

Towards improved children automatic speech recognition

Thesis

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Abstract

Speech is a fundamental communication skill used for social interaction, express feelings and needs, among others. Unfortunately, there is an increasing number of people who suffer from debilitating speech pathologies, including children. A childhood speech disorder that is not properly diagnosed and treated can have long-term negative effects in social, communication and educational situations. Hence the importance of speech therapy.

In recent years, machine learning has proven to have many applications in the health field, both for diagnosis and monitoring. Particularly, there is a growing interest in the development of automatic tools to assist speech-language pathologists. Furthermore, some automatic tools allow patients to do exercises outside the session with the therapist, for example at home. In the case of children, the exercises can be provided in a gamification context, which contributes to motivate them to practice more often. Most of these tools require reliable automatic speech recognition systems, as these are commonly used to provide pronunciation quality scores. However, despite recent significant improvements in the performance of automatic speech recognition systems for healthy adults, these systems' performance dramatically drops when considering speech from children and speakers with speech pathologies. This is mainly due to the high acoustic variability and the reduced amount of training data available.

In this proposal, we propose to investigate how machine learning algorithms in speech recognition tools for children can enable more complex exercises and better feedback. We first present a summary of existing automatic speech recognition techniques for children. In view of the need for a robust oracle model for children, this proposal will mainly focus on improving the automatic speech recognition system for children. Therefore, we identified several methods for improvement. i) Knowledge transfer on Hybrid and end-to-end models by using transfer and multi-task learning and ii) Adapter transfer in end-to-end speech recognition for a parameter efficient transfer.

Keywords

Automatic Speech Recognition Children, Children speech, Atypical speech, Atypical speech recognition

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1

Introduction

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The faculty of expressing or describing thoughts, feelings and needs by using language is a fundamental ability in our daily lives. As per by the Oxford dictionary, *language is defined as the system of communication in speech and writing that is used by people of a particular country or area*. Consequently, language can be conceptualised as a intricate and rule-governed system that empowers individuals to convey abstract concepts, share experiences, and take part in nuanced forms of communication. Effective communication through language begins with the conceptualization of the message to be transmitted (*conceptualisation*), followed by the selection of appropriate lexical items and subsequent grammatical encoding, culminating by their meaningful organisation (*formulation*). Subsequently, the linguistic representation is transformed into sound through the transmission of this representation from the brain to the muscles of the complex speech system including lips, larynx, glottis, lungs, jaw and tongue (*articulation*) [8]. Unlike language, which encompasses both spoken and written forms, speech specifically refers to the spoken manifestation of language.

This capability to speak and comprehend language is not inherently present but rather develops gradually over time with experience. Babies instinctively engage in pre-linguistic communication, using gestures, facial expressions, and vocalisations to articulate their basic needs. As language acquisition progresses, children transition to the babbling stage, experimenting with sound patterns. Eventually, speech emerges, marking a crucial milestone in communication development, and typically, children reach specific language milestones at particular ages. For example, around 12 to 18 months, a child usually utters their first words and starts imitating sounds. By the age of 4 to 5, children tend to formulate sentences and grasp more intricate concepts. Regarding speech sounds, younger children, approximately 1 year old, can produce basic speech sounds like /p/, /b/, /m/ while older children, around 5 years old, can articulate more complex sounds such as /r/ and /th/. This developmental stage is referred as language acquisition, and it plays a crucial role in a child's overall development. Indeed, our daily dependence on social and communication skills endures throughout our entire lives. Consequently, it is imperative for children to develop the capacity to interact effectively with others to achieve seamless integration into society across all aspects of their lives.

Regrettably, a subset of children experience speech disorders stemming from congenital conditions such as cleft palate, cerebral palsy, and prelingual deafness. Alternatively, certain individuals may acquire speech-related issues during childhood, encompassing cognitive developmental delays, breathing-feeding-swallowing disorders and traumatic brain injuries. Notably, in 2012, empirical data [9] highlighted that 7.7% of children aged 3 to 17 in the United States exhibited communication disorders, with 5.0% of this cohort specifically presenting speech-related problems.

Furthermore, findings [10] suggests that individuals afflicted with childhood speech disorders may confront an increased prevalence of mental health challenges, diminished social well-being, and reduced academic accomplishments in comparison to their peers. This highlights the complex nature of speech

disorders in children, the consequences of which extend into adolescence and adulthood. Hence, early identification and intervention play a pivotal role in mitigating the enduring effects on these children's social interactions, society integration, communication skills, educational progress, and overall well-being.

Pediatric Speech and Language Pathologists (SLPs) play a crucial role in providing therapy to help children overcome the effects of speech disorders and offer early diagnosis. The therapy typically includes exercises and assessments, which can be based on perceptual speech evaluations or standardised tests. To effectively engage children, these activities are often presented as games, taking into consideration the inherently limited attention span of children. Notably, SLPs frequently maintain long-term follow-ups with their patients, allowing them to monitor the evolution of speech quality over time and tailor exercises to the specific needs of each child. The adoption of this individualised therapeutic approach is essential for helping children achieve improved speech and communication skills.

However, a prominent challenge arises concerning the accessibility and availability of speech therapy services. Numerous children, particularly those residing in underserved or remote areas, encounter obstacles in accessing speech therapy resources. Additionally, the hospital environment introduces an additional layer of stress for children. While clinically necessary, the setting may inadvertently contribute to heightened anxiety and discomfort, as children may perceive it as intimidating. Furthermore, the logistical challenges associated with frequent hospital visits impose a substantial financial and time burden on families.

Another obstacle pertains to the continuity and consistency of therapy. Children may experience interruptions in their therapeutic journey due to factors such as financial constraints, scheduling conflicts, or alterations in healthcare coverage. These disruptions have the potential to impede progress and undermine the effectiveness of the therapy. Lastly, it is imperative to acknowledge that, despite professional training, inter-expert variability in perceptual assessments may persist, resulting in disparate diagnostic conclusions. To address these challenges, adopting a hybrid approach that combines in-person therapy with technology holds potential benefits [11, 12]. Teletherapy, for instance, has emerged as a promising avenue to bridge geographical gaps and deliver therapy services remotely [13].

In this context, Speech and Language Technologies (SLT) have arisen as highly pertinent within the domain of speech therapy. These technologies encompass a spectrum of computational tools designed to analyse, understand and provide objective and precise automated assessments. Another benefit lies in their potential integration into gamification frameworks, thereby augmenting children's involvement during therapy. Moreover, the ability to record speech utter by the patient during a session using SLT enables post-session thorough analysis and long-term monitoring by the therapist. Due to the aforementioned reasons, the development of such tools has gained considerable attention, empowering patients to engage in exercises beyond therapy sessions, notably in a home setting. Several SLT examples



Figure 1.1: Illustrated herein are some examples of children's Speech and Language Technology applications that were developed during the course of this thesis. On the left is a running platformer game, where the user's voice controls the character. Pitch dictates running and jumping actions, while energy modulates the velocity of these actions. On the right, a reading task game is depicted, wherein a robot instructs the user to read designated words.

developed within the scope of this thesis are illustrated in Figure 1.1.

1.1 Problem statement

Recent years have seen an increased integration of SLT into various aspects of our daily lives, impacting a wide range of environments, including homes, transportation, education, and even the military. Noteworthy examples encompass voice assistants, hands-free computing, healthcare systems, automatic helplines, and speech-to-speech translation services. The performances progress in these applications was made possible through the use of machine learning techniques, especially deep learning approaches, the increasing computational capacities of our devices, and the ever-growing volume of data available to train and improve these systems.

Children represent a promising target audience for SLT due to the inherent complexities of conventional computer interfaces, which pose challenges for them, limiting their capacity to fully benefit from digital platforms. Children commonly face difficulties to manipulate mouse and keyboard inputs. Additionally, the abstract nature of traditional man-machine interfaces can impede the understanding necessary for effective interaction. In this context, speech-based systems emerge as a promising alternative, offering a more natural and accessible means for children to interact with technology. Through the use of speech recognition technologies, these speech systems mitigate the barriers associated with conventional interfaces, providing a fluid and intuitive interaction paradigm that aligns more closely with the developmental stages and cognitive abilities of young users.

As previously mentioned, SLTs are gradually making their way into the field of atypical speech, particularly for children. While these automatic tools are currently in their early stages and have limitations, there is indeed a rising interest in implementing atypical speech and language therapy cutting-edge systems with a focus on assisting SLPs. In this context, systems capable of automatically recognis-

ing speech content, assessing pronunciation quality and detecting speech pathologies could be highly valuable in supporting pediatric SLPs and patients.

All of these objectives require the implementation of a robust automatic speech recognition (ASR) system specifically tailored for healthy children, serving as a foundational model. Nevertheless, while speech recognition technologies have made substantial advancements, leading to increased accuracy, the performance of ASR systems for children remains underperforming in comparison to their adult-oriented counterparts. This discrepancy results leads into unreliable systems for children's speech. The diminished performance can be attributed to a combination of factors, including intra- and inter-speaker variability, limited linguistic and phonetic knowledge, and the scarcity of available data.

In this thesis, we will undertake a comprehensive investigation into the intricacies of children's speech, closely examining the inherent differences between children and adults in the domain of ASR. Through this examination, the objective is to analyse the constraints associated with the application of adult-based systems to children's speech and, subsequently, to outline methodologies for enhancing ASR systems specifically designed to accommodate the variability present in children's speech. The overarching aim is to establish a robust foundational system that effectively addresses the recognition of children's speech characteristics.

Our work specifically aims to answer the following research questions:

1. Which knowledge transfer approach is best for efficiently modelling and improving automatic recognition of children's speech? Can these approaches be used to efficiently exploit low-resource children's speech data from multiple languages?
2. How do end-to-end automatic speech recognition models achieve state-of-the-art results for children's ASR when finetuned from an adult model? Particularly, what are the components that are most important to fine-tune?
3. Is it possible to develop an speaker-based, parameter-efficient automatic speech recognition model?
4. Is it possible to use children's synthetic speech to extend the amount of children's data? How can we control the quality and speakers' variability?

1.2 Contributions

1.3 Structure for the thesis

In Chapter 2, we survey related work relevant to this proposal. First, we explain how children's speech differs from adults' speech in terms of frequency, language, and available data. We then present a brief summary of automatic speech recognition systems and the most recent approaches to solving children's

automatic speech recognition challenges. In Chapter 3, we present our work done on the hybrid speech recognition framework, while in Chapter 4 we present the work based on end-to-end models. Finally, in Chapter 6 we discuss the current and future work and the associated timeline.

2

Background - Children automatic speech recognition

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Automatic Speech Recognition (ASR), or Speech-to-text (STT) refers to the process of mapping a raw spoken audio utterance into its corresponding text. The growing number of potential applications drives the demand for reliable ASR systems. With some of the applications being in the academic, medical, industrial or military fields. Significant progress has been made in recent years, thanks to increased attention from industry and government. Leading to the release of applications such as voice assistants, hands-free interfaces, healthcare assistance, live translation, etc.

Nowadays, most ASR applications are based and developed on adult speech and perform well under matched conditions. However, when faced with mismatched scenarios, like children's speech, these algorithms exhibit a stark performance decline, often two to five times worse [14]. This discrepancy in performance can be attributed to the intra- and inter-speaker variability of children's speech. In fact, speech serves as a conduit not solely for linguistic content but also for paralinguistic cues that unveil aspects of the speaker's identity, including age, gender, health status, emotional state, and regional origin. All of these paralinguistic characteristics within speech present a challenge and might limit accurate speech recognition. Moreover, numerous additional factors, including noise, speaker variability, mispronunciation and data scarcity, have a negative impact on the performance.

More recently, the potential applications of automatic speech recognition in education and entertainment have led to a growing interest in automatic speech recognition for children. Indeed, children represent a demographic that aligns well with these applications for several compelling reasons. The complexity of traditional computer interfaces, like keyboards and mice, can pose challenges for young children, making voice interfaces a more accessible and user-friendly option. Furthermore, voice and language applications, including reading tutors and speech and language acquisition assistants, hold the promise of addressing educational inequalities among children and facilitating their integration into broader societal contexts.

In this chapter, we will first present the different challenges of children's ASR and then review the ASR architectures. Next, we will present state-of-the-art methods to address these challenges from different perspectives. Finally, we will discuss the different perspectives and possible contributions in the context of this proposal.

2.1 Children speech recognition challenges

Children's speech exhibits significant differences from adults, primarily due to the impact of growth and intellectual development. This discrepancy between adult and child speech can be explained by various factors and constitutes the challenges of children's ASR. In this section, we will delve into three primary factors and hurdles linked to children's ASR. To begin, we discuss the acoustic variability of children's speech. Subsequently, we review the complexities of linguistic and phonetic knowledge aspects. Finally,

we will present the issue of the limited availability of children's speech corpora.

2.1.1 Speech variability

Speech production is a complex process involving the coordinated actions of various components in the speech production apparatus, including the vocal folds, tongue, lips, and mouth. This coordination results in changes in air pressure, which generate the speech waveform, essentially an air pressure measurement over time. Human speech waveforms encompass a range of frequency components spanning from 20Hz to 20kHz, which are detected and processed by the human auditory system. A precise comprehension of these frequency components, such as the fundamental and formant frequencies, is essential for the development of effective speech-processing tools.

The fundamental frequency, often referred to as F0, holds a crucial role in the analysis of speech signals. It characterises the (quasi-)periodic average oscillations produced by the vibrations of the vocal folds. Measured in Hertz (Hz), F0 is often referred to as the acoustic correlate of pitch. F0 exhibits an inverse relationship with the vibrating mass of the vocal folds, leading to distinctive F0 values across different demographic groups. Typically, adult men have lower F0 values (approximately 100-150Hz), women tend to have higher F0 values (usually 200-300Hz), and children often demonstrate even higher values (generally 300-450Hz). Furthermore, it is important to note that typically, F0 is not stationary, it exhibits continuous variation within a sentence. This dynamic nature of F0 allows it to be used expressively, conveying nuances such as emphasis and questions in speech.

A formant is a concentration of acoustic energy around a specific frequency in a speech waveform. Formants are essential in characterizing vowel sounds and distinguishing between them. Typically, the first three formants, denoted as F1, F2, and F3, are used in speech analysis. These formant frequencies are vital for vowel recognition and contribute to the timbre of speech sounds.

In 1952, Peterson and Barney [15] conducted a pioneering study on vowel formant components of children's speech, comparing them to those of adult men and women. Their research revealed significant variations in vowel frequencies based on the speaker's age and gender. Subsequent studies [1, 2, 16] provided further insights, revealing a correlation between acoustics and children's age, primarily explained by the growth of the children's vocal tract. The scaling behaviour of formant frequencies as a function of age is illustrated in figure 2.1(a). Here the vowel space, defined by the four reference vowels (/IY/, /AE/, /AA/ and /UW/), linearly decreases with age and aligns with the adult level around the age of 15. Moreover, [1] highlighted that the vowel space becomes more compact with increasing age showing a downward trend in the dynamic range of formant values. These variations and age-related differences emphasise the critical challenge of inter-speaker variability, especially for young children.

In addition to inter-speaker variability, children's speech exhibits intra-speaker variability, meaning that the same individual's speech can vary in different ways. This variability arises from two sources.

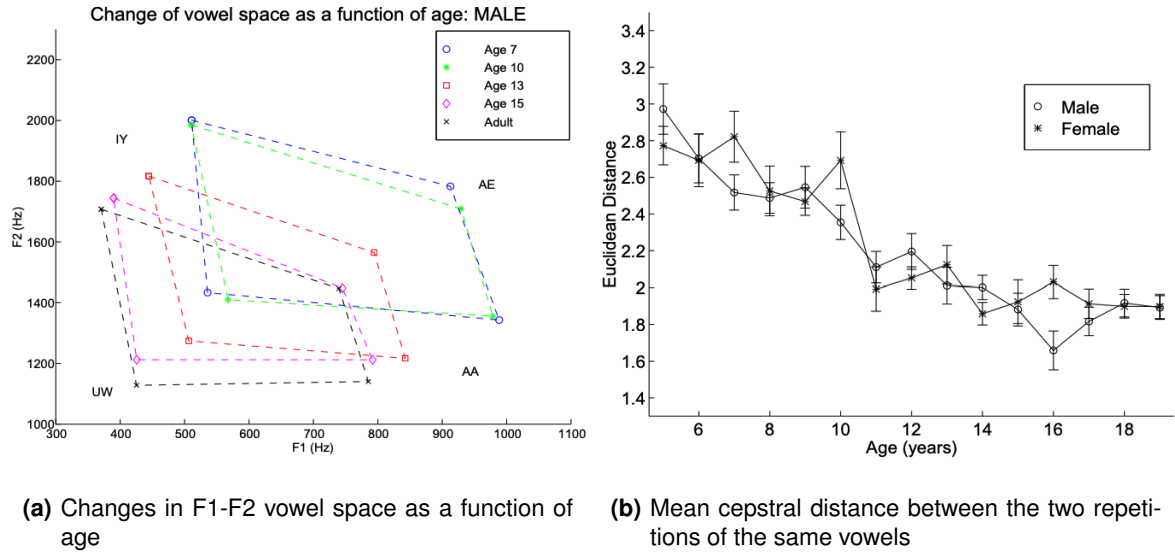
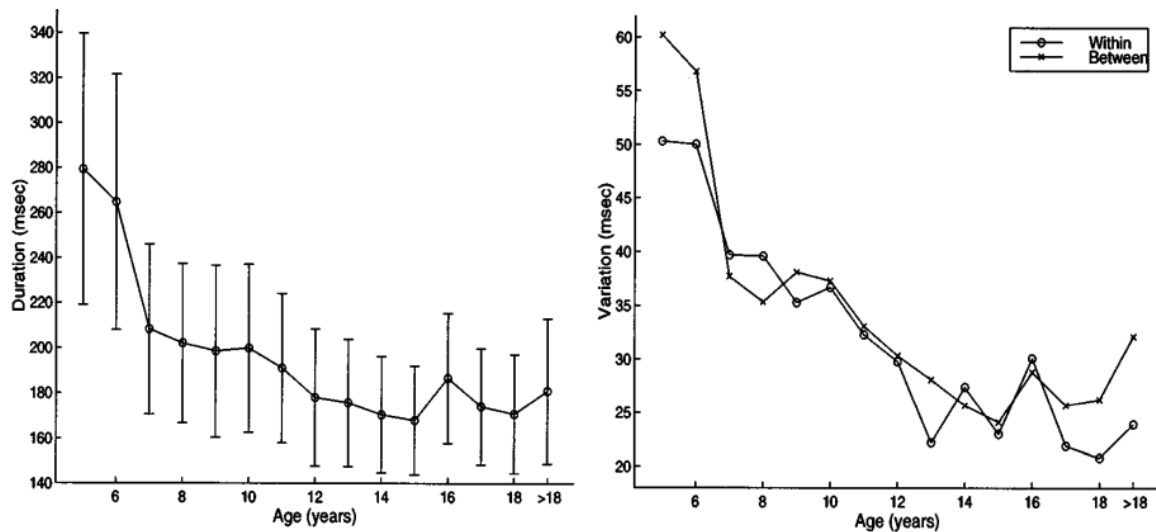


Figure 2.1: Formant and cepstral variability. Figures taken from [1]

Firstly, as previously discussed, the acoustic characteristics of the same child can significantly differ at different ages due to the growth of their vocal apparatus over time. Secondly, even at the same age, the same child may produce variable speech, even when articulating the same vowel. As depicted in Figure 2.1(b), the average cepstral distance between two repetitions of the same vowels by the same child decreases with age, particularly after the age of 10. This reduction is attributed to the progressive mastery of speech articulation components.

Segmental duration is another important aspect of human speech. A segment, as defined by Crystal [17], is: *"Any discrete unit that can be identified, either physically or auditorily, in the stream of speech"*. These segmental durations could be vowel or sentence duration. The literature emphasizes the significance of vowel duration in vowel discrimination. Research, as presented in [2], investigates how these characteristics change in children's speech. As demonstrated in Figure 2.2(a), the average duration of vowels in children varies with age. On average, younger children tend to have longer vowel durations, resulting in a slower speaking rate. However, as children become more comfortable with speaking processes with age, vowel duration gradually decreases. Similarly to children frequency variations, segmental duration exhibits intra-speaker variability, as illustrated in Figure 2.2(b).

In conclusion, dealing with both intra- and inter-speaker variability in children's speech, particularly in those under the age of 15, poses a substantial challenge for developing high-performance speech processing models for children, especially in the context of ASR, where the age of the child is often unknown. Moreover, the processes involved in speech development in children, including the growth of vocal tract structures, changes in speech components, and the maturation of speech motor control, are simultaneous and overlapping. This development makes it considerably more challenging to disentangle



(a) Averaged-vowel duration across all vowels and subjects in each age group

(b) Within- and between-subject variations. The between-subject variation is reduced by a factor of 2.0

Figure 2.2: Segmental duration variability. Figures taken from [2]

and model their acoustics effects accurately.

2.1.2 Language and phonetic knowledge

Speech is hierarchically composed of sentences, words and phonetic units. Language refers to the cognitive processes of understanding and selecting the appropriate words or phonetic units to convey meaning, as well as arranging these words into coherent sentences. Children's ability to use language effectively develops with age, achieving adult capabilities around the age of 13 [2], enabling them to transition from producing simple sounds and words to generating more complex sounds and fully articulated sentences. During this learning process, due to their limited and developing linguistic knowledge, children make pronunciation errors and disfluencies [18] such as:

- Substitution: Incorrectly pronouncing an entire word
- Omission: Not pronouncing an entire word
- Mispronunciation: Incorrectly pronouncing some part of a word
- Pause and Hesitation: Partially pronouncing a word
- Filler and mumbling: Using unintelligible speech or noise as filler for hesitation or pause
- False-start: Stop and start a sentence in the middle of one

- Sound-out: Pronouncing word as a sequence of distinct units rather than continuous sounds

Potamianos and Narayanan [19] found that inter-speaker variability is about twice more than intra-speaker variability. They also showed that there is twice more mispronunciation for children of 8 to 10 years old compared to children of 11 to 14, while the trend was reversed for filler pause. Furthermore, younger children also produce more false starts and breathing.

Mispronunciation and disfluencies in children’s speech are not necessarily present in adult speech. This statement was proven by [20, 21] employing a language model trained on children’s speech and demonstrating the benefits over the use of adult language models. Demonstrating that children’s language differs from adults in terms of grammatical structures, mispronunciation, and disfluencies.

2.1.3 Data scarcity

With the introduction of deep learning in recent years, increasing computational power and a large amount of data available have resulted in considerable improvements in the ASR area. This success has been attributed to deep neural networks (DNN) ability to better approximate the non-linear functions needed to model speech. Despite recent efforts to collect larger children datasets [22–24], the majority of publicly available children corpora include fewer than fifty hours of speech, whereas adult speech corpora containing hundreds, or even thousands, of hours can be easily accessible [25]. Furthermore, the majority of the accessible children’s data are English corpus [22, 24, 26–28]. However, English is a large-size, resource-rich pluricentric language which should be seen more as an exceptional case, rather than an average representative. Furthermore, collecting speech data from children is a difficult and time-consuming task. Indeed, many factors severely reduce the quality of the gathered speech. These factors include children’s short attention spans, recording environments that might be noisy (such as a classroom), and the quality of the speech, which is highly dependent on the task at hand (reading tasks are generally more complex for children).

The need for a large database of children’s speech to cover a higher variability has been proven by [29]. In this work, the authors trained an automatic speech recognition model using a large in-house corpus of children’s speech, comparable in size to an adult speech corpus. This resulted in state-of-the-art performance competitive with adult speech recognition systems.

2.2 Introduction to automatic speech recognition

In this section, we provide an overview of the main trends and modules of automatic speech recognition systems, as this understanding is necessary for the rest of this proposal. This overview is by no means exhaustive, and some subjects that are outside the scope of this thesis are excluded. More thorough information regarding ASR can be found in [30–32]. This section is divided as follows: First, we will present traditional ASR systems, and then we will explain the end-to-end paradigm for ASR. Finally, we will present automatic speech recognition metrics.

2.2.1 A brief history of Automatic Speech Recognition

2.2.1.A Early Days

The first projects for a technology that resembles to speech recognition date back to the 1950s-1960s. At this time, the work focused on isolated word in a speaker dependent context. The earliest project was a digit recognizer created at Bell Telephone Laboratories in 1952. This recognizer automatically recognize telephone-quality digits spoken at normal speech rates by a single adult-male individual, and obtained an accuracy up to 99%. This system relied on formant frequency approximations to recognise whole word. No model of syllables, consonants, vowels or any other sub-word units are present in this recognizer. The word was treated as a single unit, and compared with ten standard digit patterns to find the best match. In order to do this the circuit extracted two frequency ranges, frequencies below and above 900 Hz. These two frequency ranges roughly approximate with the first two formants of speech. Then, these two formants approximation are plotted on a 2D-plot with a trace interruption period of 10ms. Finally, when a new audio is provided to the recognizer the newly generated plot is compared the to ten references plots of the ten digits and the closest match is return by computing the highest relative correlation coefficient. Figure 2.3 shown an example of the 2D representation of the ten digit.

Nonetheless, extended such a system would be not possible. This template-matching approach required each word representation to be saved on disk and compare the unknown spoken word with all this representations. When extend this approach to a larger vocabulary, the time and disk usage complexity become an issue. In addition, in order for the circuit to deliver an accuracy of the same range for a new speaker, a preliminary analysis of the speech of this individual and circuit adjustment is required.

Later, in 1962, IBM developed Shoebox, a device that recognize 16 spoken words, the ten digits and command words such as "plus", "minus" and "total" using a similar pattern matching algorithm.

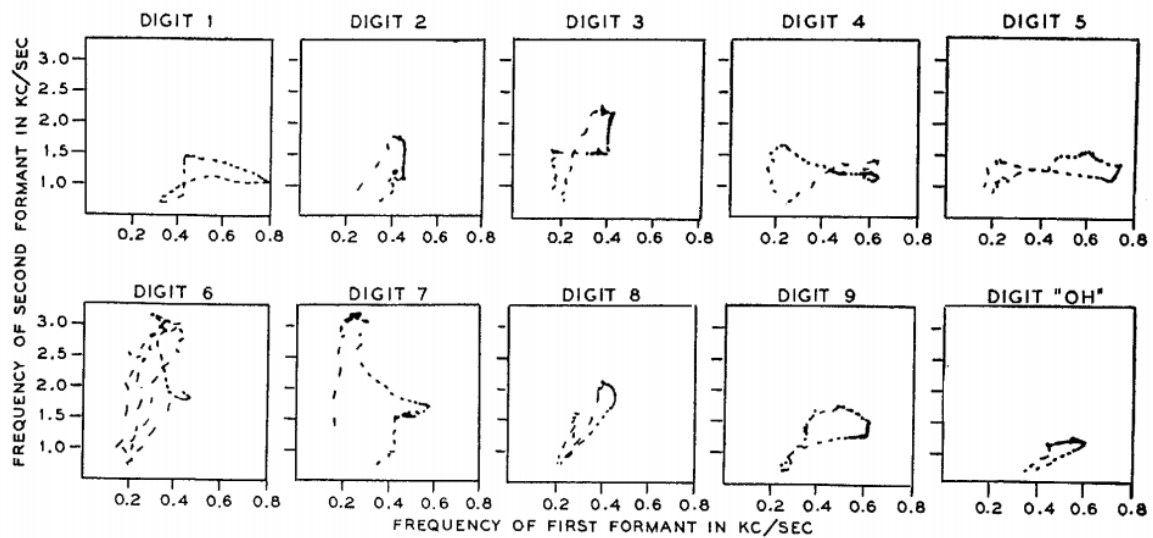


Figure 2.3: Example of a standard digit pattern from Davis et al. 1952

2.2.1.B The Speech Understanding Research program

In the early 1970s, after the initial success of pattern matching algorithms for single word recognition, the Advanced Research Program Agency of the U.S. department of Defense, ARPA, funded a five-year program Speech Understanding Research (SUR). The goal was to "obtain a breakthrough in speech understanding capability that could then be used toward the development of practical man-machine communication system ". It is in the context of this program that four different research groups have been funded: two groups from Carnegie-Mellon University (CMU), one from Bolt Beranek and Newman Inc. (BBN Hwim) and the last one from System Development Corporation (SDC). For each group, they were given a different task: Facts about ships, travel budget management and document retrieval. Where the end goal task was to create a system that can recognize simple sentences in the context of the task from a vocabulary of 1,000 words with a 10% WER in a reasonable time.

They realised that the pattern matching word identification strategy could not be applied to the problem of sentence understanding and they had to redesign the single word identification system. First of all, acoustic characteristics of words can vary considerably depending on the context of the sentence; the phonetic and phonological processes of English. Storing on disk each words and all his different variations was not possible. In addition, discovering each words boundaries was almost impossible and compute the pattern-matching, even if the word boundaries were found, would be long and would not fit in the reasonable time requirement. Indeed, the system would compare the query word with each of the 1,000 words stored and all the possible variations. Secondly, output sentence length is variable and unknown, which is not the case in a single word identification. Therefore it was decided to use a smaller unit than the word to model speech, the phonemes. The phonemes are the smallest distinctive

and meaningful unit that compose speech. For each language is associated a set of finite number of phoneme, usually under 50, that can be use to create words.

Among all proposed systems of the project, the Harpy system, implemented by Lowerre (1976) for the Carnegie Mellon team, showed the best performances. Harpy is a speaker-specific system, using pattern matching algorithm at the phoneme level instead of word level. The Harpy system use a set of 98 phonemes and diphones, a pronunciations of all words and a graph where all accepted sentences are compiled using 15k states. When a new utterance arrived in the system, it is first low pass filtered at 5kHz, digitized at 10k samples per second, then a 14 linear prediction coefficients are computed with a 10ms shift and group it together similar adjacent acoustic segments to speed-up the decoding process. These audio segments are then compared against the 98 phonemes templates and the best path over the graph is returned. Figure 2.4 show a decoding graph example.

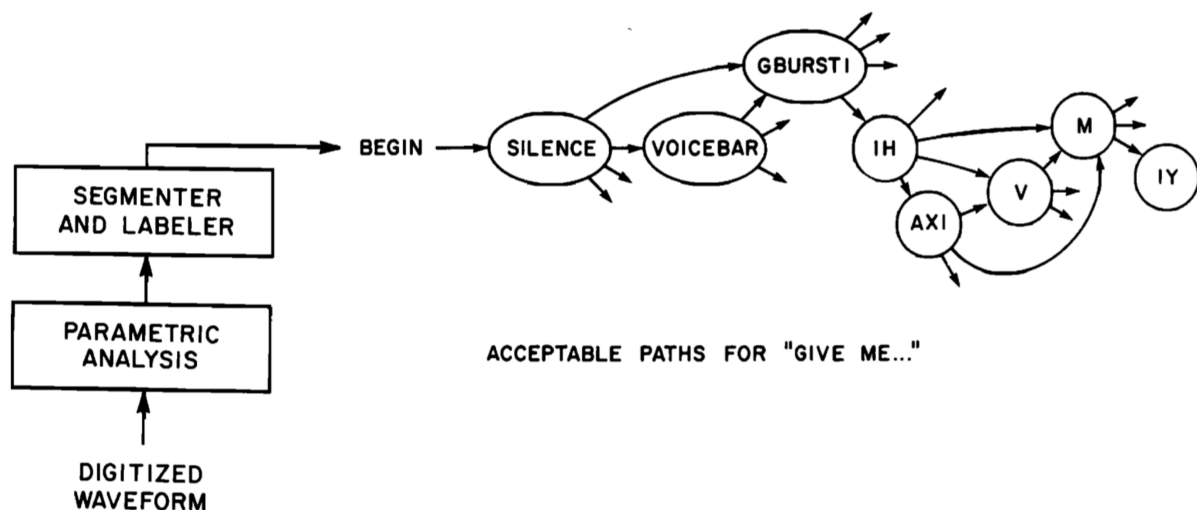


Figure 2.4: Example of a decoding graph from the Harpy system for the sentence "GIVE ME" from [REFERENCE HERE]

Despite the success of the Harpy system, it is a speaker-specific system. For a new speaker, the system must be tuned over the 98 phonemes templates. In addition, this system can not understand more the 1,000 words and only use a simple handcrafted grammar, making this system not reliable for spontaneous speech. Finally, the decoding time was far from real time.

2.2.1.C From template matching to statistical modeling: HMM based models

With the introduction of the Hidden Markov Models (HMMs) in the 1970's, Gaussian Mixture Models (GMMs) and statistical grammars in the 1980's, automatic speech recognition research shift from pattern matching approach toward statistical modeling in the 1990's. All of these components were successfully

combined into the Hidden Markov Model Toolkit (HTK), with first benchmark on the DARPA's Resource Management task in 1992 [CITATION]. The components introduced in the HTK toolkit shown in figure 2.6 are still at the core of modern HMM-GMM systems for speech recognition.

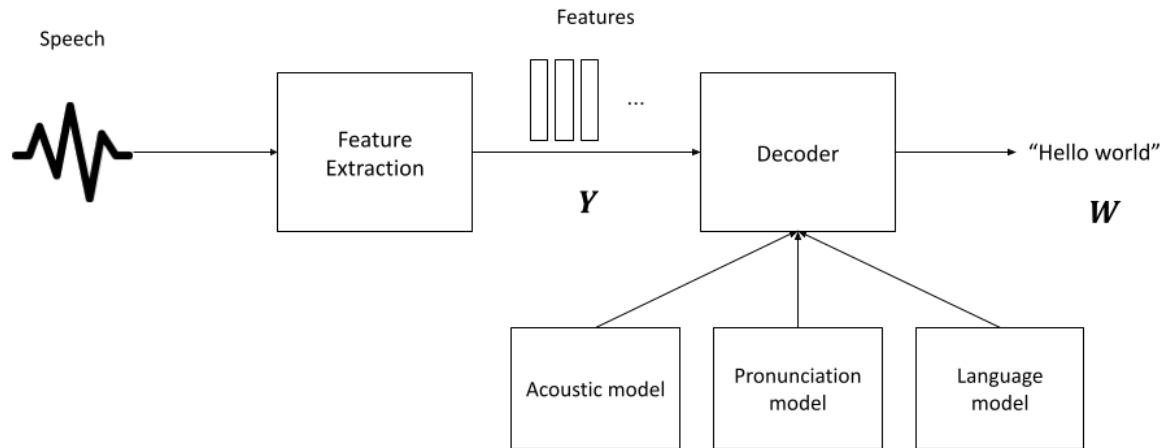
In this statistical framework, the continuous speech audio waveform is converted to a sequence of fixed size acoustic vector $Y_{1:T} = y_1, \dots, y_T$. Then the decoder try to find the sequence of word $W_{1:L} = w_1, \dots, w_L$ which is most likely to have generated Y . More formally:

$$\hat{w} = \underset{w}{\operatorname{argmax}} \{P(w|Y)\} \quad (2.1)$$

However, this probability is difficult to model directly, but using Bayes' Rules gives:

$$\hat{w} = \underset{w}{\operatorname{argmax}} \{P(Y|w)P(w)\} \quad (2.2)$$

Here, the likelihood $P(Y|w)$ is determined by the acoustic model component while $P(w)$ is determined by the language model component. The following sections describe the different components and process in more detail.

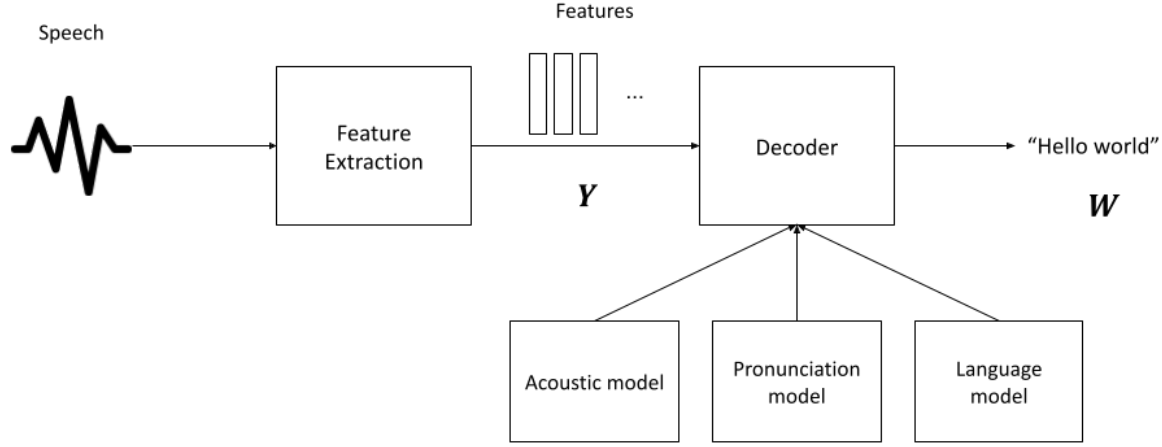


architecture.png

Figure 2.5: Architecture of a HMM-based speech recognition system

2.2.2 Traditional automatic speech recognition systems

The introduction of Hidden Markov Models (HMMs) in the 1970s, Gaussian Mixture Models (GMMs) in the 1980s, and statistical grammars in the 1990s, pushed automated speech recognition research away from pattern matching toward statistical modelling [33]. For example, the Hidden Markov Model Toolkit (HTK) [34], a toolkit extensively adopted by the community, proposed the first benchmark on DARPA's Resource Management challenge in 1992 [35]. Those components of the traditional ASR, illustrated in



architecture.png

Figure 2.6: Architecture of a HMM-based speech recognition system

figure 2.6, are still at the core of modern HMM-based speech recognition systems. Later, ASR systems performances improved significantly as a result of the transition from HMM-GMM to the deep neural networks paradigm HMM-DNN [36], called hybrid models.

In this statistical framework, the continuous speech audio waveform is converted to a sequence of fixed size acoustic vectors $X = x_1, \dots, x_T$. Then the ASR system tries to find the sequence of words $w = w_1, \dots, w_L$ which is most likely to have generated X . More formally:

$$\hat{w} = \underset{w}{\operatorname{argmax}} \{P(w|X)\} \quad (2.3)$$

However, this probability is difficult to model directly, but using Bayes' Rules gives:

$$\hat{w} = \underset{w}{\operatorname{argmax}} \left\{ \frac{P(X|w)P(w)}{P(X)} \right\} = \underset{w}{\operatorname{argmax}} \{P(X|w)P(w)\} \quad (2.4)$$

Here, the likelihood $P(X|w)$ is determined by the acoustic model component while $P(w)$ is determined by the language model component. As $P(X)$ is not required to determine the maximum, it can be omitted. The following sections describe the different components and processes in more detail.

2.2.2.A Feature extraction

The feature extraction component's purpose is to extract useful information about the speech's linguistic content. To this end, at each time step, the continuous waveform is transformed into a small fixed-size vector. An acceptable approximation is that speech is considered stationary on the period covered by a single vector. As a result, feature vectors are generally generated every 10-milliseconds with a 25-millisecond overlapping window.

Mel-frequency cepstral coefficients (MFCCs) [37] are the most widely used features for speech recognition on HMM-GMM and HMM-DNN systems. These are generated by: i) Computing the spectrum of the windowed speech signal by using a discrete-time Fourier transform. ii) Then convoluted with triangular Mel filters (usually 20, called filterbanks), where those filters are narrow in the lower frequencies and wider in higher frequencies, which is similar to the human cochlea. iii) A log transformation is applied. iv) In order to decorrelate the filterbank energies a discrete cosine transform (DCT) is used. Note that some HMM-DNN models are using the filterbanks from ii) directly as input features.

For each time step, the feature vector only contains information about the current speech window. It does not, however, provide information on the dynamics of the signal. To bring to these spectral coefficients some context, the first and second-order derivatives are concatenated (*Delta* and *Delta*, respectively). These first-order derivative coefficients are calculated as follows:

$$\Delta_i = \frac{\sum_{n=1}^N n(f_{i+n} - f_{i-n})}{2 \sum_{n=1}^N n^2} \quad (2.5)$$

Where the f_i is the feature at the instant i . Usually, n is equal to 2. Furthermore, the second-order derivate $\Delta\Delta$, also written Δ^2 , can be computed using the same calculation 2.5 by replacing the spectral feature f_i by the first-order derivative Δ_i . Thus, the concatenation of the derivatives Δ and Δ^2 features with the spectral features is denoted x_i as follows:

$$x_i = [x_i^T \quad \Delta_i^T \quad \Delta_i^{2T}]^T \quad (2.6)$$

2.2.2.B Acoustic model

The role of the acoustic model (AM) is to determine $P(X|w)$. In other words, identify which phone or none was uttered in each frame. The most straightforward method would be to use any classifier, such as GMM models with one GMM per phone. However, such a classifier does not account for the temporal dependencies of speech (co-articulation). Indeed, to properly categorize each frame, the current frame and its context (previous and following frames) are necessary. In addition to co-articulation, there is an acoustic difference at the beginning, middle and end of each phone. Therefore, the HMM framework has been presented as a solution to these concerns [38]. Indeed, HMM provide temporal flexibility (e.g., self-looping) as well as a well-understood framework with effective learning (Expectation Maximization) and decoding (Viterbi) algorithms.

In the HMM terminology, the observed variables are called observations (here, the acoustic features), while the hidden variables are referred to as states. The states are generated using a first-order Markov process, where the i^{th} state S_i is only dependent on the previous state S_{i-1} . It is assumed that all observations are independent given the states that generated them. To better model acoustical differ-

ences, one phoneme is typically represented by a three states linear HMM (generally the beginning, middle and end of the phoneme). Finally, with HMM-GMM, each state is modelled by a GMM to determine the likelihood of the observation in that state (i.e. determine if the speech segment matches the phoneme pronounced). A monophone GMM-HMM contains one model for one phoneme. For example, since English has 44 phonemes [39], a monophone system on English will have 44 separate GMM-HMMs. However, monophone systems are limited to modelling natural speech in its full complexity. This is due to co-articulation effects, in which the pronunciation of a phoneme is influenced by its context. Therefore, ideally, there would be a model for every phoneme in every phonetic context. In other words, the phoneme with its left and right context. Such system is called a triphone system. Consequently, there should be N^3 models to train for a language with N phonemes. For instance, as English has 44 phonemes, there should be $44 * 44 * 44$ or 85,184 models. Since there is not enough data to see all these contexts, they are clustered using a decision tree.

The idea of hybrid models emerged in the 1990's with the usage of multi-layer perceptrons (MLP) as substitutes for GMMs in HMM-GMM system [40, 41]. In 2012, the use of deep neural networks (DNN), MLPs with a large number of hidden layers, led to considerable improvement in a wide variety of ASR tasks [36]. Due to the large number of parameters induced by the many hidden layers, the models can capture complex and highly non-linear relationships between inputs and outputs. Even though HMM-DNN and HMM-GMM models are comparable, their training is different. Indeed, feed-forward network training is dependent on GMM training to provide labelled data. In fact, neural networks require input (audio features) and output (phoneme labels) in order to be trained. However, most standard speech training data lacks this degree of detail, with simply audio waveforms and utterance transcriptions available. As a result, the HMM-DNN model needs to use alignment generated by an HMM-GMM. More precisely, the first step is to use HMM-GMM flat-start monophone training, then iterate into triphone steps with more precise alignments. In consequence, the more precise the HMM-GMM alignment is, the better the DNN model training will be.

2.2.2.C Pronunciation model

The role of the pronunciation model, also known as a dictionary or lexicon, is to map phonemes into words. This mapping takes the form of an entry where all possible words are associated with their corresponding sequence of a basic unit (phones). Traditionally, this mapping is obtained manually, based on phonetic and linguistic knowledge. Note that it is possible to have many different pronunciations for a single word. In addition, statistical grapheme-to-phoneme (G2P) tools [42] allow to generate pronunciations for words that are not included in the dictionary.

2.2.2.D Language model

The role of the language model (LM), also known as grammar, is to determine $P(\mathbf{w})$ of equation 2.4. Such models are widely used in a broad range of application, such as Natural Language Processing (NLP) [43], computational biology [44] or data compression [45]. There two most successful approaches to language modelling largely adopted in ASR are respectively statistical and deep learning based.

Statistical language models use traditional techniques such as HMM and n-grams. N-grams are the simplest approach for language modelling and are defined as follows:

$$P(\mathbf{w}) = P(w_1, w_2, \dots, w_L) = \prod_{i=1}^L P(w_i | w_{i-n}, \dots, w_{i-1}) \quad (2.7)$$

The objective is to infer the probability of the next word from the context of the n preceding words by counting N-gram occurrences to form maximum likelihood parameter estimates. In the particular case when $n = 1$, the 1-gram -or unigram- model ignores any conditional context. It evaluates each word independently:

$$P(\mathbf{w}) = P(w_1, w_2, \dots, w_L) = \prod_{i=1}^L P(w_i) \quad (2.8)$$

However, even though a larger n of context yields superior results, in practice, a higher n becomes more combinational and may not be a realistic option due to computer limitations. As a result, most ASR applications employ trigrams or 4-grams. Furthermore, precisely determining the start of the sequence probability might be difficult. Finally, because of the reliance on training data, n-grams cannot estimate the likelihood of an unseen word without smoothing [46]. Therefore, those issues drive the community to seek alternative technologies, such as neural networks. Deep learning-based language models demonstrate good modelling capacity by using neural networks with complex architecture. In addition to being very flexible, they are easy to train and do not require as many resources as n-grams to be efficient. Recently, the current state-of-the-art for language modelling tasks are built on deep learning networks such as [47]. They become prominent primarily as a result of the attention mechanism. In contrast to n-grams that only pay attention to the n previous words, the attention mechanism gives more importance to the most important words in the sentence.

2.2.2.E Decoder

As mentioned in the previous section, the role of the decoder is to use the language, acoustic and pronunciation models to search for the most likely word sequence $\hat{\mathbf{w}}$ given a sequence of acoustic feature \mathbf{Y} (equation 2.3).

Dynamic programming, such as the Viterbi algorithm [48], provides an effective technique to decode. However, when the vocabulary size and n-gram order increase, the Viterbi algorithm becomes un-

feasible. As a result, several ways have been proposed, ranging from constraining the search space [49] to dynamically extending it as the search advances [50]. As an alternative, weighted finite-state transducers (WFST) were introduced following extensive community research [51, 52]. The advantage of WFSTs is their ability to integrate hierarchical information (sentences, words and phonemes) into an optimised graph. The Kaldi speech recognition toolkit [53] is an example of a widely adopted toolkit that uses this type of decoding.

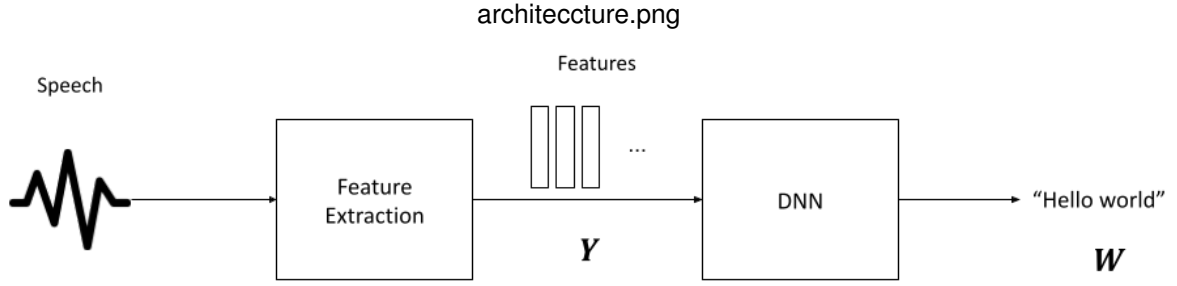


Figure 2.7: Architecture of an end-to-end speech recognition system

2.2.3 End-to-end automatic speech recognition

End-to-end architectures propose to simplify the ASR process by combining all HMM-based components (acoustic model, pronunciation model, and language model) into a single neural network, as illustrated in 2.7. This shortens training and decoding time as well as allows joint optimisation. One of the key disadvantages of hybrid models is the factorized training of all modules independently, which can lead to error accumulation and mismatches between the different components. Furthermore, end-to-end models map audio input feature sequences directly into label sequences, eliminating the requirement for pre-aligned training data and post-processing of outputs. To this end, word-level transcriptions are converted into character-level transcriptions. Thus, given the sequence of fixed size acoustic vector $\mathbf{X} = x_1, \dots, x_T$ and the corresponding sequence of characters $\mathbf{Y} = y_1, \dots, y_N$, where T and N are respectively the numbers of frames and the length of the phone sequence, the goal of end-to-end models is to learn the conditional probability of the frame y_i given the input \mathbf{X} and the previous output $y_{<i}$:

$$P(\mathbf{Y}|\mathbf{X}) = \prod_{i=1}^N P(y_i|\mathbf{X}, y_{<i}) \quad (2.9)$$

Recently, the increased interest in ASR end-to-end architectures enabled performances that match or exceed HMM-based models for a wide range of ASR datasets [54,55]. However, end-to-end systems, in contrast to hierarchical HMM-based models, are data-driven models and therefore require a large amount of training data to work correctly [56].

Two paradigms stand out in the end-to-end literature: connectionist temporal classification and sequence-to-sequence-based architectures. The following sections describe these two different approaches in more detail.

2.2.3.A Connectionist Temporal Classification

The first step towards end-to-end automatic speech recognition was introduced by Grave et al. [57] with the Connectionist Temporal Classification (CTC) objective function. Used for labelling unsegmented

phoneme sequences. In contrast to HMM-based models, the CTC loss does not require pre-segmented training data since alignments between N input speech frames X and the output sequence of T phones Y are learned automatically if $N \leq T$. For this purpose, the CTC is divided into two sub-processes: Path probability calculation and Path aggregation.

Let \mathcal{V} a set of possible paths of phone-label sequences of length T and p_k^t the probability of observing the label k at time t . Note that as CTC requires the length of the label sequence Y to be equal to T , the length difference between N and T is compensated for by introducing a blank label “-”, which corresponds to the probability of observing no label.

First, the role of the path probability calculation is to compute the conditional probability of any path $\pi \in \mathcal{V}$ as follows:

$$p(\pi|X) = \prod_{t=1}^T p_{\pi_t}^t, \forall \pi \in \mathcal{V} \quad (2.10)$$

Where π_t corresponds to the label at position t of the sequence π .

Secondly, path aggregation merge the same contiguous labels and deletes the blank label. For example two different paths “b-ii-r-d” and “b-i-r-dd” becomes “bird”. Then the CTC algorithm assigns the probability $p(Y|X)$ by summing over the probability of all possible alignments between X and Y :

$$p(Y|X) = \sum_{\pi \in \theta_Y} p(\pi|X) \quad (2.11)$$

Where θ_Y is a subset of \mathcal{V} of all possible path π corresponding, after aggregation, to the label sequence Y .

2.2.3.B Sequence to sequence

The second paradigm of end-to-end automatic speech recognition systems is the sequence-to-sequence (Seq2seq) architecture. Initially, proposed by Sutskever et al. [58] for machine translation. The objective of machine translation is to convert word sequences from one language to word sequences in another. Where in general, input and output sequences have different lengths. With the addition of attention mechanisms [59], seq2seq architecture has shown to be useful for many applications such as image captioning [60], conversational modeling [61], text summarisation [62] and ASR [63]. Seq2Seq architecture is usually composed of two components, an encoder and a decoder. The encoder processes the variable length input sequence to a variable length sequence of vectors (also called “internal state” or “hidden representation”). The decoder takes this vector and generates an output sequence of tokens, one at a time:

$$p(y_1, \dots, y_T) = \prod_{i=1}^T p(y_i | y_0, \dots, y_{i-1}, f(H)) \quad (2.12)$$

Where $f(H)$ is a function of the encoder's output $H = (h_1, \dots, h_N)$. In attention-based seq2seq model, attention mechanism [59] is implemented in $f(\cdot)$. For each target token, the attention mechanism searches through H to locate parts that are relevant for the prediction of the current target token. Unlike CTC-based models, seq2seq models do not make independent assumptions about output labels.

2.2.4 Automatic Speech Recognition metrics

Word Error Rate (WER) is a commonly used metric for assessing speech-to-text systems. The ASR system output word sequence is matched with a reference transcription, and the number of substitutions (S), deletions (D), and insertions (I) are summed. As a result, WER is calculated as follows:

$$WER = \frac{S + D + I}{N} \quad (2.13)$$

Where N is the number of words in the reference transcription. As a result, the lower the WER score, the better. The WER computation is based on the Levenshtein distance and works at the word level rather than the phoneme level. Because the denominator N represents the number of words in the reference, a WER score greater than 100% is attainable when the number of mistakes is larger than N . Furthermore, when there is no error in the ASR hypothesis as compared to the reference, the minimum score possible is 0%. Other metrics, based on the same equation but functioning at different levels of transcription, can be found in the literature, such as Phone Error Rate (PER) for phone-based languages or Character Error Rate (CER).

2.3 Children automatic speech recognition

In order to address the different challenges mentioned in section 2.1, efforts have been made on many different aspects of the speech recognition pipeline. At the feature level, with new extractions and adaptations, using data augmentation, modifying annotation detail, updating and creating new acoustic model structures or using new training procedures.

This section reviews state-of-the-art for each of these aspects in more detail. At the end of this section, we will identify the different approaches on which we will focus for the rest of this study.

2.3.1 Feature extraction and adaptation

The feature extraction stage is critical for identifying relevant speech signal components for automatic speech recognition. Achieved by discarding speaker-dependent information such as fundamental frequency while retaining phoneme-dependent information such as formant frequencies. However, the acoustic fluctuation of children's speech leads to close fundamental frequency and formant values, as well as phonetic class overlap due to formant value variability, making typical short-term spectral-based feature extraction challenging. Several strategies have therefore been proposed to improve children's acoustic features, either directly at the extraction stage or at the feature level.

2.3.1.A Feature extraction

Feature extraction methods aim to extract acoustic features directly from the raw speech signal that are capable of better suit children's speech characteristics. The initial step toward this direction was the introduction of perceptual linear predictive (PLP) features in 1990 by [64], where PLP features demonstrated a more accurate formant representation of children's speech. More recently, [65] proposed to shift the feature extraction stage from a hand-crafted fashion to data-driven with a feature learning strategy directly from the raw signal. The intuition behind this end-to-end approach is that hand-crafted features rely on adult speech analysis, whereas children's speech has greater acoustic variability and may not be well suited. In this study, data-driven feature extraction using Convolution-neural-network (CNN) based models discovered relevant features that yield better results than the standard hand-crafted ones. A similar idea has been proposed by [66] in which the convolutional layer of the feature extractor has been replaced by a SincNet layer [67] to effectively adapt an adult end-to-end features extractor to children.

2.3.1.B Feature adaptation

Another perspective to reduce children's acoustic variability is to work directly at the feature level, using standard feature extractors and directly adapting speech features to reduce children's acoustic variability. To this end, Vocal Track Length Normalization (VTLN) has been widely used to wrap spectral features

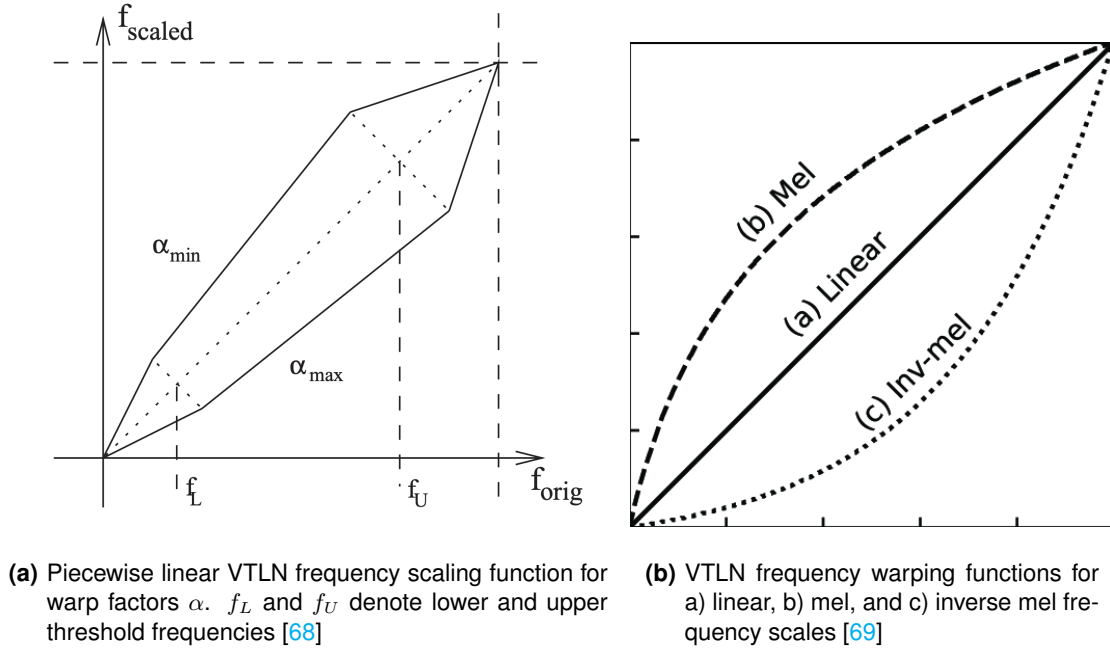


Figure 2.8: Examples of VTLN frequency warping functions

into a canonical space [68, 70]. Applied during the feature extraction process, after the windowing and Fourier transform, VTLN stretch or compress the frequency axis of the spectrum depending on a warping function (several examples of warping function are presented in figure 2.8). As a result, the spectral representation of speaker variance is reduced, and children’s speech features are normalised. In addition, adapting acoustic models with Maximum Likelihood Linear Regression (MLLR) or Maximum A-Posteriori (MAP) and using Speaker adaptive training (SAT) based on feature MLLR (fMLLR) or Constrained MLLR (CMLLR) were found to be effective to improve the performance of children ASR [1, 21, 71, 72].

Recently, some research explored normalising several aspects of speech: Pitch to reduce the spectral mismatch between children [73–75], formant values to better match adult ones [76] or speaking rate with a time-scale modification approach [77]. Moreover, in accordance with the trend of shifting from knowledge to data drive framework, some recent studies attempted to use data-based feature adaptation. For example, [78, 79] used adversarial multi-task learning to produce age-invariant features which minimize the acoustic mismatch between adult and children’s speech.

2.3.1.C Additional features

In addition to feature extraction and feature-level adaptation, some research has demonstrated that concatenating speaker embeddings, such as i-vectors, to acoustic characteristics leads to more speaker-independent models [80]. Similarly, [81] proposes concatenating various prosodic variables such as

loudness, voice intensity, and voice-probability to standard acoustic features, resulting in decreased inter-speaker variances and enhanced phoneme class discrimination.

2.3.2 Detail of the annotation

The material provided in speech corpora is frequently restricted to the audio signal, the corresponding text transcription, and some anonymised speaker's identifier. However, some additional information might be useful in improving children's speech recognition. The incorporation of the speaker's age would allow the development of age-dependent ASR systems in which the acoustical fluctuation caused by vocal track growth would be reduced [82]. Some work showed the efficiency of an annotation at the sub-word level instead of the word level [83] especially to be more robust to mistakes such as mispronunciations or hesitations. Alternatively, in [71, 84] user-dependent pronunciation lexicons are employed to tackle the pronunciations divergence from the canonical and adult patterns.

2.3.3 Structure of the acoustic model

The acoustic model is a key element in the recognition of children's speech since acoustic variability plays a major role in the degradation of children's speech recognition (compared to linguistic variability). As a result, the structure of the acoustic model is essential to be more robust to children's speech variability. In the rest of this section, we will review children's acoustic modelling structures in the context of hybrid models and end-to-end systems.

2.3.3.A Hybrid models

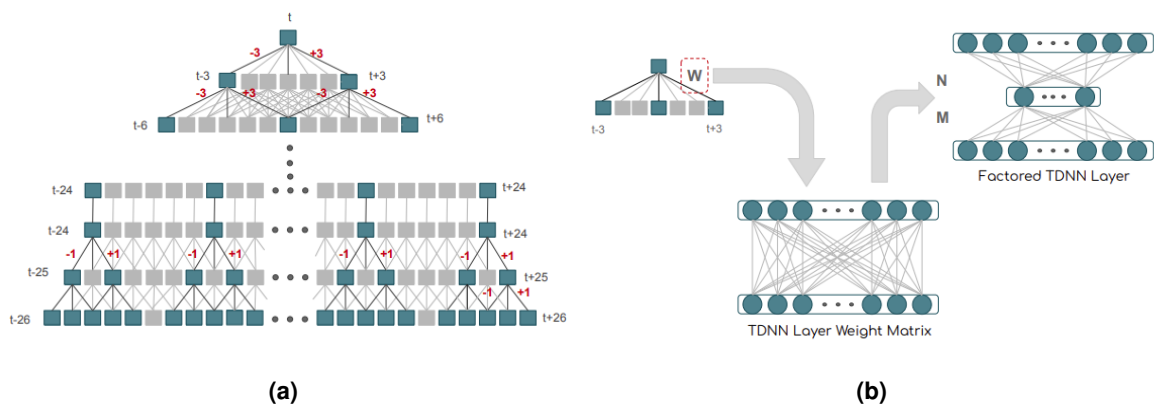


Figure 2.9: (a) TDNN layers with sub-sampling & (b) Factorized TDNN layer from [3]

Acoustic models for children generally follow the latest advances in acoustic modelling for adults. For example, [5] proposed switching from GMM to DNN for hybrid models. However, the traditional fully

connected neural network does not provide the contextual information required for efficient phoneme recognition. Thus, time-delayed neural networks (TDNN) have been proposed ([85]). Working similarly to a one-dimensional convolutional neural network, where at each time step the current time step and its corresponding left and right context are provided, as shown in figure (2.9(a)). Additionally, the use of multiple layers of TDNN allows for the capture of broader contextual information, which leads to better phoneme recognition. More recently, factorized time delay neural networks (TDNN-F) were introduced as a TDNN improvement [3] and proved to be effective for children ASR [86]. They differ from TDNNs by applying a singular value decomposition (SVD) on the weight matrices W , which factorize them into two smaller rank matrices:

$$W = U\Sigma V^T = MN \quad (2.14)$$

Where U is a $m \times m$ complex unitary matrix, Σ a $m \times n$ non-negative rectangular diagonal matrix, V a $n \times n$ complex unitary matrix, M a $m \times k$ real matrix and N a $k \times n$ real matrix. With a suitable value of k (much small than m and n), this decomposition preserves the capabilities of the model, reduces the number of parameters and acts as a bottleneck layer (as shown in figure 2.9(b))

2.3.3.B End-to-end models

More recently, the ability of end-to-end models to outperform the hybrid HMM-DNN models in terms of absolute WER for children has been demonstrated [87–90]. Nonetheless, a fundamental issue for this type of model is the requirement for a large amount of data to function correctly. Therefore, in most cases, training reliable end-to-end model for children from scratch is difficult. As a result, different training approaches are required, such as transfer learning (which will be detailed in further detail in section 2.3.5.A) or semi and self-supervised learning, when there are few or no transcriptions, respectively [91].

2.3.4 Data augmentation

Deep learning's success is mostly due to its capacity to effectively utilise massive amounts of data to recognise patterns and be robust to variances. As a result, the lack of data on children's speech plays a significant part in performance deterioration as compared to adults. Data scarcity for children's speech is especially problematic for languages other than English, for which fewer resources are accessible in general. As a consequence, the need for large amounts of data has inspired research into a variety of data augmentation approaches, the goal of which is to artificially increase the amount of data for training. There is two way of conducting this augmentation, according to the literature: using solely the data that is currently accessible or incorporating external data.

2.3.4.A Using external data

The use of out-of-domain adult data has been shown to be effective in improving the automatic recognition of children’s speech [92,93]. In particular, improvements were observed using adult female speech, as the acoustic mismatch between females and children is smaller than adult men. Similarly, [94] proposed to augment using directly additional children data.

Furthermore, several studies proposed leveraging new data generated by deep learning systems as additional data during training. This is done in the literature using two families of algorithms: generative adversarial network (GAN) and text-to-speech (TTS).

Generative Adversarial Networks are generative models that generate new data instances that are similar to the data on which they were trained. For that purpose, the GAN model is separated into two parts: a generator, which generates the synthetic data, and a discriminator, which receives either the created or actual data sample and attempts to differentiate between the two. During training, each module competes with the other, with the generator attempting to trick the discriminator and the discriminator attempting not to be fooled. A GAN was employed in [95] to transform children’s speech to adult speech in order to reduce variability. This was accomplished by feeding the generator children’s utterances and utilising adult data as real data. In this approach, the discriminator seeks to differentiate between altered children’s speech and real adult speech. At inference time, the discriminator is removed, leaving just the generator to turn children’s speech into adult speech.

Text-to-speech systems aim to generate speech examples using directly the transcription as input. With recent advance of TTS systems such as Tacotron2 [96] or VITS [97], more realistic speech utterances are produced. Naturally, some research tried to use TTS outputs as data augmentation for ASR task [98] since the transcription and speech data are available. However, children’s speech contains more complex traits than adults such as substandard or unclear pronunciation in addition to acoustics variability. As a result children’s TTS quality is often inconsistent. In consequence, [99] proposed different data selection strategies such as speaker embedding similarity between the reference speaker and the speaker embedding extracted from generated speech utterance. With this data selection strategy, they have significantly reduced the CER for different children’s speech recognition tasks.

2.3.4.B Using available data

A common procedure for enhancing the robustness of the model by leveraging existing data is to construct a copy of the original data with different noises (such as babel noise, white noise, and music) and reverberation added [100–104]. More recently, vocal tract length perturbation (VTLP) [105] and spectral augmentation (SpecAugment) [4] became popular data augmentation techniques, in particular for end-to-end systems. They create a new copy of data by warping the frequency axis. SpecAugment also randomly masks time and frequency bands of the original audio at the feature level as shown in figure

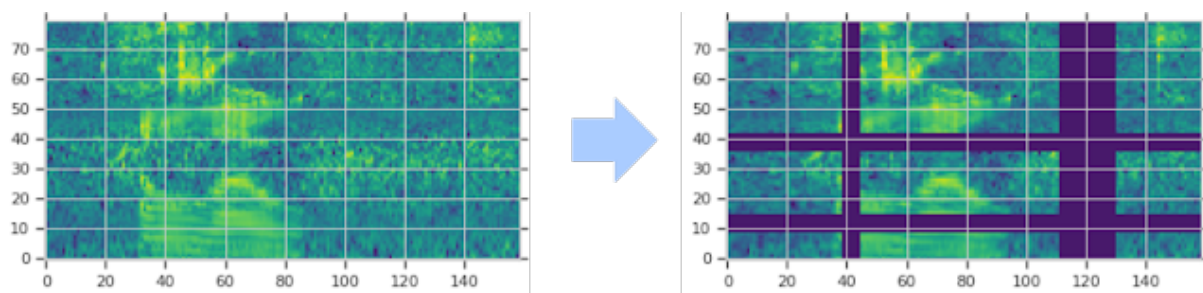


Figure 2.10: Before-After specaugment augmentation with warping of the time and time steps and Mel frequency masking (figure from [4])

2.10.

Finally, as mentioned in section 2.1.2, children's speech contains many disfluencies and errors which can complicate the learning process of the ASR model. Therefore, to make the model more robust to such errors, [106] proposed to synthetically create manual reading errors, such as word insertion or substitution by cutting the signal or adding speech elements produced by other children.

2.3.5 Training procedure for children speech recognition

The most typical method for training a deep learning system is to alternate between forward and backward passes. The corpus input is supplied to the network in the forward pass to create a prediction. The loss is calculated using both predicted and actual target values. To reduce the number of prediction errors, network weights are adjusted using the gradient descent technique in the backpropagation phase. The model is considered trained after a number of loops between these two passes. There are, however, variations of this kind of pipeline in the literature such as transfer and multi-task learning, which stand out in the context of children's voice recognition.

2.3.5.A Transfer learning

When presented with a new problem in biological intelligence, people can employ information from a previous task as an inductive bias. Humans may avoid having to learn everything from scratch by transferring knowledge. It may be defined as the capacity to identify and utilise past task knowledge as a starting point for new tasks. In contrast, algorithms in machine learning are generally developed from scratch on a specific task. Transfer learning (TL) or parameters transfer is a concept that has emerged to bridge the gap between artificial and biological intelligence. To do this, the model's parameters are initialised with parameters obtained from another well-resourced model trained on a source-related task. Then, this model is adapted with the data from the new domain by adjusting the parameters to better fit the target task. The resulting model leverages various underlying characteristics that have been captured by the different layers of the neural network during the training on the source domain step. A

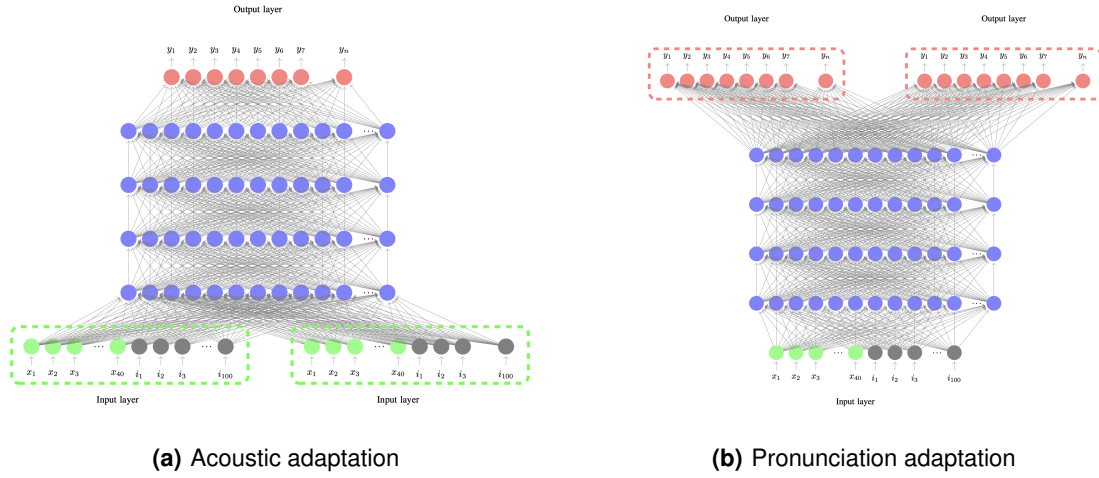


Figure 2.11: Transfer learning approaches. Figures from [5]

common assumption in deep learning is that the bottom layers, closest to the input, capture more signal-specific characteristics. While higher layers, near the output, capture task-specific information [107, 108]. Furthermore, because the target model is based on a pre-trained model, one advantage of TL is that it requires less training data for adaptation.

In recent years, TL has been successfully used in a wide variety of applications, especially for low-resource tasks, such as language understanding [47], character recognition [109] and dysarthric speech recognition [110] among others. These successes have motivated its use for children’s speech recognition. In particular, using large adults corpora as source domain. Indeed, recent acoustic models trained on adults are increasingly efficient and contain many acoustic and phonetic information that can be used for efficient adaptation to children’s speech. Because children’s speech variabilities are in both acoustics and pronunciation, [5] proposed to study three different methods of TL to respectively access the contribution of acoustic adaptation, pronunciation adaptation and the combination of the two.

Acoustic adaptation is based on the well-established idea that lower-level layers capture acoustic properties. In consequence, to solve children’s acoustic variability that affects the lower-level layers, acoustic adaptation consists of freezing the top-level layer’s weights and using TL on the lower-level layers as shown in figure 2.11(a). For [5] and [111], acoustic transfer learning from an adult model for children, by retraining only the first layers of the model led to respectively 38% and 26% WER relative improvements compared to adult models performances. Furthermore, with a 4.9% relative WER improvement, the children’s acoustic adaption outperforms a randomly initialised acoustic model trained on the same children’s data.

Pronunciation adaptation is based on the assumption that higher-level layers capture task characteristic information, which in the context of ASR is pronunciation. Pronunciation is the act or manner of

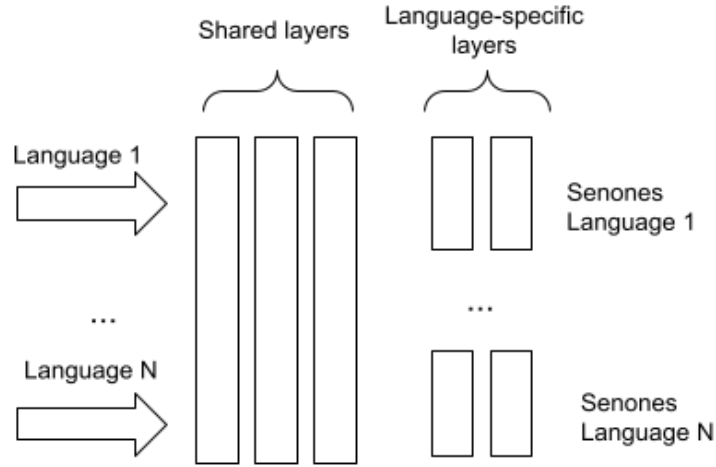


Figure 2.12: Multilingual approach using each language as a task in a multi-task learning context.

pronouncing words using phonetic symbols, phonetic symbols which correspond to the different outputs of the acoustic model. Pronunciation adaptation, as illustrated in figure 2.11(b), adapt higher layer, keeping lower-level layers weight frozen. In [5], pronunciation adaptation on the last layers of the model led to 31% relative WER improvement compared to the adult model's performance. However, as compared to a randomly initialised acoustic model trained with the same children's data, pronunciation adaptation degrades by 5.6% relative WER.

Finally, fine-tuning the entire network outperforms both acoustic and pronunciation adaptation alone. Similar findings have recently been seen in end-to-end models, where TL leveraging an adult pre-trained model outperformed training from scratch using only children's data [87, 88].

2.3.5.B Multi-task learning

Multi-task learning (MTL), like transfer learning, is inspired by biological intelligence. By training all tasks in simultaneously, MTL tries to discover shared representations between related tasks. MTL differs from TL in that it does not limit learning to only the source and target tasks in sequential training, but instead permits learning with as many tasks as possible in parallel. In general, a typical MTL model consists of two different parts. The first part is the sub-network shared by all tasks, while the second is a task-specific output sub-network (see figure 2.12). The joint representation learned in the shared layers is more robust, allowing the model to be more reliable. More formally, for any task i the corresponding output of the forward pass will be:

$$f(X_i; \{M_i, M_c\}) = f_i(f_c(X_i, M_c); M_i) \quad (2.15)$$

where X_i is the data associated with the task i , M_i represents the task-specific parameters of the model, and M_c corresponds to the parameters that are shared (or common) across all tasks.

Therefore, the performances of the MTL greatly depends on the task-relatedness. Indeed, the MTL is sensitive to outlier tasks that are unrelated to the rest of the tasks. This is due to the difficulty to learn common representations for tasks that are unrelated to each other [112].

MTL has been used effectively in a variety of areas, including natural language processing [113], computer vision [114] and bioinformatics [115]. MTL has been naturally applied in the field of automatic speech recognition [116] with directly application to low-resource ASR [117]. Given that child ASR is a resource-limited task, MTL has been proposed to address the issue of data scarcity by utilising various datasets of children’s speech. [111] and [118] successfully applied MTL to Mandarin and English-speaking children, with a 16.96% relative improvement in WER for the English children.

2.4 Children Corpora

As described in chapter 2.1.3, notwithstanding recent efforts to collect databases on children’s speech, there are still fewer data available than for adults. In addition, collecting speech data for children is challenging in many ways, from the child’s attention span, mispronunciations, and ungrammatical expression to the use of out-of-ordinary words. Therefore, the scarcity of publicly available child speech corpora and their small size impeded research and further development of reliable ASR systems for children.

Table 2.1 provides a list of existing corpora of children’s speech. Notably, one-third of the available corpora are in English. Similarly, one-third is focused to children under the age of 4. Furthermore, there is always a trade-off between speaker diversity, total duration, and utterance number.

The remainder of this section will go through the children’s speech corpora that were used in this proposal.

2.4.1 LETSREAD

LetsRead database [135] is a read-aloud speech database of European Portuguese from children aged 6 to 10, from 1st to 4th grade. This corpus is composed of a total of 284 children, 147 girls and 137 boys, whose mother tongue is European Portuguese. Children from private and public Portuguese schools were asked to carry out two tasks: reading sentences and a list of pseudo-words. The difficulty of the tasks varies depending on the school year of the child. For this proposal, we excluded all utterances from the pseudo-word reading task because we do not include pseudo-words in the language model and lexicon in our experiments.

Corpus	Languages	# Spkrs	# Utt	Dur.	Age Range	Date
Providence Corpus [119]	English	6		363h	1-3	2006
Lyon Corpus [120]	French	4		185h	1-3	2007
CASS_CHILD [121]	Mandarin	23		631h	1-4	2012
Demuth Sesotho Corpus [122]	Sesotho	4	13250	98h	2-4	1992
NITK Kids' Speech Corpus [123]	Kannada	160		10h	2-6	2019
CHIEDE [124]	Spanish	59	15444	8h	3-6	2008
CUChild [125]	Cantonese	1986			3-6	2020
EmoChildRu [126]	Russian	100	20000	30h	3-7	2015
CNG Portuguese children [127]	Portuguese	510		21h	3-10	2013
AusKidTalk ¹ [24]	English	750		600h	3-12	2021
PF-STAR-SWEDISH [128]	Swedish	198	8909	6h	4-8	2005
PF-STAR Children British [28, 128, 129]	English	158		14.5h	4-14	2006
AD-child. RU [130]	Russian	278			4-16	2019
TBALL [131]	English	256	5000	40h	5-8	2005
SPECO [132]	Hungarian	72		12h	5-11	1999
UltraSuite [133]	English	86	14456	37h	5-14	2019
CID read speech corpus [134]	English	436			5-18	1996
Letsread ² [135]	Portuguese	284	4629	14h	6-10	2016
CMU kids Corpus [26]	English	76	5180		6-11	1997
CFSC [136]	Filipino	57		8h	6-11	2012
IESC-Child [137]	Spanish	174	19793	34h	6-11	2020
CU Children's read and prompted [138]	English	663	66300		K-G5	2001
Chorec ² [139]	Dutch	400	3065	25h	6-12	2008
ChildIt2 [140]	Italian	96	4875	9h	6-14	2016
TIDIGITS [141]	English	101			6-15	1993
CSLU Kids' Speech Corpus [27]	English	1100	1017		K-G10	2007
SingaKids-Mandarin [23]	Mandarin	255	79843	125h	7-12	2016
ChildIt corpus [142]	Italian	171			7-13	2007
VoiceClass Database [143]	German	170			7-14	2010
Deutsche Telekom telephone [143]	German	106			7-14	2010
Jasmin [144]	Dutch			63h	7-16	2008
Tgr-child corpus [142]	Italian	30			8-12	2007
SponIt corpus [142]	Italian	21			8-12	2007
Swedish NICE Corpus [145]	Swedish	75	5580		8-15	2005
CHIMP spontaneous speech [19]	English	160			8-14	2002
SpeeCon corpus [146]	20 Languages				8-15	2002
Rafael.0 telephone corpus [82]	Danish	306			8-18	1996
Boulder Learning - MyST [22]	English	1371	228874	384h	G3-G5	2019
CU Story Corpus [138]	English	106	5000	40h	G3-G5	2003
ETLT ² [147]	L2 German		1674	6h	9-16	2020
Lesetest corpus [148]	German	62			10-12	2000
FAU Aibo Emotion Corpus [149]	German	51	13642	9h	10-13	2002
PIXIE corpus [150]	Swedish	2885				2003
Takemaru-kun corpus [151]	Japanese	17392				2007
CALL-SLT [152]	German		5000			2014

¹ At the day of this proposal, data collection for this dataset is not complete.

² Information displayed here correspond to a subset of the original data used in this proposal.

Table 2.1: Non-exhaustive comparison of children's speech corpora. This table has been sorted by age range. Blanks indicate unavailable information. Entries highlighted in bold correspond to the corpora used in the experiments presented in this PhD thesis proposal. K: Kindergarten. G: Grade

2.4.2 PFSTAR_SWEDISH

The PFStar children’s speech corpus [128] was collected as part of the EU FP5 PFSTAR project. It contains more than 60 hours of speech. This corpus is divided into two parts: native-language speech and non-native language part. The native-language speech part contains recordings of British English, German and Swedish children, from 4 to 14 years old. The non-native language part consists of speech by Italian, German and Swedish children speaking English. In this work, we only used the native language Swedish part, consisting of speech by 198 native Swedish children, between 4 and 8 years old recorded in the Stockholm area, imitating an adult who read the text from a screen.

2.4.3 ETLTDE

Extended Trentino Language Testing (ETLT) corpus [147] has been collected in northern Italy for assessing English and German proficiency of Italian children between 9 and 16 years old, by asking them to answer questions. The data collection was carried out in schools. On average the signal quality is good, but some background noise is often present (doors, steps, keyboard typing, background voices, street noises if the windows are open, etc). In addition, many answers are whispered and difficult to understand. For this proposal, we only used the German-transcribed subset (named ETLTDE), around 6h divided into training and test partitions.

2.4.4 CMU_KIDS

The CMU kids corpus [26] contains English sentences read aloud by children, 24 males and 52 females, from 6 to 11 years old. In total, 5,180 utterances were recorded with one sentence per utterance. This database was created to train the SPHINX II [153] automatic speech recognition system within the LISTEN project at Carnegie Mellon University (CMU).

2.4.5 CHOREC

The Chorec corpus [139] consists of 400 Dutch-speaking elementary school children, between 6 and 12 years old, reading words, pseudo-words and stories. The difficulty of the reading task was adapted to children with 9 different levels. Recordings were made in schools, leading to some environmental noises (school bells, children entering the playground etc.). For this proposal, similarly to the LETSREAD dataset, we discarded pseudo-word utterances.

2.4.6 MyST

My Science Tutor (MyST) Children Speech Corpus [22] is currently one of the largest publicly available corpora of English children’s speech, with around 400 hours. This is about 10 times more than all other English children’s speech corpora combined. It consists of conversations between children and a virtual tutor in 8 scientific domains. Speech was collected from 1,372 students in third, fourth and fifth grades. Partitioning of the corpus is already available, ensuring reasonably representation of each scientific domain and that each student is present in only one partition. However, only 45% of the utterances were transcribed at the word level. Furthermore, for the purposes of this proposal, we decided to remove all utterances shorter than one second and longer than 30 seconds. After this filtering, 81971 utterances from 736 speakers for a total of 151 hours remain.

2.5 Summary

In this chapter, we provided a brief overview of children’s automatic speech recognition, its challenges, and the research community’s response. Indeed, ASR in children has long been acknowledged as challenging. This is mostly due to the development of the vocal apparatus, which produces an acoustic gap with adult speech; nonetheless, this mismatch diminishes until the age of 15. Despite all efforts, it remains an active and challenging area of research. In this chapter, we also discussed inductive bias approaches, such as transfer learning and multi-task learning, which would be applied in the work detailed in this proposal. Furthermore, we present the most complete comparison of children’s speech corpora to our knowledge.

3

Hybrid models for children automatic speech recognition

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3.1 Introduction

In Chapter 2, we saw how a HMM-GMM or HMM-DNN ASR system may integrate knowledge into the automatic speech recognition pipeline. It uses knowledge from language model, acoustic model, and vocabulary to reduce the amount of speech data required to generate appropriate results. According to the literature, these results can be further improved with inductive bias approaches such as transfer learning and multi-task learning [111]. For years, hybrid configurations have been a privileged setting for the children’s ASR community. As a matter of fact, between 2009 and 2020, 80% of published research on children’s speech recognition was based on hybrid systems, with 45% using HMM-GMM and 35% HMM-DNN. However, during the same period, 63% of published work was conducted for English [154]. As a result, it is uncertain how children’s speech from other languages relates to the various approaches used in English, particularly transfer and multi-task learning.

This chapter will investigate these strategies in a variety of scenarios employing non-English data. First, we present transfer and multi-task learning using adult speech as an inductive bias, as is common in the literature. Second, because there is a lack of data for both adult and children’s low-resource languages, we investigate the same methodologies using only children’s speech. Finally, we present our approach, multilingual transfer learning, which combines transfer and multi-task learning to produce a more robust model for speech recognition in low-resource children setting.

3.2 Multi-task and Transfer learning using adult and children data

3.2.1 Methodology

Motivated by the success of knowledge transfer approaches for ASR children using adult data in the research [5, 111, 118], we intend to validate these findings using a low-resource language. Indeed, using adult data for pre-training makes sense since adult speech is more stable and less prone to variation. Using adult speech to train a speech recognition algorithm makes it simpler to extract and recognize intrinsic and meaningful speech patterns.

For this proposal, we assess children’s speech recognition performances in four distinct configurations:

1. **Adult model:** Using a model trained from scratch with only adult data.
2. **Children model:** Using a model trained from scratch with only children data.
3. **Multi-task model:** Using a model trained jointly on adult and children data in parallel using multi-task learning.

4. **Transfer learning:** Using a model that has been fine-tuned on children data from the adult model of configuration 1.

3.2.2 Corpus

As stated in the introduction, we aim to evaluate the performance of children’s speech in a low-resource language. To this end, we decided to use European Portuguese corpora. European Portuguese can be considered a low-resource language since most adult speech corpora do not exceed 100 hours [155]. In this experiment, we used LetsRead, a child corpus, described in section 2.4 and BD-PUBLICO as adult corpus. The statistics of all these two corpora are provided in the following table 3.1. The rest of this section provides further information about the BD-PUBLICO corpus.

Corpus name	Train	Test
BD-PUBLICO	8085 utt	412 utt
<i>Adult</i>	21h48	01h10
LETSREAD	3590 utt	1039 utt
<i>Children</i>	12h00	02h30

Table 3.1: Number of utterances and duration of the different corpora for multi-task and transfer learning experiments using adult and children data

BD-PUBLICO

The BD-PUBLICO database (Base de Dados em Português eUropeu, vocaBulário Largo, Independente do orador e fala COntínua) [156] consists of reading sentences extracted from Portuguese newspaper PÚBLICO. The sentences that are read correspond to a total of 6 months of news (equivalent to 10M words and 156k different forms). It is composed of 120 speakers, and graduate and undergraduate students from Instituto Superior Técnico (Lisbon). This corpus is considered an adult dataset since all students are between 19 and 28 years old. All recordings were performed in good noise condition, in a soundproof room at INESC-ID (Lisbon), at a sampling frequency of 16kHz and using a high-quality microphone. In addition, a pronunciation lexicon with citation phonemic transcriptions for each word was produced. Finally, manually corrections were applied to the automatically generated transcriptions.

We divided the BD-PUBLICO corpus into three unique sets with balanced gender partitioning: 1) A training set of 80 sentences by 100 speakers. 2) A development set of 40 sentences performed by a total of 10 speakers. Finally, a test set of 40 sentences by 10 speakers.

3.2.3 Experimental setup

All experiments were carried out using the Kaldi open-source toolkit [53]. First, for each corpus, an independent HMM-GMM acoustic model was trained to produce the necessary alignment for the HMM-DNN model. Then, HMM-DNN acoustic models were trained using 40-dim filter-banks (fbanks) in addition to a 40-dim Spectral Subband Centroid (SSC) features [157]. These features are known to have similar properties to formant frequencies. Thus, we expect them to help vowel recognition and lead to better recognition of children’s speech. The resulting 80-dim input features are then augmented by a 100-dim i-vector. Concatenating speaker embeddings to the input features helps to improve model speaker robustness [80]. For our experiments, we use an i-vector extractor trained on a set of pooled children data from different languages.

Data augmentation was applied to all training corpora by perturbing the speaking rate of each training utterance by 0.9 and 1.1 factors; as well as volume perturbation. This helps the network to be more robust to rate and volume variability on the test sets. To further improve the robustness of the model, SpecAugment [4] was applied on top of the fbanks and SSC features by randomly masking time and frequency bands.

For all experiments, we kept the same HMM-DNN acoustic model architecture using lattice-free maximum mutual information (LF-MMI) objective with a learning rate of $2.0E-4$. The acoustic model architecture is divided into two parts: i) six convolutional neural network layers and seven TDNN-F layers of dimension 1024 and followed by ii) two TDNN layers of dimension 450 and a fully-connected layer.

For the transfer learning experiments, only the first part of the network will be fine-tuned, while the second part will be dropped and replaced by randomly initialized ones. Similarly, for the multi-task learning experiment, the first part will be shared between the adult and the child, while the second part will be independent.

3.2.4 Results

Method	Adult WER ↓	Children WER ↓
Adult model	3.82%	102.83%
Children model	45.56%	26.88%
Multi-task model	4.59%	27.65%
Transfer learning	-	25.36%

Table 3.2: WER results using adult data for knowledge transfer methods

The WER scores for all settings are presented in table 3.2. In the first row, we notice that employing a model trained on adult data yields a WER of 102.83% on the children’s test set. This model achieves 3.82% for BD-PUBLICO. This degradation in the children’s compared to adults’ scores demonstrates

the presence of considerable variability in children's speech, which has a detrimental impact on the ASR scores. It supports the idea that an acoustic model designed exclusively for children is necessary because child speech is currently unusable with adult systems.

Training the acoustic model directly on the children's data, on the other hand, considerably improved the word error rate on the children's data to 26.88%. Since the model observes acoustic variability during training, it becomes more robust to it. While the model improved for children, it deteriorated adult speech recognition performance to 45.56 % WER. This confirms the acoustic mismatch between adult and children speech once more. We compare transfer and multi-task learning approaches using these two experiments as a baseline.

When the model was trained jointly utilising adult and children data in the scenario of multi-task learning, the recognition score of the adults and children decreased marginally when compared to the adult and child model baselines. Unlike in the adult and child models, where the mismatch significantly reduced the children's score in the adult model and the adult's score in the child model, both recognition scores in this multi-task learning scenario are comparable to their respective "trained from scratch" baselines. These results were achieved by including corpus-specific layers into the acoustic model architecture. Indeed, the model's shared component will learn the key characteristics of Portuguese speech, while the corpus-specific part will focus on how to apply them to adults and children, respectively.

In the fourth line, Training over children data with a pre-trained Portuguese adult model as initialization enhanced the result to 25.36% WER. When compared to weights random initialization, it is shown that the weights of the adult model are a beneficial starting configuration and allow the transfer learning model to learn relevant patterns for children. It avoids the need for the model to learn these patterns from scratch, using data from a highly variable source. As a result, transfer learning may be considered a viable strategy for improving the ASR performance for children's speech. This finding is consistent with the literature on hybrid models [5, 111].

3.2.5 Summary and discussion

In this study, we conducted a knowledge transfer technique analysis to improve the results of ASR systems for children. We corroborate the acoustic mismatch between adult and child speech and the importance of the model encountering child data and its variability. Our investigations revealed that the transfer learning approach is a promising way to improve low-resource children's speech recognition scores. Furthermore, multi-task learning was found to be helpful in the setting of mixed adult-child ASR acoustic modelling. However, in this study, we focus on the transfer from adults to children. It is not clear how such a system can work using only children's data.

3.3 Multi-task and transfer learning using multilingual children data

3.3.1 Motivation

In this section, we study whether the performance of children’s ASR for low-resourced languages may be improved by combining children’s resources from different languages. In many cases, there is limited or no data for both adults and children. Therefore, we propose using several small-sized corpora of children from various languages to overcome the substantial acoustic variability and data scarcity issues. The current study extends standard multilingual training and transfer learning for hybrid HMM-DNN ASR by combining them in a meaningful way to use knowledge from heterogeneous data. First, a multilingual model trained with a multi-task learning objective tries to optimise network parameters to the specific characteristics of children’s speech in multiple languages/tasks simultaneously. Subsequently, this multilingual model is used to improve ASR for a target language –potentially different from those used in the multilingual training stage– by using transfer learning. We address the following research question: Does this two-step training strategy outperform conventional single language/task training for ’s speech, as well as multilingual and transfer learning alone?

3.3.2 Proposed approach

We propose to combine transfer learning (TL) and multi-task learning (MTL) together for improved acoustic modelling of hybrid HMM-DNN ASR. The proposed approach consists of a two-stage procedure using both MTL and TL that extends the existing techniques since these are usually applied separately. First, a multilingual model trained with a multi-task learning objective attempts to optimize the network parameters to the particular characteristics of children’s speech in multiple languages in parallel. In this work, the model is considered multilingual because all the tasks trained during multitask learning are a corpus of children from different languages. Secondly, we adapt this model for a specific children’s corpus with TL. The motivation for using TL as a second stage is to take advantage of the robust pre-trained model trained during the MTL phase. Indeed, this pre-trained model has potentially learned cross-linguistic information about children’s speech but has also seen more children’s data than a model trained in a single language. For this purpose, the acoustic model is divided into two parts: the layers close to the input are shared across all languages and the top layers are language-specific. That is, there are as many output layers as there are languages, i.e. children corpora. Notice that one can incorporate a new language/-task in this second stage by adding a new language-specific output, even if this new language/task has not been seen during MTL training (figure 3.1). Our hypothesis is that the more data has been seen by the acoustic model, the better the shared layers can capture the underlying characteristics of children’s speech during the first stage of the procedure. These characteristics can be used effectively, later, by the language-specific layers and during the second step of the procedure (figure 3.1).

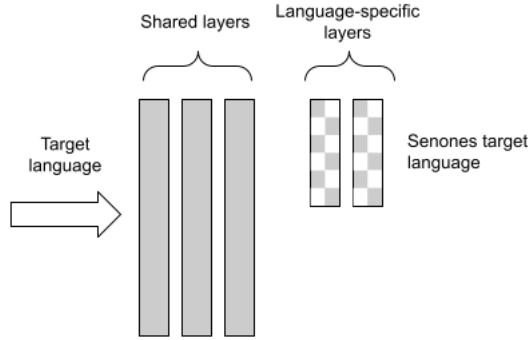


Figure 3.1: Multilingual transfer learning approach. Language-specific layers can be randomly initialized for a language not present during the MTL phase or use the corresponding pre-trained layers in case the target language was present during the MTL phase. Grey blocks are pre-trained during MTL phase.

Although the approaches adopted in this work have been used previously in other studies, for instance [111] and [118] where they successfully applied MTL using children speaking Mandarin and English, obtaining a relative improvement of 16.96% WER in the English children case, it is clear that successful performance of a methodological approach in the case of English cannot be expected to generalize to other contexts and languages. As we all know, English is a large-size, resource-rich pluricentric language which should be seen more as an exceptional case, rather than an average representative. Against this background, it is important to emphasize that there is a need for research that investigates whether methods that have already been tested for English also work in new contexts such as those of mid-sized languages with fewer resources than English, like Dutch, Portuguese, Swedish and German.

3.3.3 Setup

All experiments were conducted using five children corpora, each from a different language. Namely PFSTAR_SWE, ETLTDE, CMU, LETSREAD and CHOREC. All those datasets have been described in section 2.4. Table 3.3 presents statistics about the duration, number of utterances and language. Notice that in this work we have only used small datasets to better reflect the average size of the available children’s speech corpora.

We employ the same experimental design as the prior experiment with adult data from section 3.2.3 setup, where the acoustic model is divided in two. The first part is shared across all languages, whereas the second is language specific. Furthermore, each corpus, i.e. each language, uses an independent language model and lexicon that is constant throughout all experiments in order to assess solely the acoustic model contribution.

Corpus name	Language	Train	Test
PFSTAR_SWE	Swedish	6030 utt 04h00	2879 utt 01h48
ETLTDE	L2 German	1445 utt 04h41	339 utt 01h06
CMU	English	3637 utt 06h26	1543 utt 02h45
LETSREAD	Portuguese	3590 utt 12h00	1039 utt 02h30
CHOREC	Dutch	2490 utt 20h12	575 utt 04h42

Table 3.3: Statistics on the different corpora of children's speech.

	PFSTAR_SWE	ETLTDE	CMU	LETSREAD	CHOREC
Language	<i>Swedish</i>	<i>German</i>	<i>English</i>	<i>Portuguese</i>	<i>Dutch</i>
Single language	54.36%	44.69%	21.26%	26.88%	25.15%
MTL	54.95%	42.46%	23.01%	27.45%	25.10%
TL from PFSTAR_SWE	-	42.23%	20.62%	26.47%	24.65%
TL from ETLTDE	53.60%	-	20.90%	26.61%	25.42%
TL from CMU	52.83%	41.54%	-	26.49%	24.58%
TL from LETSREAD	52.50%	41.77%	20.41%	-	24.60%
TL from CHOREC	52.20%	40.28%	19.77%	26.05%	-
TL Average	52.78%	41.46%	20.43%	26.41%	24.81%
TL Best	52.20%	40.28%	19.77%	26.05%	24.58%
MLTL	51.67%	38.04%	19.33%	25.75%	23.78%
MLTL-olo	51.58%	40.05%	19.67%	26.20%	24.57%

Table 3.4: WER results of multilingual-transfer learning and cross-lingual experiments. MTL: Multi-Task Learning, TL: Transfer Learning, MLTL: Multilingual Transfer Learning, MLTL-olo: Multilingual Transfer Learning one-language-out

3.3.4 Multilingual-transfer learning experiment

Table 3.4 presents the WER results of the multilingual transfer learning (MLTL) approach compared to three different methods: baseline, trained on each corpus individually for 4 epochs; Multi-task Training (MTL) alone, trained jointly using all corpora for 4 epochs ; Transfer Learning (TL) alone, adapted for the target language using in turn one of the other 4 baseline models as a source, leading to 4 results per target language. In addition, for clarity, we summarise the transfer learning scores with the average of the 4 scores and the best of the 4 for each target.

Firstly, it is important to emphasise that the baseline scores correctly reflect the different tasks the children were asked to perform and the corresponding amount of data available for each corpus. The best WER score, 21.26% for CMU, can be explained by the reading-aloud-sentences task nature of this corpus. Thus, the language model can more easily compensate for the acoustic model errors. In addition, Chorec and LetsRead, as the largest corpora in our experiment, also yield relatively good results for children's speech recognition. On the other hand, ETLTDE and PFSTAR_SWE show the

worse WER results with 44.69% and 54.36% WER, respectively. This can be explained by the amount of data available and by the language model which does not compensate as much as the CMU model. Especially for ETLTDE, since it is the only corpus that does not contain scripted text, but spontaneous responses. In addition, the age range of PFSTAR_SWE children also plays a critical role in performance, since younger children generally yield worse performance scores [5].

Turning to multi-task learning, among all the approaches presented, only MTL fails to improve the baseline performance for almost all languages, which is in contradiction with [111]. However, it can be explained by the differences in terms of the size of the child’s speech corpora used. The smaller the size of the corpora used, the more difficult it is to model the acoustic variation in the children’s speech.

Concerning TL, all performance scores outperform their corresponding baseline, confirming that TL is an adequate method for children’s ASR since it allows the system to be confronted with more children, thus with more variation. Precisely, table 3.4 shows that the best pre-trained model for knowledge transfer is Chorec. This makes sense since Chorec is the largest corpus, representing about 40% of the total data used in our experiments.

Finally, MLTL shows an average relative improvement in WER of 7.73% compared to the baseline, slightly higher than the average (TL Avg) and the best (TL Best) transfer learning performance, with an average relative improvement of 4.50% and 2.66%, respectively.

The strength of MLTL is that it can benefit both from MTL and TL, minimizing some of their associated weaknesses. Attending to our results, MTL does not improve single language training. We believe that the unbalanced amount of data, the significant differences among data sets and the use of segmental optimization (lattice-free MMI) can partially explain these results. Nevertheless, we hypothesize that the multi-task objective leans the network towards better optimization of the lower layers, rather than optimizing the upper language-specific layers, can still be beneficial for TL. Regarding TL, one can observe considerable performance variations depending on the pre-trained model used as the source model, probably due to a poorer initialisation of lower layers that is less efficient for TL. The MLTL experiments show that we can overcome these drawbacks by combining both MTL and TL, thus, validating the effectiveness of this approach for robust speech recognition of children.

3.3.5 Cross-lingual validation

In the previous section, we saw that the MLTL approach yields better results than separate multi-task and transfer-learning frameworks.

To further validate the hypothesis that the shared lower layers are able to learn meaningful information about children’s speech characteristics, regardless of the language, we perform a cross-language experiment following a leave one-language-out cross-validation setting. In this experiment, we keep one language out of the multi-task training and use it only during the TL phase to adapt the acoustic model

parameters.

We repeated this procedure for each corpus in our experiment. As in the previous experiment, we used 4 epochs for each learning phase. The last row of Table 3.4 presents the results of the cross-language experiment.

For all corpora, the MLTL one-language-out (MLTL-olo) approach outperforms the baseline WER score with an average relative improvement of 5.56%. Improvements are more important for the small corpora ETLTDE and CMU, with a relative improvement of 14.88% and 9.07%, respectively. PFS-TAR_SWE does not benefit as much, with only 5.05% relative improvement. This is mainly due to the age differences with the children in the other corpora used in the MTL phase. Indeed, the children in PFSTAR_SWE are much younger (see section 3.3.3 for more details). Therefore, we conclude that the shared layers have learned the underlying multilingual features of children.

It is also interesting to compare MLTL-olo with the results of transfer learning alone. In both cases, the pre-trained models used have never seen the target language data. We observe that the results between the MLTL-olo and TL Best are extremely close, with small improvement with the MLTL-olo, only the best transfer learning model on LetsRead is slightly better than MLTL. This means that during multilingual training the system learned, at least, the best representation of the available children’s characteristics. This is consistent with our hypothesis of the important role of the multilingual training phase in our two-step procedure.

3.3.6 Summary and discussion

In this work, we addressed the following research question: Does the two-step training strategy we propose in the current chapter outperform conventional single language/task training for children’s speech, as well as multilingual and transfer learning alone. Our results provide a positive answer to this question, by showing that the limitations of MTL and TL can be overcome by the multilingual transfer learning approach, even in a low-resource scenario, leading to an average relative improvement of 7.73%. Multilingual pre-training is also beneficial for transfer learning with an unseen language, with an average relative improvement of 5.56%. Multilingual transfer learning thus seems to be an appropriate method to address children’s speech recognition in a challenging context.

3.4 Conclusion

In this chapter, we look at the current state of the art for a Hybrid HMM-DNN speech recognition system for children. We illustrated that transfer learning is the most promising strategy for addressing children’s ASR variability because it makes efficient use of the knowledge contained in the pre-trained source model. A pre-trained model that can be trained on both adults or children. The multi-task learning does

not produce the greatest results alone, but we showed that the shared part of the model is capable of learning relevant information for all tasks jointly. Furthermore, we proposed in this chapter to combine these two approaches in our multilingual transfer learning system. Using the capacity of learning relevant information of the multi-task learning approach and the capabilities of efficient use of pre-existing knowledge from transfer learning.

4

End-to-End children automatic speech recognition

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4.1 Introduction

With the growing popularity of deep learning, numerous successful attempts to apply it to ASR have been made. It was only recently, that end-to-end models have shown their capability to outperform hybrid HMM-DNN systems for a variety of speech recognition tasks, including children's ASR. The major advantage of end-to-end speech recognition systems is the merging of the whole training process into a single neural network that eliminates the possibility of behavioural incompatibilities between modules that have been trained independently. However, because of the problem of children's data scarcity, the application of the end-to-end paradigm for children's ASR is relatively new and has not been extensively investigated [87–90]. In addition, end-to-end models often require more parameters to provide such robustness and flexibility. As a result, training on small datasets becomes increasingly difficult [158].

With the recent increased interest in end-to-end speech recognition, several architectures have been developed, including recurrent neural networks [159] and neural transducers [160]. However, one architecture stands out and consistently provides state-of-the-art results in large-vocabulary speech recognition for both adults and children, the Transformer.

This section dives more into details of the Transformer design as well as the adapter transfer for children ASR, a parameter-efficient transfer for Transformer models that we have recently proposed.

4.2 Transformer models

First proposed in 2017 [6], the Transformer architecture is a sequence-to-sequence encoder-decoder architecture that relies entirely on self-attention, eliminating recurrence, convolutions entirely and vanishing gradient issues. Another notable difference with recurrent neural networks is that the Transformer computes the dependencies between each pair of positions simultaneously, rather than one by one, by encoding the symbol position in the sequence. This, enables more parallelisation, resulting in faster training. Since its release, the Transformer architecture had tremendous impact in various areas, including NLP [47], computer vision [161], and speech [63]. The transformer encoder-decoder architecture is presented in figure 4.1, with c) the encoder and d) the decoder. The encoder's role is to transform an input sequence $X = x_1, \dots, x_T$ into a series of continuous representations $Z = z_1, \dots, z_T$ which are then fed into a decoder. The decoder, constructs an output sequence $Y = y_1, \dots, y_N$, one element at a time. At each time step, the decoder receives the encoder outputs together with the last decoder output, in an auto-regressive manner.

The information about the relative position of the tokens in the sequence is given by the summation between the input/output embedding and the positional embedding. Although there are many choices

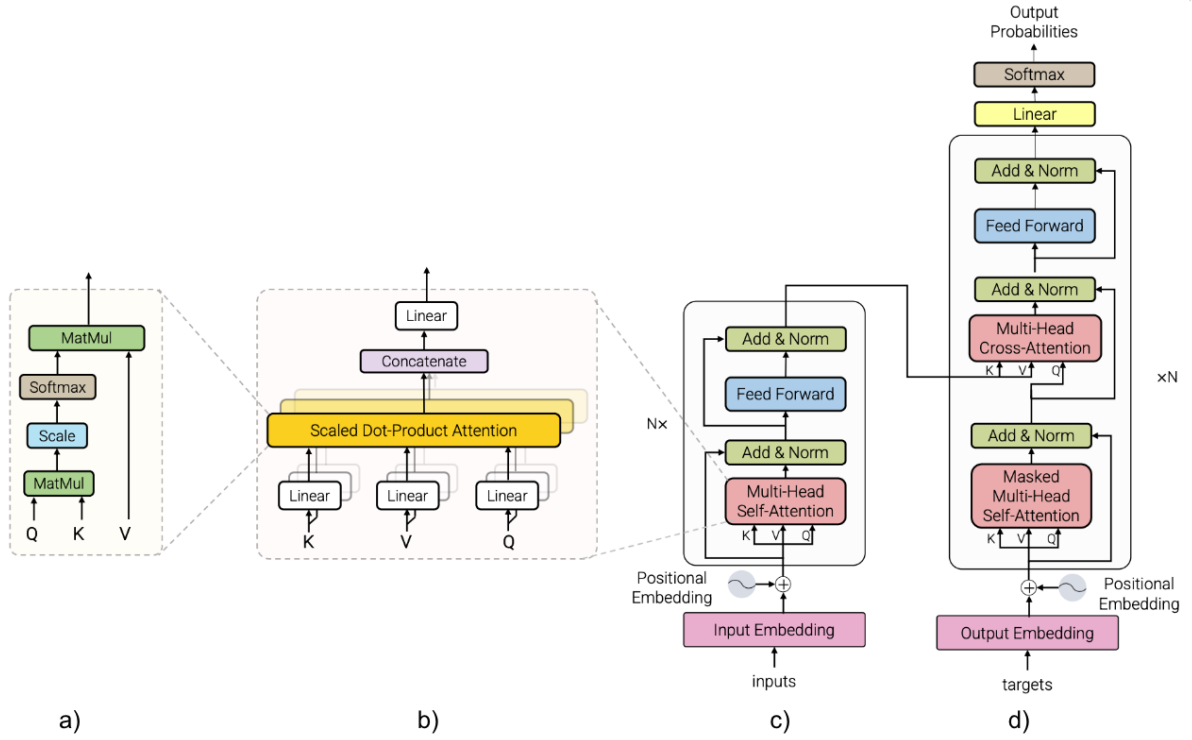


Figure 4.1: Architecture of the standard Transformer [6]. a) scaled dot-product attention, b) multi-head self-attention, c) Transformer-encoder, d) Transformer-decoder.

of positional encodings, [6] proposed to use sine and cosine of different frequencies as follows:

$$PosEnc_{(pos, 2i)} = \sin\left(pos/10000^{2i/d_{model}}\right) \quad (4.1)$$

$$PosEnc_{(pos, 2i+1)} = \cos\left(pos/10000^{2i/d_{model}}\right) \quad (4.2)$$

Where pos is the current token or label position and i is the dimension.

The encoder is composed of a stack of N identical Transformer layers. Each layer consists of a multi-head self-attention module and a feed-forward fully connected neural network module. Each of these modules are followed by a normalization with a residual connection. A multi-head self-attention (MHA) modules relies on the scaled dot-product attention [6], illustrated in figure 4.1.a). Scale dot-product attention focuses on how relevant a particular token is with respect to other tokens in the sequence. And is defined as follows:

$$Attention(Q, K, V) = \text{softmax}\left(\frac{QK^T}{\sqrt{d_k}}\right)V \quad (4.3)$$

Where the input consists of queries Q , keys K of dimension d_k and values V of dimension d_v . The dot product of the query with all keys is each divided by $\sqrt{d_k}$, then passes through a softmax function to

obtain attention weights on the values. When d_k is large, the scaling $\frac{1}{\sqrt{d_k}}$ restrains the dot product from growing large in magnitude.

Instead of performing a single scaled dot-product attention, the MHA module linearly projects h times K, V and Q with different, learned, linear projections to dimensions d_k, d_k and d_v respectively. On each of h projected versions is performed the attention function 4.3 in parallel. The output of each of the h attention function, of dimension d_v is concatenated and projected one last time as pictured in figure 4.1.b). More formally:

$$\text{MultiHead}(Q, K, V) = \text{Concat}(\text{head}_1, \dots, \text{head}_h)W^O \quad (4.4)$$

$$\text{where } \text{head}_i = \text{Attention}(QW_i^Q, KW_i^K, VW_i^V)$$

Where the different projections matrices are $W_i^Q \in \mathbb{R}^{d_{\text{model}} \times d_k}$, $W_i^K \in \mathbb{R}^{d_{\text{model}} \times d_k}$, $W_i^V \in \mathbb{R}^{d_{\text{model}} \times d_v}$ and $W^O \in \mathbb{R}^{hd_v \times d_{\text{model}}}$.

In comparison to the encoder, the decoder contains a third sub-layer, which performs MHA over the output of the encoder using the prior sub-layer of the decoder as the query. The decoder's inputs, which are targets during training and the previously decoded label during inference, are offset by one position. This combined with a modified MHA prevents the attention to use subsequent positions, ensuring that the prediction at time-step i solely depends on the previous $< i$ time-steps.

4.3 Adapters for Transformer based models

Age-dependent acoustic models have shown promising improvements, as children's age is highly correlated with acoustic variability [1, 21]. In particular, some studies found that variability decrease with the age, reaching the adult level at 15 years old [2]. In parallel, research on End-to-end (E2E) architectures has shown equivalent or even superior performance in a large number of speech recognition tasks compared to traditional hidden Markov models approaches [56]. E2E architectures propose to combine different modules of the ASR pipeline into a single deep neural network (DNN), resulting in benefits to avoid error accumulation and mismatch between components. However, for these models to work properly, they need to be trained with a large amount of data, which is not commonly available for children's speech. Thus, to overcome children's data sparsity issue for E2E models training, [87, 88] successfully used transfer learning by fine-tuning an adult pre-trained model on children's speech.

In this work, we propose to apply adapter modules on top of an adult acoustic model as an alternative to the transfer learning strategy for automatic children's speech recognition. Adapters are a method recently proposed for Transformer-based systems that consist of a small set of additional layers that are attached to a source model [7, 162]. Adapters are typically less expensive both in terms of training speed

and storage cost, which is a desirable property in the case of aiming at the development of children’s age-dependent models. In addition, adapters overcome the problem of catastrophic forgetting. Indeed, after using transfer learning, the source model is completely overwritten by the newly trained weights, leading to a drop of performance on the source task. Whereas in adapter transfer, the backbone model remains frozen, thus preserved if adapter layers are removed. Adapters are therefore very practical in the context of small device computing where it can be expensive to load and store a large number of models for adults and children of different ages. Finally, in this work, we also propose a novel version of adapter layers inspired by variational auto-encoders (VAE) [163], so-called variational adapters or Vadapters. We hypothesize that the ability of VAEs to estimate variability can be applied in adapters to make them more suitable for parameter-efficient automatic children’s speech recognition.

4.3.1 Related work

4.3.1.A Transformer model for children ASR

Recently, E2E-based ASR models have demonstrated their ability to achieve state-of-the-art performance on a wide variety of speech recognition tasks [56]. This fact motivated the assessment and comparison of different E2E architectures for children ASR [87, 88]. These works found that Transformer-based architectures, described in the previous section 4.2, yield the best results when an adult pre-trained model is fine-tuned for children using transfer learning with the help of the joint attention and CTC objectives [57]. Usually, these two objectives are combined as follows:

$$\mathcal{L}_{ASR} = \lambda_{ctc}\mathcal{L}_{ctc} + (1 - \lambda_{ctc})\mathcal{L}_{s2s} \quad (4.5)$$

where \mathcal{L}_{ctc} and \mathcal{L}_{s2s} are the CTC and attention losses, respectively. A hyper-parameter $\lambda_{ctc} \in [0, 1]$ is used to control contribution of each loss.

4.3.1.B Adapters

Adapters were first introduced in the NLP field, motivated by the need for a parameter-efficient adaptation to fine-tune large models, like Transformer, on various text classification tasks [162]. They are a simple alternative to full model fine-tuning, as they involve only a small number of newly inserted parameters at each layer of the transformer. While different positions have been proposed [7, 162], they are generally plugged after the feed-forward layer (see Figure 4.2.a). The key idea for training adapters is to freeze the backbone model’s parameters and only update the adapter’s parameters. Adapter modules are based on a bottleneck architecture (projection-down followed by a projection-up) as shown in Figure 4.2.b. Adapters solve a number of drawbacks associated with full model fine-tuning, such as parameter

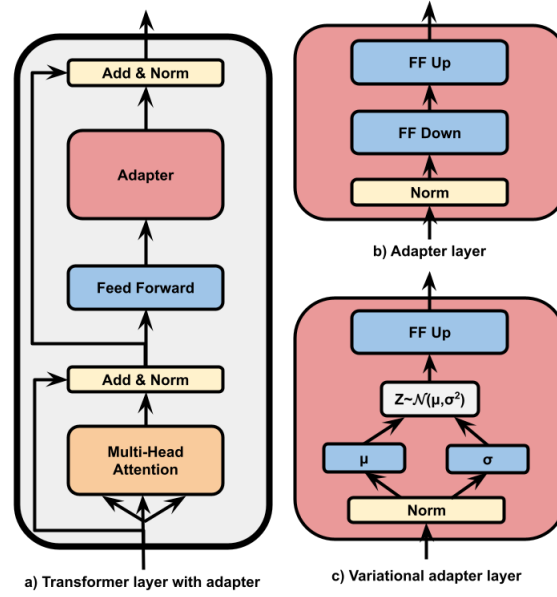


Figure 4.2: a) Example of a Transformer layer with an adapter layer (adapted from [7]); b) Adapter layer; c) Vadapter layer

efficiency, faster learning iterations and a highly modular design

Since it was first proposed, adapters have been successfully used in a wide range of NLP tasks such as language understanding [7] and neural machine translation [164]. Some researchers proposed to use adapters for ASR tasks, such as in multilingual ASR [165]. More recently, [166] studied adapters for atypical speech, in particular pathological and accented speech. More recently, [167] proposed to use adapters inside of self-supervised models for children ASR by refining the whole model together with the weights of the adapters. Our work differs because our aim is to update only the adapters' weights in order to keep both the parameter efficiency and modular properties of adapters.

4.3.1.C Variational Auto-Encoders

Variational auto-encoders (VAE) [163] are a probabilistic generative models, that has been successfully applied in different speech tasks such as transformation [168] and enhancement [169]. The main strength of VAE is their ability to learn a smooth representation of the latent space. Indeed, rather than producing a single value to describe each element of the latent space, as a standard auto-encoder, VAE provides a probability distribution:

$$p_{\theta}(\mathbf{x}, \mathbf{z}) = p_{\theta}(\mathbf{x}|\mathbf{z})p_{\theta}(\mathbf{z}) \quad (4.6)$$

where \mathbf{x} is the observed data generated by a random process using latent data \mathbf{z} and θ denotes the distribution parameters. In this model, the likelihood function $p_{\theta}(\mathbf{x}|\mathbf{z})$ quantifies how the generation of \mathbf{x} is conditioned by \mathbf{z} , while the prior $p_{\theta}(\mathbf{z})$ is used to regularize the latent data \mathbf{z} . Typically, a standard

Gaussian distribution is used for the prior distribution

$$p_\theta(\mathbf{z}) = \mathcal{N}(\mathbf{z}; 0, I) \quad (4.7)$$

while the likelihood is defined as a multivariate Gaussian distribution:

$$p_\theta(\mathbf{x}|\mathbf{z}) = \mathcal{N}(\mathbf{x}; \mu_\theta(\mathbf{z}), \sigma_\theta^2(\mathbf{z})) \quad (4.8)$$

where $\mu_\theta(\mathbf{z})$ and $\sigma_\theta^2(\mathbf{z})$ are obtained using \mathbf{z} . However, since the posterior distribution $p_\theta(\mathbf{z}|\mathbf{x})$ is intractable, it is approximated with the auxiliary distribution $q_\phi(\mathbf{z}|\mathbf{x})$ that plays the role of an encoder:

$$q_\phi(\mathbf{z}|\mathbf{x}) = \mathcal{N}(\mathbf{z}; \tilde{\mu}_\phi(\mathbf{x}), \tilde{\sigma}_\phi^2(\mathbf{x})) \quad (4.9)$$

We also want to ensure that the approximate posterior $q_\phi(\mathbf{z}|\mathbf{x})$ and the true posterior $p_\theta(\mathbf{z}|\mathbf{x})$ are similar by minimizing the Kullback-Leibler (KL) divergence between the two distributions.

$$\min KL(q_\phi(\mathbf{z}|\mathbf{x})||p_\theta(\mathbf{z}|\mathbf{x})) \quad (4.10)$$

It is possible to minimize expression (4.10) by maximizing the following expression as shown in [168]:

$$\mathbb{E}_{q_\phi(\mathbf{z}|\mathbf{x})} \log p_\theta(\mathbf{x}|\mathbf{z}) - KL(q_\phi(\mathbf{z}|\mathbf{x})||p_\theta(\mathbf{z})) \quad (4.11)$$

Where the first term is the reconstruction error and the second term a regularisation.

Thus, the VAE loss function can be define as followed:

$$\mathcal{L}_{VAE} = \mathcal{L}_{recons} + \mathcal{L}_{KL} \quad (4.12)$$

$$= \mathcal{L}_{recons} + \sum_j KL(q_\phi^{(j)}(\mathbf{z}|\mathbf{x})||p_\theta(\mathbf{z})) \quad (4.13)$$

for each dimension j of the latent space.

4.3.2 Variational adapters

Adapters and auto-encoders (AE) share a similar encoder-decoder structure. Although the purpose of these two architectures is different, the role of their encoders is similar: map relevant characteristics of the input into a unique latent vector. On the other hand, their architecture differs in the decoders: AEs use the decoder to reconstruct the input, while adapters project the information contained in the latent vector to be processed by the next layer. Consequently, adapters suffer from the same problems as

AEs, a poor capability to model variability. In order to be more robust to the high variability of children’s speech, we propose to represent each latent value in probabilistic terms. To this end, we propose Vadapter, a new adapter architecture in which the encoder structure of the adapter is replaced with the structure of a VAE’s encoder as shown in Figure 4.2.c.

Consequently, during training, instead of a down-projection that maps the input into the latent representation, we now have two branches, producing the mean μ and variance σ . During inference, μ is used directly as a deterministic latent vector, discarding σ . We hypothesise that this deterministic inference allows Vapters to capture variability in the σ branch while keeping the μ more robust. In addition, dropping the σ branch during inference keeps the number of parameters equivalent to normal adapters, thus preserving the parameter efficiency.

Similarly to VAEs, the regularisation term which ensures that the distribution of $q_i(\mathbf{z}|\mathbf{x})$ for each Vadapter at layer i is similar to the standard normal distribution $p(\mathbf{z})$ is required. However, as there are many Vadapter layers we normalise the sum of all regularization terms by the number of Vadapter layers:

$$\mathcal{L}_{KL_{all}} = \frac{\sum_L^i KL(q_i(\mathbf{z}|\mathbf{x})||p(\mathbf{z}))}{L} \quad (4.14)$$

where L is the total number of Vapters in the model. Then, we inject this regularisation loss into the E2E ASR loss defined in equation (4.5) as follows:

$$\mathcal{L}_{ASR} = \lambda_{ctc}\mathcal{L}_{ctc} + (1 - \lambda_{ctc})\mathcal{L}_{s2s} + \beta\mathcal{L}_{KL_{all}} \quad (4.15)$$

where β is an hyper-parameter to control the regularization’s contribution.

4.3.3 Experiments

4.3.3.A Corpus

Training	Validation	Test
60897 utterances	10044 utterances	4079 utterances
566 speakers	79 speakers	91 speakers
113 hours	18 hours	13 hours

Table 4.1: My Science Tutor Children Speech Corpus statistics

In this work, we decided to use the Boulder Learning My Science Tutor (MyST) corpus, described in section 2.4 given the task assigned to the children, which is to speak spontaneously. Indeed, the end-to-end model encapsulates the acoustic model and language model in the same network. As a result, if we train a model with a restricted amount of prompts on a data set of reading tasks, the model will learn and overfit the prompts. Thus, yielding unreliable results. Furthermore, for the purposes of our

experiments, we decided to remove all utterances shorter than one second and longer than 20 seconds. The details of the filtered corpora used in our work are presented in Table 4.1.

4.3.3.B Implementation details

All experiments were performed using the SpeechBrain toolkit [170]. We used 12 Transformer layers for the encoder and 6 Transformer layers for the decoder, all with dimensions 512. This model has been pre-trained using the LibriSpeech dataset [25] and is publicly available¹. Furthermore, for all of our experiments, we used the same Transformer language model, trained on 10 million words in order to only evaluate the contribution of the E2E model. The adapter architecture consists of a linear layer down-projection from dimension 512 to 256 with a ReLU activation, followed by a linear layer up-projection from dimension 256 back to 512. We analysed the influence of the latent dimension, i.e. the number of parameters, on the ASR performance in more detail in section 4.3.4. The Vadapters σ and μ branches consist of a linear layer down-projection from 512 to 256 each, while the up-projection remains the same as the aforementioned traditional adapters. For all the experiments, models were trained with a batch size of 16, $\lambda_{ctc} = 0.3$ and the same decoding hyper-parameters. All models were trained with a learning rate of $1.5e-4$, for 40 epochs. Finally, for all our Vadapter experiments we choose $\beta = 1.0$.

4.3.3.C Experiments description

In our first experiment, we will attempt to determine which component of the transformer model is most important to ASR children. As a result, this information will be used to determine the best location of the adapters in the transformer layer. Indeed, the adapter should come after the most important component since it will project the output of that component into the expected transformer space. In order to do this, we studied the role of each transformer layer sub-module by fine-tuning one or two of them with the children’s speech data.

Secondly, we analyze the performance of adapters in three scenarios: i) Adapters in all layers of the E2E model, ii) adapters only present in the encoder layers, and iii) adapters only in the decoder layers. These experiments are motivated by the fact that the encoder is closely related to the acoustics generating a high-level representation of speech, while the decoder generates output tokens related to the linguistic domain. The objective is then to evaluate which components, the encoder (acoustics) or the decoder (linguistics), benefit more from the adapter transfer. In order to compare our new architecture with traditional adapters, we reproduce the three scenarios mentioned above by replacing the adapters with our Vadapters. Furthermore, we evaluate the combination of Vadapter and traditional adapter in two scenarios, Vadapter in the encoder and adapter in the decoder, and vice versa.

¹ <https://huggingface.co/speechbrain/asr-transformer-transformerlm-librispeech>

4.3.4 Results

4.3.4.A Transfer learning experiments

Fine-tuned part	WER ↓	Trained parameters
None	25.04%	-
Full model	13.50%	71.5M
Norm	18.08%	57.9K
MHA	13.40%	25.2M
FFN	12.57%	37.8M
MHA + FFN	12.78%	63.0M
Norm + FFN	12.92%	37.9M
Norm + MHA	13.52%	25.3M

Table 4.2: Results of the fine-tuning on part of the model only

Table 4.2 shows results of the transfer learning on sub-modules of the Transformer model. Fine-tuning all the transformer’s parameters, in the same way as the previous work [87, 88], gives better results than using the model trained only on adult data with 13.76% compared to 25.04% WER respectively. The fine-tuning of all normalisation weights improved the score compared to the adult model with 18.04% but still under-perform compared to the full fine-tuning. Thus, the normalisation contribution in the children’s transfer learning is limited. In contrast, fine-tuning the MHA or FFN yields better, results compared to the full transfer learning with 13.40% and 12.57% WER respectively. While always outperforming a full model update, the use of transfer learning on a combination of different model components reduces performance when compared to FFN alone. Transfer learning becomes more difficult by updating the weights of all components of the transformer as well as the non-transformer weights (i.e., Convolution blocks and embedding blocks), which explains why the entire fine-tuning produces worse results. In conclusion, FFN modules are the most relevant to fine-tune using transfer learning. This is because transformer feed-forward layers are key-value memories [171], where each key correlates with patterns in the training examples, and each value produces a distribution over the outputs. Consequently, adapters should be placed after the FFN sub-modules in order to achieve better results. This is consistent with Pfeiffer’s work for NLP tasks [7].

4.3.5 Adapters and Vadapters results

4.3.5.A Adapters for children ASR

Table 4.3 presents the word error rate (WER) results of the different approaches. Firstly, the pre-trained Transformer adult model without any adaptation gives the worst result, with a WER of 25.04%, while adult performances on Librispeech corpus are usually less than 6%. This result shows the impact of the variability in child speech. Secondly, the adaptation of all 71.5 million parameters for children’s speech

Method	WER	Trained parameters
No fine-tune	25.04%	-
Fine-tune	13.50%	71.5M
Adapter	14.33%	4.8M
Adapter encoder only	14.56%	3.2M
Adapter decoder only	20.10%	1.6M
Vadapter	14.19%	7.1M (4.8M)
Vadapter-enc + Adapter-dec	14.05%	6.3M (4.8M)
Adapter-enc + Vadapter-dec	14.35%	5.5M (4.8M)
Vadapter encoder only	14.51%	4.7M (3.2M)
Vadapter decoder only	20.23%	2.4M (1.6M)

Table 4.3: Results of the different approaches; In parenthesis are shown the number of parameters needed for inference after dropping the σ branch.

resulted in a considerable improvement, with 13.50% WER. This result correctly reflects the state-of-the-art performance obtained in the literature for the MyST corpus [88]. Regarding adapters, similarly to previous work in NLP [162] and ASR [166], we observe that they perform slightly worse than fine-tuning, with a score of 14.33% WER. However, it is important to note that adapters require less than 10% of all parameters of the full fine-tuning. We also investigate adapter transfer for encoder and decoder only. Table 4.3 shows that adapters are more relevant when plugged into the encoder with 14.56% WER while compared to the decoder with 20.10% WER. This result confirms that acoustic variability plays a critical role in the degradation of children ASR performance [172].

Additionally, we also evaluated how different adapter hidden-dimension, i.e. the number of parameters, influence the speech recognition performance compared to the fine-tuning model. Figure 4.3 displays the relative WER delta over the ratio of trainable parameters compared to the fine-tuned model. As a reference, the relative WER delta of the adult model with respect to fine-tuning is 85.5%. We observe that the performance difference between fine-tuning and adapters decreases as the number of trainable parameters increases. While with only 2% of trainable parameters, adapters manage to surpass by a large margin the source model performance, adapters need a minimum amount of parameters to get close to the fine-tuning performance. Nevertheless, adapter transfer outperforms full fine-tuning, when the number of parameters used is around 30% of the number of the entire model. There is therefore a trade-off between performance and parameter efficiency. A similar observation has been made in [167].

4.3.5.B Variational-adapters

Concerning the Vadapter architecture, with the exception of the cases where the Vadapters are placed in the decoder, all the scores are higher than their conventional counterpart, approaching the full fine-tuning score. The best configuration, Vadapter in the encoder and adapters in the decoder, reaches 14.05% WER. In the same way, as for the conventional adapters, we can see that the Vadapters are

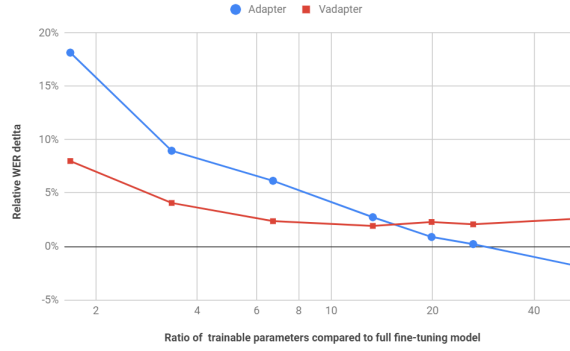


Figure 4.3: Relative WER delta over the ratio (%) of trainable parameters compared to full fine-tuned model.

more advantageous when placed in the encoder since the score is 14.51% WER for Vadapters in the encoder only and 20.23% WER for the decoder only. We believe that this is because Vadapters are designed to be more robust to acoustic variability, which is mainly present at the encoder level. Thus, the Vadapters in the decoder does not manage to improve the score of their conventional counterpart.

Thus, we tested several possible combinations between Vadapter and adapters. We observe that the configurations with Vadapters in the encoder are giving the best results, with 14.19% WER for Vadapters in both the encoder and the decoder, as well as 14.05% for Vadapters in the encoder and adapters in the decoder. However, when Vadapters are placed in the decoder in combination with adapters in the encoder the result is not as good as adapters everywhere with 14.35% WER. We believe again that this is due to the variability being more present in the acoustic than in the linguistic component.

Finally, as shown in Figure 4.3, Vadapter outperforms conventional adapters when the number of parameters is less than 15% of the full model. Indeed, the Vadapters are always under 10% relative WER delta compared to full fine-tuning and reach under 5% with less than 4% of the ratio of trainable parameters, where conventional adapters start above 15% relative WER delta and need more than 10% of the ratio of trainable parameters to be under 5% relative WER delta. These results confirm the proposed Vadapter architecture as a more convenient alternative for parameter-efficient transfer learning. However, when the number of parameters increases, the results drop compared to conventional adapters. We hypothesize that this is due to the more complex and subject to variability sampling of z during Vadapters training.

4.3.6 Discussion

In this work, we demonstrate the usefulness of adapter transfer in the context of children’s speech. With less than 10% of the total number of fine-tuning parameters, adapters are able to efficiently model children’s speech. Noticeably, the adapter performance approaches fine-tuning, as the number of pa-

rameters increases. Furthermore, our Vadapter architecture outperforms conventional adapters in terms of acoustic variability robustness in a parameter-efficient setting. Using a combination of Vadders in the encoder and conventional adapters in the decoder allows for further improvement, getting closer to the fine-tuning performance while keeping a small number of parameters. This seems to demonstrate their effectiveness in modelling highly variable data, such as children’s speech.

4.4 Summary

We covered the state-of-the-art for end-to-end children’s speech recognition in this chapter. Particularly, the usage of the Transformer architecture in conjunction with transfer learning. In a similar way as chapter 3, to avoid a drop in performances attributable to an acoustic mismatch between children and adults, the end-to-end model should be trained with children’s data. In contrast to previous work, we demonstrated that fine-tuning only a portion of the transformer modules, particularly the FFN sub-module, yields better results since it serves as a key-value memory. As a result, we placed our adapter subsequent to it. The adapter’s role is to accomplish knowledge transfer, which is related to transfer learning. Rather than updating the complete model’s weights, we just tweak an extra module, hence fewer parameters. This adapter transfer achieves almost identical results as the entire model fine-tuning. In addition, adapters are useful in the context of customized models, where training and storing a whole model for each age group or each child can be expensive and time-consuming.

In addition, we proposed the variational adapter, a variant of the traditional adapter based on variational auto-encoders. Compared to the adapter, which takes a bottleneck encoder-decoder structure with a linear layer as encoder and a linear layer as decoder, the variational adapter’s encoder consists of two branches, μ and σ . The outputs of these two branches are used as the mean and variance vector to sample the input of the decoder. By doing so, we enforce the adapter’s input variability to be contained in the σ branch. A branch which is suppressed during inference. As a result, we reduce input variability while maintaining the same size as standard adapters.

5

Work Plan

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This chapter is structured as follows: First, we summarise the contribution of this proposal in section 5.1. Then, section 5.2 provides an overview of the current subjects we intend to address in the future of this thesis, as well as a brief review of related work. Finally, Section 5.3 outlines tasks and milestones for the work proposed in the previous section, as well as a timeline for completion.

5.1 Proposal summary

In this proposal, we have introduced the difficulties inherent in children’s automatic speech recognition, namely inter- and intra- speaker variability, as well as the lack of data. We defined the purpose of this thesis: Improve children’s automatic speech recognition as an oracle for pathological speech applications for children. It is in that direction, that the work described in chapters 3 and 4 proposed to improve children’s ASR in different settings.

In chapter 2, we reviewed the different children’s ASR challenges, including acoustic variability due to the developmental changes in the child’s speech production apparatus, the limited linguistic knowledge and the lack of available data. Later, we presented the different approaches to tackle the aforementioned challenges, distributed in different aspects of the speech recognition pipeline: robust feature extraction and adaptation, data augmentation, detail of the annotation, the structure of the acoustic model and finally, the use of new training procedures. We identified knowledge transfer, with transfer and multi-task learning, as the most promising technique to reduce the performance gap between adult and children ASR.

The experiments on hybrid speech recognition models, especially HMM-DNN, are presented in Chapter 3. We started by investigating how we can exploit adult data to improve children’s speech recognition, even in a low-resource setting. The objective was to answer to the research question: *Which knowledge transfer approach is best for efficiently modelling and improving automatic recognition of children’s speech?* To this end, we first created two separate acoustic models, each trained solely on either adult or child data. We observed that adult speech recognition outperformed that of children. Additionally, when we used the opposite model to decode child and adult speech, we found a significant drop in recognition score, demonstrating the acoustic mismatch between adult and child speech. We then used transfer learning and multi-task learning to train two additional acoustic models. The goal was to see if using an inductive bias for the model’s weights would be beneficial in a low-resource task. In this experiment, we discovered that transfer learning produced the highest speech recognition score for children. The fitted model, however, no longer works on adults due to the catastrophic forgetting problem of transfer learning. It is in this case that multi-tasking learning comes in handy. Indeed, combined parallel learning of adult and child data allows the adult- and child-specific parts of the model to make effective use of the intrinsic speech characteristics captured by the model’s shared component.

However, because of parallel learning, children’s performances are slightly lower, since they compete with adult recognition in the training.

With this in mind, we explored whether we could improve children’s ASR using only data from children. Indeed, in most previous work in the literature, most of the work has been done in English, a language well endowed with resources. But this is not the case for most languages, where even data on adults may be lacking. In this experiment, we decided to use small children’s speech datasets from different languages, none of which exceeded 30 hours. We then evaluated the performances of the different knowledge transfer methods mentioned above. Like the previous experiment, we observed that multi-task learning did not improve the model trained only on each corpus independently. The reason is that all small corpora competed during multi-task learning, which leads to changes in the learned pattern in the shared component, making the task of language-specific layers more challenging. In contrast, transfer learning from any pre-trained child language consistently increased recognition score. It confirms that transfer learning is still relevant, even when the source task is low-resource, because the model eventually observed more children data, hence more variability. This experiment answered our research question: *Can these knowledge transfer approaches be used to efficiently exploit low-resource children’s speech data from multiple languages?*

Finally, motivated by the consistency of the different observations, we proposed our multilingual transfer learning strategy (MTL). MTL is a two step procedure that combines multi-task learning and transfer learning. During the first step, a model is trained using multi-task learning. It enables the shared component of the model to be supplied with as much kid data as possible. Then, in a second stage, we fine-tuned the multi-task model for a single target language using transfer learning. We demonstrated experimentally that this strategy still exceeds the best transfer learning of a model trained on a single task - language - alone. Furthermore, when we removed the target language from the multi-task learning stage, we discovered that our technique used the pre-trained shared component of the model efficiently.

In Chapter 4, we shifted our research to the end-to-end paradigm because of its recent success in the field of child speech recognition [87,88]. We presented the Transformer, the most state-of-the-art end-to-end speech recognition architecture. Transformer is a sequence-to-sequence design that relies only on the self-attention mechanism. Our initial approach was to investigate transfer learning for children using a pre-trained adult model. However, in prior literature, the weights of the entire model were modified during the children transfer learning. Therefore, the origin of the improvement was not clear. Hence our research question: *Why do end-to-end automatic speech recognition models achieve state-of-the-art results for children’s ASR? Particularly, what are the components that are most important to fine-tune?* As a result, we decided to study transfer learning for children using Transformer by modifying only a portion of each Transformer layer. The feed-forward network was discovered to be the most significant module on which to perform transfer learning (which consists of an up-projection followed by a down-

projection).

As highlighted in previous work, age- or speaker-dependent systems perform best for automatic recognition of children’s speech, as speaker variability is reduced within the same group. However, an end-to-end model can be expensive both in terms of training and storage. As a result, applying transfer learning to the entire model may not be the best solution. Especially with the catastrophic forgetting problem of transfer learning requires training and storage for each group. To solve these problems, we propose to use adapters for children [7, 162]. An adapter is a new set of weights inserted into the layers of a transformation model. Adapters offer an alternative to full model fine-tuning for each downstream task while preserving performance. Adapter transfer consists of training only the adapter weights while keeping the original model fixed. As a result, all previously discussed issues such as storage, training time and catastrophic forgetting are eliminated. Knowing the role of the position-wise feed-forward network during fine-tuning we decided, for each transformer layer, to place an adapter immediately after them. We demonstrated that adapter transfer produces marginally worse results than complete fine-tuning in the context of children’s ASR, but it required less than 20% of the original amount of parameters. This achievement inspired us to develop our own adapter architecture, the variational adapter. Unlike a typical adapter, which consists of a linear down-projection followed by a linear up-projection, we proposed using a variational auto-encoder architecture instead of the linear down-projection. It consists of two linear down-projection, giving a mean and variance vector respectively and a sampling approach based on a Gaussian distribution parameterized with this mean and variance. When utilized in the Transformer model encoder, the variational adapter outperformed the regular adapter. This confirms our hypothesis that Transformer encoder represents more audio information, but the decoder is more linguistic ones. These experiences on Adapters allow us to start answering positively to our research question: *Is it possible to develop an age-based, parameter-efficient automatic speech recognition model?*

5.2 Ongoing and future work

In order to answer the following research questions: *Is it possible to develop an age-based, parameter-efficient automatic speech recognition model?*; *Is it possible to use children’s synthetic speech to extend the amount of children’s data?* *How can we control the quality and speakers’s variability?*; - *Given that self-supervised representation based ASR for adults matches or surpasses current state-of-the-art, are these representation appropriate for children’s speech?* , we want to pursue three research directions in the future work of this thesis: As a first direction, we want to keep exploring adapter transfer. For example, as proposed by Pfeiffer [173], employing multiple adapters trained on different age groups or children corpus and combining them with an attention mechanism. It may also be interesting to investigate the use of explicit speaker information for robust adapter transfer [174]. Furthermore, as explained

in section 4.3.2, Adapter module is structurally similar to an auto-encoder, thus it would be interesting to modify the structure of the adapter to follow the latest development in auto-encoder research. For instance, using neural discrete representations with the help of vector-quantized codebook [175] at the end of the adapter’s down-projection.

As a second direction we want to investigate on the use of Text-to-speech (TTS) data augmentation. Indeed, as mentioned in section 2.1.3, one of the most significant obstacles in children automatic speech recognition is the lack of training data. One solution to this problem is voice conversion, in which the adult speech is transformed to child speech and then used for augmentation. However, the modified adult data can only capture a subset of the aspects of children’s speech. As a result, it is important to generate children voice data directly from text [99]. A multi-speaker TTS system using speaker embeddings and text as input can be used to produce a synthesized utterance with the variability of child speaker. Indeed, the speaker embedding includes acoustic variability informations, which we want to find in the output utterance [97, 176]. One of the most challenging aspects of this approach is that the TTS model for children produces unequal quality speech due mainly to acoustic variability, and hence the ASR system suffers when trained with this additional synthetic data. Furthermore, because we intend to augment the original data with TTS data, there may be a domain shift that worsens the ASR performance. To address both of these concerns, a speaker embedding-based data selection method has been suggested, based on the computation of the cosine-similarity between the input speaker embedding and the speaker embedding obtained from the output utterance. [99]. More recently, in order to minimise domain shift, two separate normalisation layers have been employed, one for the original data and one for the TTS data. [177]. In our research, we want to combine these two approaches to maximise the contribution of TTS data during training. In addition, we want to investigate whether the use of a GAN [178], which could create an artificial children embedding, could alleviate the problem of the limited number of speakers during training.

The final research direction we want to explore in this thesis is self-supervised learning (SSL) as a front-end feature rather than typical filter banks or MFCCs. For these models, the training process is separated into two stages. The first phase of training is self-supervised, which implies that no labels are used during training. The objective of this first phase is to present a large amount of unlabelled data to the system so that it learns a good speech representation. The second stage of learning is supervised fine-tuning, in which the model is taught to predict specific phonemes using the robust representation acquired in the previous stage with the help of a small amount of labelled data. In this category, two models stand out as state-of-the-art: Wav2Vec 2.0 [179] and HuBert [180]. As a preliminary experiment, to assess the usability of such frameworks for children ASR, we trained a BiLSTM model using the output of a variety of frozen self-supervised systems. For this experiment we used a subset of 50h of the Myst corpus [22], and the preliminary findings are displayed in the table 5.1

Front-end	UER	WER
Fbanks	12.29%	35.14%
TERA [181]	11.31%	31.80%
Audio Albert [182]	12.28%	34.69%
Wav2Vec2.0 Base	7.37%	19.76%
Wav2Vec2.0 Large	7.00%	18.76%
Distill HuBert [183]	9.22%	25.75%
HuBert Base	7.40%	19.77%
HuBert Large	6.03%	15.41%

Table 5.1: Results without language model of Self-supervised front-end

Where Base, and Large represent the same model with different number of parameters (in the order Base < Large). Even though we did not use a language model in this pilot experiment, the results are of the same order as those reported in section 4.3.4 obtained with a transformer and a transformer language model. Such results demonstrate that SSL learns substantial speech characteristics. For future research, we aim to explore in depth what information is encoded in SSL models and why they work well on children, and how we may use this knowledge to enhance children’s ASR.

5.3 Proposed timeline

A chronogram with the proposed time-span of each of the research direction can be found in Figure 5.1



Figure 5.1: Proposed timeline - November 2022 - September 2023.

6

Pathology detection from speech

As mentioned in the previous section, SLT can assist paediatric speech therapists by automatically assessing pronunciation quality and identifying pathological conditions. Although the primary aim of this thesis was to improve ASR for reliable assessment of pronunciation quality. We also contributed to the identification of pathological conditions from speech, which will be discussed in this section.

The potential of speech as a non-invasive biomarker for evaluating a speaker's health for both physical and psychological disorders has repeatedly been proven by the results of several works [184, 185]. Traditional speech-based disease classification systems have focused on carefully researched, knowledge-based features. However, these features do not always capture the full disease's symptomatology and may even ignore some of its more subtle signs. This has led research to move towards generic representations that intrinsically model the symptoms. However, there are not enough pathological speech data available to train a large model directly. In our work [186], we proposed to assess speaker embedding, such as *i-vectors* [80] and *x-vectors* [187], applicability as a generic feature extraction method to the detection of Parkinson's disease (PD) and Obstructive Sleep Apnea (OSA). All disease classifications were performed with a support-vector-machine (SVM) classifier. Our experiments with European Portuguese datasets support the hypothesis that discriminative speaker embeddings contain information relevant to disease detection. In particular, we found evidence that these embeddings contain information that hand-crafted features fail to represent, thus proving the validity of our approach. It was also observed that x-vectors are more suitable than i-vectors for tasks whose domain does not match the training data, such as verbal task mismatch and cross-lingual. This indicates that x-vectors embeddings are a strong contender in the replacement of knowledge-based feature sets for PD and OSA detection.

Later, in [188], we proposed to extend the aforementioned work by classifying Alzheimer's disease with the conjunction of both acoustic and textual feature embeddings. In this end, speech signals are encoded into *x-vector* using pre-trained models. For textual input, contextual embedding vectors are first extracted using an English Bert model [47] and then used to feed a bidirectional recurrent neural network with attention. This multi-model system, based on the combination of linguistic and acoustic information, attained a classification accuracy of 81.25%. Results have shown the importance of linguistic features in the classification of Alzheimer's disease, which outperforms the acoustic ones in terms of accuracy.

Finally, we further extend the idea of using pre-trained representation to automatically detect COVID-19 from cough recordings. We leverage transfer learning to develop a set of COVID-19 classification subsystems based on deep cough representation extractors called experts. Individual decisions of three experts are fed to a calibrated decision-level fusion system. This ensemble of expert subsystems based on cough representations is expected to produce well-calibrated log-likelihood scores over a wide range of operating points. The output can be more easily interpreted by a human expert and incorporated into the decision-making process. Our results show competitive performance compared to hand-crafted

features, although they are still far from those required to become a reliable tool to assist COVID-19 screening.

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