the sounds of hammering, page turning, traffic and wildlife, and reverberation. With SLT, which was recorded in a purpose-built studio. As noted above, the Child corpus – on the other hand – was recorded using a headset microphone in the home of one of the authors: it is more difficult to get a child into the studio than a paid voice talent. Consequently, the recordings contain considerable background noise, including the sounds of traffic and wildlife, and reverberation. The speech was collected without interruption and so utterances were interspersed with sighs, page turns, and other non-speech sounds which it was not possible in all instances to remove without also excising valuable speech data. Figure 1 shows spectrograms of three short excerpts from the recordings showing background noise fairly typical of these data.

C. Speaker Variability

Children do not typically have the vocal and emotional control necessary to minimise inconsistency during and between recording sessions. Use of a coherent ‘script’ (storybook text) enjoyable to the speaker increases the effects of emotional engagement with what is being read, which may be problematic from a speech synthesis perspective. In the present case, it led at times to fluctuations in speech quality and a variable reading style very different from that which can be heard in the SLT data, with the child speaker screaming, singing, and chanting at appropriate points in the story and deliberately altering voice quality to act out characters’ parts. Children can only be persuaded to record data in short sessions, with the result that the recordings of the Child corpus were made over several months, an additional cause of inter-session variation.

Artistic engagement with the recording script and short recording sessions aside, a high degree of innate variability is a well-known characteristic of children’s speech, in comparison with the speech of adults (e.g. [9]). Fig. 2 plots of $F_0$ values, power and duration of a single phone type (/aa/) from the Child and SLT databases. It can be seen that not only does the child speech have generally higher fundamental frequency values, lower power, and longer duration values than that of the adult, but all these factors have wider ranges.

D. Database coverage

An established way to create prompts for speech synthesis databases is to select utterances from a large corpus of text according to some criterion of phonetic coverage (e.g. [8]). Recording scripts resulting from this type of procedure, while phonetically well-balanced, are not coherent texts that a seven-year-old child could be persuaded to read. It is more feasible to use, for example, story books familiar to the child, as was done in the work presented here. This results in a corpus with comparatively poor phonetic coverage. Table I gives figures for triphone and quinphone coverage of four corpora. The information given for each corpus includes number of tokens, number of triphone and quinphone types along with Type-Token Ratios (TTR) for triphones and quinphones. TTRs are a measure of the richness of the content of the corpora, a TTR of 1.0 indicating that a each token of a corpus is of a different type, thus representing the maximum coverage a corpus of that size could provide. The first two corpora are...
Fig. 2. Plots of $F_0$, power and duration of phone /aa/ for Child and SLT corpora. Medians are shown as red bars across boxes indicating quartiles, and whiskers extend to 1.5 times the inter-quartile range. It can be seen that not only does the child speech have generally higher fundamental frequency values, lower power, and longer duration values than that of the adult, but all these factors have wider ranges.

TABLE I


(a) All types/tokens counted

<table>
<thead>
<tr>
<th>Database</th>
<th>Child</th>
<th>SLT</th>
<th>SLT2</th>
<th>Snark</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tokens</td>
<td>37707</td>
<td>38866</td>
<td>37707</td>
<td>37707</td>
</tr>
<tr>
<td>Triphone types (TTR)</td>
<td>6376 (0.17)</td>
<td>9546 (0.25)</td>
<td>9430 (0.25)</td>
<td>8165 (0.22)</td>
</tr>
<tr>
<td>Quinphone types (TTR)</td>
<td>20478 (0.54)</td>
<td>28662 (0.74)</td>
<td>27912 (0.74)</td>
<td>26601 (0.71)</td>
</tr>
</tbody>
</table>

(b) Types/tokens containing pause not counted

<table>
<thead>
<tr>
<th>Database</th>
<th>Child</th>
<th>SLT</th>
<th>SLT2</th>
<th>Snark</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tokens</td>
<td>18555</td>
<td>30377</td>
<td>18555</td>
<td>18555</td>
</tr>
<tr>
<td>Triphone types (TTR)</td>
<td>4773 (0.26)</td>
<td>8482 (0.28)</td>
<td>6676 (0.36)</td>
<td>5775 (0.31)</td>
</tr>
<tr>
<td>Quinphone types (TTR)</td>
<td>11246 (0.61)</td>
<td>24035 (0.79)</td>
<td>15665 (0.84)</td>
<td>14531 (0.78)</td>
</tr>
</tbody>
</table>

the Child corpus and SLT. SLT2 denotes a large subset taken from the beginning of the SLT corpus which contains the same number of tokens as the Child corpus. This subset was included because corpus size influences the magnitude of type-token ratios; a subset of the same length, however, allows fair comparison of TTRs. Finally, Snark is a corpus compiled especially for this analysis. A story (Jack London’s Voyage of the Snark) from the same domain as the Arctic texts (and in fact included among them) was split into sentences, which in turn were turned into linguistic specifications with the front end of the Festival synthesiser and finally context-dependent labels. A subset was taken from the beginning having the same number of tokens as the Child corpus. The Snark corpus was assembled to provide a midpoint for comparison: the fact that it was taken from the same domain as Arctic but that utterances were taken sequentially from the story rather than being selected according to a criterion of phonetic coverage means that the figures for this corpus allow us to assess in isolation the influence of the selection procedure on the coverage of the resulting corpora.

The top half of Table I gives counts and ratios over all units in the corpora. It was noted that Child contains a much greater number of pauses (represented by the ’phone’ pau) than the other databases, including a great number of consecutive pause tokens due to relatively long hesitations and disfluencies. Although these pauses are one of the factors which strongly characterise the Child corpus, it was thought informative to compute similar information, discounting tokens containing the pau token (either as central phone or any of the four possible left and right contexts). This information is given in
The coverage statistics computed here (triphones and quinphones) reflect some of the contextual factors considered during HMM-based voice building (see Section III below). There are a great many other phonetic, prosodic and syntactic factors considered during voice building that have been ignored in this analysis. Note also that although the criterion used in the construction of the SLT corpus (diphone coverage) is related only indirectly to triphone and quinphone coverage, the triphone and quinphone coverage are improved by prompt selection using the diphone criterion. That is, both when pauses are counted and when they are ignored, for both triphone and quinphone, SLT2’s TTRs are higher than Snark’s. The Child database has lower TTRs than those of either SLT2 or Snark in all cases. That is, corpora collected using children’s stories has worse coverage not only than the purpose-built database which exemplify the sort of repetition found extensively in these children’s stories:  

(1) She knew how to set up a tent, she knew how to build a camp fire, she knew how to cook camp food.  
(2) The car began to roll. Faster and faster and faster.

This section has outlined some of the differences between the target speaker data available to us and a conventional speech synthesis database. Despite the imperfect nature of the Child corpus, we wished to produce a text-to-speech system with the voice characteristics of the child target speaker. The measures taken to overcome the difficulties presented by the data consisted firstly of especially careful preparation of the data and secondly of choice of synthesis methods appropriate to this data. We conclude Section II with an account of the front-end processing involved in preparing the Child corpus before turning to synthesis methods employed in the following section.

### E. Data preparation

The data were recorded without interruption and so had to be split into shorter fragments in order to exclude disfluencies, screaming, singing, sighs, page turns, and other non-speech sounds. We did not attempt to incorporate these elements into the synthetic voice. The data were hand-transcribed in standard orthography. Special care was taken to deal with mispronunciations and word-fragments in such a way that the final phonetic transcription would accurately reflect the contents of the audio files. Where there was a word in the lexicon that matched the speaker’s mispronunciation, this word was used in the transcription (e.g. the speaker often read ‘cells’ as ‘seals’, and so the second word was used in the transcription). Where there was no existing lexical item to match the speaker’s pronunciation of a word or fragment, an invented word was used in the normal spelling transcription, and then this invented word was added to the lexicon with the speaker’s pronunciation before the phonetic transcription was generated.

A phone transcription was produced for the rest of the data with the Multisyn voice-building tools [3]. An initial phone transcription was produced by performing lexical look-up from the augmented lexicon. This initial transcription was then refined by forced alignment with the audio, in which vowel reduction and the insertion of pauses between words are allowed where supported by the audio data. Pause insertion is particularly important in the case of such hesitantly read data.

Table II lists the 10 most common quinphones (excluding quinphones containing the pau token) of Child, SLT2 and Snark, which give clues about the reasons for the worse coverage of Child. Eight of the SLT2 quinphones come from sequences of function words or function words and verbs (‘it was’, ‘there was’, ‘he was’, ‘he had’ etc.); only 1 comes from a person’s name (‘Phillip’). On the other hand, 7 of the Child quinphones come from people’s names (‘Pickle’, ‘Mrs’, ‘Dragon’, ‘Christy’, ‘Greg’), and only 1 quinphone from a function word–verb sequence (‘he said’) occurs among the most frequent ten. One of the reasons for Child’s relatively poor phonetic coverage, then, is its repetition of proper names. We note that repetitiveness is also seen more generally in the corpus. Note the following utterances from the corpus which exemplify the sort of repetition found extensively in these children’s stories:

(1) She knew how to set up a tent, she knew how to build a camp fire, she knew how to cook camp food.  
(2) The car began to roll. Faster and faster and faster.

### III. THE SYSTEMS

#### A. Speaker Dependent & Speaker Adapted HTS Voices

HMM-based synthesis was our chosen method for building voices because of its robustness to imperfect recording conditions [5], its integrated data-driven method for synthesising
IDENTIFYING LETTERS USED FOR EACH SYSTEM. TRANSFORMATIONS ARE TO THE TARGET SPEAKER IN ALL CASES EXCEPT FOR THE DURATION ADAPTATION OF SYSTEM I. HMM: HMM ADAPTATION (CSMA/LR + MAP), VC: VOICE CONVERSION. FIRST GROUP (A–F) IS DESIGNED FOR COMPARISON OF SPEAKER-DEPENDENT AND SPEAKER-ADAPTIVE TECHNIQUES WITH 3 TARGET SPEAKER CORPUS SIZES, THE SECOND GROUP IS DESIGNED FOR COMPARISON OF VOICE CONVERTED UNIT SELECTION SYSTEMS WITH HMM BASED ONES AND TO ASSESS THE IMPACT TRANSFORMING DURATION TO TARGET SPEAKER IN BOTH CASES, AND THE LAST TWO GROUPS ARE DESIGNED FOR MORE CONTROLLED COMPARISON OF HMM ADAPTATION AND VOICE CONVERSION METHODS WITH 2 SIZES OF TARGET SPEAKER CORPUS.

<table>
<thead>
<tr>
<th>ID</th>
<th>Type</th>
<th>Base voice speaker(s)</th>
<th>Target data</th>
<th>Transformation method</th>
<th>Spec.</th>
<th>F0</th>
<th>Dur.</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>HTS</td>
<td>Child</td>
<td>15 min</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>B</td>
<td>HTS</td>
<td>6 adults</td>
<td>15 min</td>
<td>HMM HMM HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>C</td>
<td>HTS</td>
<td>Child</td>
<td>30 min</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>D</td>
<td>HTS</td>
<td>6 adults</td>
<td>30 min</td>
<td>HMM HMM HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>E</td>
<td>HTS</td>
<td>Child</td>
<td>94 min</td>
<td>—</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>F</td>
<td>HTS</td>
<td>6 adults</td>
<td>94 min</td>
<td>HMM HMM HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>I</td>
<td>HTS</td>
<td>6 adults</td>
<td>94 min</td>
<td>HMM HMM HMM</td>
<td>—</td>
<td>—</td>
<td>— (to SLT)</td>
</tr>
<tr>
<td>J</td>
<td>Unit sel.</td>
<td>SLT</td>
<td>94 min</td>
<td>VC</td>
<td>VC</td>
<td>—</td>
<td></td>
</tr>
<tr>
<td>K</td>
<td>Unit sel.</td>
<td>SLT</td>
<td>94 min</td>
<td>VC</td>
<td>VC</td>
<td>—</td>
<td></td>
</tr>
<tr>
<td>L</td>
<td>HTS</td>
<td>SLT</td>
<td>15 min</td>
<td>HMM HMM HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>M</td>
<td>HTS</td>
<td>SLT</td>
<td>15 min</td>
<td>VC HMM HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>N</td>
<td>HTS</td>
<td>SLT</td>
<td>15 min</td>
<td>HMM VC HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>O</td>
<td>HTS</td>
<td>SLT</td>
<td>15 min</td>
<td>HMM HMM VC</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>P</td>
<td>HTS</td>
<td>SLT</td>
<td>94 min</td>
<td>HMM HMM HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>Q</td>
<td>HTS</td>
<td>SLT</td>
<td>94 min</td>
<td>VC HMM HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>R</td>
<td>HTS</td>
<td>SLT</td>
<td>94 min</td>
<td>VC HMM HMM</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>S</td>
<td>HTS</td>
<td>SLT</td>
<td>94 min</td>
<td>HMM HMM VC</td>
<td>—</td>
<td>—</td>
<td>—</td>
</tr>
<tr>
<td>G</td>
<td>Vocoder natural speech</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>H</td>
<td>Original speech</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

The rich context-dependency of the speech units results in a very large number of models. This in turn means that almost all models will be sparsely represented in the training data (typically we find just one example of each in the training data!) and that, at synthesis time, models of missing units will certainly need to be created. Both of these problems are solved by the use of decision-trees. In the construction of these trees during training, model parameters are pooled and then repeatedly divided by the application of yes-no questions relating to the contextual features that define the models (e.g. ‘Is the state part of a nasal consonant phone?’, ‘Is the state part of a phone that occurs at the end of a word?’ etc.). Questions are selected and ordered in the trees during training so that acoustically similar states end up pooled in the same leaf nodes of the trees. This solves the problem of data sparsity during training by allowing the parameters of acoustically similar states in a leaf node of the tree to be “tied” (re-estimated as a single distribution with the pooled training data). The trees solve the problem of unseen models at synthesis time by allowing the creation of these models: for each state of an unseen model, the relevant trees are traversed by answering the questions appropriately until a leaf node is reached. The probability distributions pointed to by this leaf node are then used to populate the relevant state of the unseen model.

Separate trees are made for spectral, F0 and aperiodicity measure distributions of each emitting state, and a single tree for duration is made for all states, resulting in 16 trees in the present set-up. This allows the clustering of units for spectral quality, F0, duration and aperiodicity measures with different trees using different context questions; as we would expect, used for the same HTS entry in the Blizzard Challenge. An account of these procedures is given below, beginning with details of parameter extraction and model structure common to all six systems.

1) Parameter Extraction: For both types of system built, the speech was parameterised as 40 mel-cepstral coefficients, log F0 and the energy of aperiodic components in 5 frequency bands, and the dynamic and acceleration features derived from all of these, to yield a 138-dimension observation vector for the HMMS. F0 was extracted using a three-stage procedure. First, the ESPS get_f0 tool was used to extract F0 for all the speech data. These preliminary F0 values were then plotted as a histogram, from which a rough F0 range for the speaker was determined. F0 values were then re-extracted within the determined range using a voting method based on get_f0, Tempo and IFAS, the final F0 for each frame being the median of the three extracted values for that frame. Spectral analysis was performed with the high quality vocoder STRAIGHT [13], and the STRAIGHT spectra were converted to mel-cepstral coefficients.

2) Model Structure and Context Clustering: For both types of system built, speech units were modelled with HMMS of 5 emitting states in a left-to-right topology. For state emission PDFs, single mixture component Gaussian distributions with diagonal covariance matrices were used. In all cases, the same set of units was used: phones dependent not only on neighbouring phones, but on an extensive list of phonetic, linguistic and prosodic contexts (see [14] for the list).

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Separate trees are made for spectral, F0 and aperiodicity measure distributions of each emitting state, and a single tree for duration is made for all states, resulting in 16 trees in the present set-up. This allows the clustering of units for spectral quality, F0, duration and aperiodicity measures with different trees using different context questions; as we would expect,
different aspects of context affect spectral quality than those affecting $F_0$.

Although tree-building starts with a set of contexts (or yes-no context questions) which are hand-crafted to specify the phonetic and linguistic contexts which we think will have an effect on the acoustics of speech units in a given language, tree-building itself proceeds automatically. That is, questions are selected one by one according to some criterion and added to the tree until a stopping condition is met. In the current procedure, nodes associated with context questions are added to the trees until the Minimum Description Length (MDL) criterion is met. The MDL criterion is a well-known information criterion for avoiding over-fitting of models to the training data and can specify an appropriate size for the decision tree [15].

3) Speaker Dependent Systems (A, C and E): For the speaker-dependent systems, model training began with the estimation of monophone models (phone models independent of context). These were then used as the basis for full-context models, which were re-estimated before decision-tree based context clustering was applied to spectral, $\log F_0$, aperiodicity and duration features separately. The clustered parameters were tied and re-estimated, then the procedure was repeated: parameters were untied and re-estimated, clustered and and re-estimated a second time.

4) Speaker Adaptive Systems (B, D and F): As noted above, the speaker-adaptive system adopted an already-trained gender-mixed average voice from previous work. Details of training are given in [12]. This gender-mixed average voice model was trained on the six adult speakers of CMU-ARCTIC speech database (four male, two female). First, two gender-dependent average voice models were trained using Speaker Adaptive training (SAT); that is, speaker normalisation was applied during estimation of the models, to avoid different speaker-dependent voice characteristics “diluting” the average models. Then, the parameters of both gender-dependent models were clustered and tied using decision-tree based clustering, with gender included as a context feature. Then the clustered HMMs were re-estimated using SAT, regression classes for the normalisation being determined from the gender-mixed decision-trees. State durations obtained during this estimation were used to initialise duration probability distributions which were then clustered. SAT was performed on the complete HSMMs to re-estimate all parameters (including duration) with speaker normalisation.

Adaptation of the gender-mixed average voice model was performed using data from the target speaker, the labels being modified to include target speaker gender. Adaptation was performed with a combination of constrained structural maximum a posteriori linear regression (CSMPLR) and maximum a posteriori (MAP) adaptation [15].

5) Synthesis: During preparation of the Child corpus, 30 sentences had been chosen for their fair degree of fluency and medium length (4–9 words) from across the recording sessions and held out from training and adaptation. These utterances were synthesised with Festival. Festival’s front-end performed the phonetic and linguistic predictions needed to provide a sequence of context-dependent labels for each utterance. Based on these predictions, parameters were generated using the models that had been trained, and waveforms were synthesised from those parameters.

B. Unit Selection Synthesis + Voice Conversion

We wished to explore the possibilities offered by voice conversion techniques for imposing a novel speaker’s voice characteristics on an existing synthesiser and to compare them with HMM-Based methods. As well as being applied to the conversion of natural speech, popular voice conversion techniques such as those based on spectral conversion with Gaussian mixture models (GMMs: [16],[17]) have been applied to the conversion of speech synthesisers’ output to the voice characteristics of new speakers. For example, in [18], the output of a concatenative (diphone) synthesiser is used to create ‘source speaker’ training data for a voice conversion model. Arbitrary novel utterances subsequently produced by the synthesiser can then be converted to the voice characteristics of the target speaker.

The application of voice conversion to the output of an existing speech synthesiser is particularly attractive in the context of sample-based concatenative methods where there is no statistical model whose parameters can be transformed. We therefore sought to compare the voice-converted output of a unit selection synthesiser with the results of speaker adaptation of HMM-based systems.

In GMM-based voice conversion, source speaker’s durations are typically used unconverted in the output speech; this is the case in our system J, where spectrum and $F_0$ of the synthesiser’s output were converted. This can be successful in cases where the differences between the durational characteristics of source and target speakers are negligible (as might be true of a pair of adult speakers). But in the present case the durational patterns which characterise the target speaker are very different from those of the adult speaker on whose base voice was trained. It is unreasonable to think that reusing source speaker’s durations unmodified could be successful in this case. Therefore we built a second system K where in addition to performing spectral and $F_0$ conversions, converted utterances’ durations were obtained by uniformly stretching by a pre-determined factor the utterances which had been output by the synthesiser.

1) Training (Systems J and K): The synthesiser used as the ‘source speaker’ in both systems J and K (see Table III) was an existing unit selection voice which had been built with the SLT corpus, using the Multisyn voice building tools ([3]).

Voice conversion models to map from the output of this base synthesiser to the characteristics of the child target speaker were trained using scripts distributed as part of the FestVox project, implementing techniques developed by Toda ([17], [19]). Training a Gaussian Mixture Model (GMM) for spectral conversion requires a parallel corpus in which the source and target speakers each utter the same words. The missing ‘source speaker’ half of this corpus was synthesised using the base unit selection synthesiser; the target speaker half was natural data consisting of the whole of our target speaker training data (94 min). At no point after the synthesis of this ‘source speaker’
data was knowledge of the linguistic contents of the utterances used; in training the conversion models and applying those models to arbitrary new speech it was assumed that knowledge of phone, word etc. alignment was unavailable.

The speech was parameterised using the STRAIGHT cepstral analysis and $F_0$ extraction described in Section III-A1 above rather than with the analysis tools distributed with the FestVox module. The only departure from the analysis procedure described in Section III-A1 was that a lower dimension static feature vector was used ($24 - \text{the 0th coefficient was not used for training GMMs}$). This was a result of initial work in which attempts at training joint GMMs on much higher order features failed even with few mixture components.

The conversion model for spectral features was trained as follows. The static parameters were supplemented by dynamic features and then joint feature vectors were obtained from source and target speech by alignment with Dynamic Time Warping. The parameters of a GMM (weights, means and covariance matrices for 128 mixture components) over the joint features were initialised using Vector Quantization, and then iteratively refined using Expectation Maximisation. The data alignment and GMM training were iterated.

The conversion model for $F_0$ was obtained by computing the mean and standard deviation of both source and target speakers’ $log F_0$. Additionally, for converting duration in system J, a duration scaling factor was computed as the ratio of the total duration of source speaker training data to the duration of that of the target speaker.

2) Synthesis and conversion: The sentences to be used in evaluation were taken from the story Goldilocks and the Three Bears; they were synthesised with Festival’s front-end as in Section III-A5, but this time waveform generation was performed by the concatenation of units selected from the SLT database by the Multisyn unit selection engine. The resulting waveform was then analysed in the same way that the training corpus had been. The spectral features were supplemented with dynamic features, and a sequence of single mixture component conditional probability density functions was determined from the GMM and input speech vectors using Viterbi selection. These PDFs were used to compute maximum likelihood static parameter sequences considering both static and dynamic parts of the distributions [17].

Source speaker’s $log F_0$ was converted by normalising the $log F_0$ contour using the source speaker’s mean and standard deviation, and then imposing the target speaker’s mean and standard deviation (computed during training) on the resulting contour.

Speech was then resynthesised using the source speaker’s power and aperiodicity measures unmodified together with the converted spectral features and $log F_0$.

For system K, the additional step of converting duration was performed by uniformly stretching the converted utterance in accordance with the duration scaling factor computed during training. Utterances’ duration was scaled using Pitch Synchronous Overlap and Add (Praat implementation: [20]).

3) Systems for Comparison: We intended to compare systems J and K with system F (described above), where transformation to the voice characteristics of the target speaker was performed using the same 94 min dataset but HMM-based adaptation rather than voice conversion methods. However, for system J the comparison is inexact, as system J does not impose any modification on source speaker’s duration. Therefore another system resembling system F in every way except for its duration model was constructed, system I. The only difference between systems F and I is that whereas in system F the average voice duration model is adapted to the duration characteristics of the child target speaker, in system I the same average voice duration model is adapted to the speaker characteristics of SLT.

C. Statistical Parametric Synthesis + Voice Conversion

Although systems J and K represent credible real-world configurations for voice-converted speech synthesis systems, comparison between them and systems F and I is compromised by two factors. Firstly, as noted above, lower order spectral features were used in training the voice conversion components of J and K than were used in building voices F and I. Lower order features result in lower quality resynthesis which will adversely affect the performance of systems J and K in evaluation. Secondly, the fact that different systems were based on very different underlying voices also complicates the interpretation of evaluation results: J and K were based on a unit selection voice made from the data of a single speaker, whereas systems F and I were based on statistical parametric average voice made from the data of six speakers. Both these complicating factors mean that it would be hard to assess the relative performance gains or losses due to the use of either voice conversion or HMM adaptation techniques in isolation. Systems L–S were built in order to rectify this situation.

All of systems L–S were transformed from the same underlying voice, an HMM-based speaker dependent voice trained on the data of SLT following the procedure outlined for speaker dependent voices A, C and E above, but using lower dimensional vectors of mel cepstral coefficients (25 static). The use of the same dimension of feature vector across both HMM-adapted and voice converted voices was intended to avoid the kind of bias towards system F in a comparison of systems F and K, inevitable due to the higher bit-rate vocoding used for F. The use of the same technology (statistical parametric) and training data (SLT) for the base voice was intended to remove the kind of discrepancies we would expect due to these factors not being kept constant in a comparison of systems F and K.

Furthermore, we wished to evaluate the contribution to system performance of each voice conversion component (GMM, scaling of $F_0$, stretching of duration) individually. Systems M, N and O (see Table III) were all designed to be compared with system L – in each of these comparisons, the transformation method of a single voice component (spectrum, $F_0$, duration) is switched in isolation from HMM adaptation (with CSMAPLR and MAP as above) to the corresponding voice conversion method. The same scheme was applied in systems P–S, but whereas in systems L–O the 15 min target speaker dataset was used for transformation, in systems P–S the whole target speaker set of 94 min was used.
The procedures followed for training voice conversion functions of spectrum, $F_0$ and duration and applying them to test utterances used for systems M–O and Q–S were almost identical to those used for voice K, described above. The only difference was that, both for training and conversion, the spectral and $F_0$ features output by the base voice were fed directly into the voice conversion components; these features did not need to be extracted from waveforms.

The ‘pure HTS’ voices L and P were built following the same general procedure outlined for speaker adaptive voices B, D and F above. The only differences were that lower dimension feature vectors were used, and instead of an average voice, the base voice for the adaptation was a speaker dependent voice (as already mentioned).

For all of voices L–S, aperiodicity measures were kept constant: in all cases they were generated by a model that had been adapted to the target speaker with HMM adaptation.

IV. EVALUATION

A. Evaluation of Speaker Dependent and Speaker Adaptive Systems (A–F)

1) Procedure: The evaluation of the various systems was carried out using a similar protocol to the Blizzard Challenge [21]. Included in the set of ‘participants’ (i.e. systems) were two benchmarks – natural speech and vocoded natural speech – where sentences held out of the corpus were used instead of synthesis. The vocoder we use (STRAIGHT for analysis and a mixed-excitation source-filter model for waveform generation) does degrade the signal slightly, and we wished to evaluate the effect of this on child speech. The higher $F_0$ value and higher formant frequencies of child speech, compared to adult speech, may cause spectral envelope estimation to be less accurate.

The listening test, which was conducted via a web browser under quiet laboratory conditions using headphones, consisted of 3 sections. A Latin Squares design was employed meaning that in any given section, a single listener group heard every system once, each time with a different utterance. Every system was used to synthesise every utterance once within each section. We used a total of 48 paid listeners, all native speakers of English between the ages of 18 and 25.

Listeners were asked in Section 1 to rate the similarity of each stimulus to the original speaker. Two natural reference utterances were provided, which listeners could play at any time. Listeners could also hear each stimulus as many times as they wished. A five point scale was used; the end points of the scale were described to the listeners as ‘1 – Sounds like a totally different person’ and ‘5 – Sounds like exactly the same person’. Second 2 followed the same format as the first, but this time listeners were asked to rate the naturalness of each stimulus on a 5 point scale, with end points described to the listeners as ‘1 – Completely Unnatural’ and ‘5 – Completely Natural’. In Section 3, listeners were asked to type in a transcription of each test stimulus. Normally, we would use Semantically Unpredictable Sentences for this type of test, to avoid ceiling effects on transcription accuracy. However, we felt that such sentences sounded extremely unnatural when uttered by a synthetic child voice. Additionally, we did not have natural recordings of the speaker saying such sentences. Therefore, we used sentences held out from the corpus for this part of the test. These sentences are listed in Table IV.

2) Results: The listening test data were analysed using the same statistical techniques used in the Blizzard Challenge 2007 [22], and we present results in Fig. 3. Significant differences between systems are presented in Fig. 4. The differences in the results for all three sections are measured by the same test used in the Blizzard Challenge 2007: a Wilcoxon signed rank test with $\alpha = 0.01$ and Bonferroni correction. WER was computed from a set of sentences of differing lengths, necessitated by the fact that these were naturally occurring sentences ‘harvested’ from the recordings rather than generated specifically for the evaluation. This had an unfortunate consequence: the within-subjects design of the Wilcoxon test used meant that significant differences between systems for WER had to be based on scores for each listener for each system already normalised for word length. However, it was not thought that the sentences vary greatly enough in length that the outcome of the significance test for WER would be seriously affected by this.

There are several trends observable in Fig. 3 which receiving partial support from the significance test. In most cases increasing the amount of training or adaptation data gives a higher median score in sections 1 and 2 and a lower mean WER in section 3 between systems of the same type, as we would expect.

In most cases, a speaker-adaptive voice yields higher median opinion scores and lower mean WER than a speaker-dependent voice trained on the same amount of data. This is a trend that we would expect in the light of previous research showing that adaptation of an average voice with a few minutes of target speaker data results in more natural synthetic speech than the training from scratch of a speaker dependent voice on a larger dataset. The phenomenon is observed in [12] and [23], and attributed to the fact that the average voice can be built from data covering a greater variety of contexts than is available for any single speaker. It should be noted that in the present case, the average voice was trained on very different speakers (adults) to our target speaker (a child), and yet the same result appears to hold. Despite speaker differences, the average
Similarity to original speaker

Natural speech
Vocoded speech
Speaker-adaptive HMMs
Speaker-dependent HMMs

94 mins 30 mins 15 mins

A B C D E F G H

Fig. 3. Listening test results. Boxplot format follows [1]: “the median is represented by a solid bar across a box showing the quartiles; whiskers extend to 1.5 times the inter-quartile range and outliers beyond this are represented as circles.”

Word error rate in intelligibility test

Similarity to original speaker

Natural speech
Vocoded speech
Speaker-adaptive HMMs
Speaker-dependent HMMs

94 mins 30 mins 15 mins

A B C D E F G H

Fig. 4. Results of pairwise Wilcoxon signed rank tests between systems; a black square shows a significant difference between systems with $\alpha = 0.01$ (with Bonferroni correction).

to the two trends mentioned above in the case of section 1 of the evaluation. When the amount of data is increased from 30 to 94 minutes in this section, the median similarity of the speaker-dependent voice to the original speaker increases whereas the median for the average voice-based system decreases slightly. This suggests that improvements in similarity to the original speaker achieved by increasing the size of the dataset are smaller when performing adaptation than when training speaker-dependent voices. Similarity to the original speaker is perhaps the aspect of the speaker-adaptive approach that needs the most improvement.

We note that previous work has indicated that 100 utterances (approximately 6 minutes) of adaptation data are enough to adapt an average voice to the characteristics of a target speaker [23]. Figure 3, however, shows that 15 minutes of child data achieve a median opinion score of only 2. We attribute this low score to the noisiness of the adaptation data and to the fact that the transformation attempted was unusually extreme: in [23], an average adult voice was adapted to another adult speaker, and not to a child.

In the evaluation of naturalness, the natural vocoded speech received a median opinion score of one point less than that of the original speech, and in the evaluation for intelligibility, it received a higher mean WER. These scores suggest that vocoding alone is causing degradation of the speech signal. Whether this degradation in quality (greater than we expected) is specific to child speech could be the subject of useful future research.

In the evaluation of similarity to the original speaker, even the natural speech received a median opinion score of 4, where we would expect 5. This might be attributed to the variability of the child speech data: the two natural speech samples given for reference in the evaluation were taken from different recording sessions and have slightly different qualities. The synthetic speech in effect “averages out” the speaker/recording condition variability across all the data, and as such is different in quality from either of the two natural samples.
B. Evaluation of Unit Selection + Voice Conversion (J and K)

1) Procedure: An XAB test was conducted in which a pairwise comparison was made of the four systems in terms of the similarity of the synthetic speech to the natural speech of the target speaker. Four reference sentences spoken by the target speaker which had been held out of the training corpus were analysed and resynthesised as described in Section III-A1 above with no manipulation of the features. They were presented at the beginning of the evaluation and at intervals throughout it as X, and listeners could listen to the samples as much as they wanted. The ten ‘Goldilocks’ sentences were synthesised with each of the four systems, and for each sentence an AB pair (randomly ordered) was made for each pair of systems, resulting in 60 AB pairs. The listening test was conducted via a web browser, with a total of 10 unpaid listeners. The 60 pairs were presented in random order and listeners were asked to choose the sentence in which the synthetic speech’s speaker characteristics were most similar to those of the natural reference samples.

2) Results: Fig. 5 shows the results of the evaluation. Significant preferences were detected for all pairs of systems except I vs. K.

The evaluation shows that the HMM-based systems were generally preferred as more similar to the original speaker than the voice converted unit selection systems. However, interpretation of these preferences for HMM adapted over voice converted systems is complicated by the factors outlined in Section III above.

Both between HMM adapted systems F and I and between voice converted systems J and K we found significant preference for the system where duration transformation to the target speaker was performed (systems F and K respectively). We note that the addition of duration transformation to the HMM adapted system (F vs I) leads to a more extreme preference than in the case where uniform duration stretching is added to other voice converted system components (J vs K). In previous evaluations of HMM adapted voices [23], the inclusion of adaptation for duration leads to improved performance in subjective evaluation, but the preference in the present work for system F over I is greater than that previous work led us to expect. We attribute the strength of this preference to the fact that speech duration patterns are very important in characterising the child voice; these patterns distinguish a child’s voice from an adult’s to a much greater extent than they distinguish the speech of an adult from that of another adult speaker.

C. Evaluation of Individual Voice Conversion Components (Systems L–S)

1) Procedure: Evaluation procedure was the same as that outlined in Section IV-B; here also, 6 pairwise comparisons were made, but this time the comparisons each focused on the impact of a single voice conversion component on system performance. Two slight divergences from the procedure in Section IV-B were made: the natural reference samples were original waveforms, not vocoded speech, and 9 listeners performed the evaluation, not 10.

2) Results: Fig. 6 shows the results of the evaluation. In all the voices transformed with the small (15 min) dataset (L–O), no significant difference is detected between transformation components based on HMM adaptation and those based on voice conversion methods (GMM and uniform scaling.
of log $F_0$ and duration). However, when 94 min of data is available (P–S), there is a significant preference for the HMM adaptation technique in every case. These findings are consistent with our expectations: given sufficient data we would expect the HMM adaptation techniques to give better results due to the fact that the decision tree used incorporates high-level linguistic and prosodic information. It is also our experience that the performance shallow voice conversion methods – informed by acoustic features only – degrades slowly as the amounts of data training data available becomes very small. Voice conversion systems can be trained using the techniques tested here with as few as 30 target speaker utterances and still perform respectively. An interesting topic for future work would be to make similar comparisons between HMM adaptation and voice conversion methods as those outlined here, but with target speaker datasets smaller than 15 minutes.

V. CONCLUSION

This paper has described the application of existing HMM-based speech systems to the synthesis of a child’s speech. Both speaker-dependent voices and speaker-adapted voices were built. Additionally, we have compared the performance of systems transformed to the child target speaker with HMM adaptation techniques to that of those where one or more components of the voice are transformed using techniques from voice conversion.

It was found that adult average voices adapted to the target child speaker data generally outperformed child speaker-dependent voices in the evaluation. This had been found to be true in the case of adult target speakers in previous work. However, we found that more target speaker data were needed to get reasonable speaker similarity ratings when adapting to the child target speaker than in the previous work with adult target speakers. Also, speaker similarity ratings level off with the largest size of target speaker database, and better ratings are obtained for the speaker-dependent system.

The adaptation of durational characteristics was found to have a greater impact on listener preference than we were led to expect from previous work on adaptation to adult speakers. Even the use of naive uniform stretching employed in the voice conversion systems led to significant preference over systems where it was absent.

In our comparison of the effectiveness of HMM adaptation techniques with voice conversion style techniques for imposing target speaker characteristics on a base voice, HMM adaptation was preferred in every case when using the full target speaker characteristics on a base voice, HMM adaptation techniques with voice conversion style techniques for imposing components of the voice are transformed using techniques from voice conversion.

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