HELSINKI UNIVERSITY OF TECHNOLOGY
Department of Electrical and Communications Engineering
Laboratory of Acoustics and Audio Signal Processing

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# A Review of Articulatory Speech Synthesis

Master's Thesis submitted in partial fulfillment of the requirements for the degree of Master of Science in Technology.

Espoo, June 5, 2006

Supervisor: Professor Unto K. Laine
Instructor: Professor Martti Vainio, University of Helsinki

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ABSTRACT OF THE MASTER'S THESIS


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Articulatory speech synthesis models the natural speech production process. As a speech synthesis method it is not among the best, when the quality of produced speech sounds is the main criterion. However, for studying speech production it is the most suitable method.

This thesis reviews the literature on articulatory speech synthesis. The objective is to look at articulatory speech synthesis form the viewpoint of basic speech research. From this viewpoint, articulatory speech synthesis is at its most interesting, when fidelity to natural speech production process is considered a model's most important aspect.

The thesis reviews methods for articulatory measurements, geometrical synthesis of articulation and synthesis of speech sounds. The work divides the measurement methods to static and dynamic methods according to whether they are capable of capturing movement or not. In surveying geometrical synthesis the text concentrates especially on movement parametrisation and trajectory generation. Central topics in speech sound synthesis are the transmission line method and a glimpse at methods for non-linear sound generation.



Keywords: Articulatory speech synthesis, articulation, coarticulation, articulators, articulatory modeling, articulatory measurements, audiovisual speech, speech production, speech production model

TEKNILLINEN KORKEAKOULU

DIPLOMITYÖN TIIVISTELMÄ

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Artikulatorinen puhesynteesi mallintaa luonnollista puheentuottoprosessia. Puhesynteesimenetelmien joukossa se ei ole parhaasta päästä, jos tuotetun puheäänen laatu on tärkein arvostelukriteeri. Sen sijaan puheentuoton tutkimiseen se sopii synteesimenetelmistä parhaiten.

Tämä diplomityö luo katsauksen artikulatorisen puhesynteesin kirjallisuuteen. Työn tarkoitus on tarkastella artikulatorista puhesynteesiä puheen perustutkimuksen näkökulmasta. Tällöin artikulatorinen puhesynteesi on kiinnostavimmillaan, kun mallin uskollisuus luonnolliselle puheentuottoprosessille on sen tärkein ominaisuus.

Teksti tarkastelee artikulaation mittaamisen, artikulaation geometrisen synteesin ja artikulatorisen äänisynteesin menetelmiä. Mittausmenetelmät työ jakaa staattisiin ja dynaamisiin sen mukaan, pystyvätkö menetelmät tallentamaan liikettä. Geometrisen synteesin osalta tutkimus keskittyy erityisesti liikkeen parametrointiin ja liikeratojen tuottamiseen. Keskeisiä aiheita äänisynteesissä ovat siirtolinjamalli ja lyhyt katsaus epälinaariseen äänentuottoon.



Avainsanat: Artikulatorinen puhesynteesi, artikulaatio, koartikulaatio, artikulaattorit, artikulaation mallintaminen, artikulaation mittaaminen, audiovisuaalinen puhe, puheentuotto, puheentuottomalli

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Otaniemi, June 5, 2006 Pertti Palo

To Kaisa
for love and patience

# Contents

# Symbols and Abbreviations

$A(x)$ Area function i.e. cross sectional area of the VT in relation to distance from glottis

$\rho$ density of the medium

$\rho_0$ constant component of the density of the medium

$p$ perturbation pressure (sound pressure)

$q'$ mass source

$U$ volume velocity

$\overline{u}$ particle velocity

$[\mathrm{a}]$ Phone $\mathrm{a}$

**1D** One Dimensional

**2D** Two Dimensional

**3D** Three Dimensional

**4D** Four Dimensional (usually three spatial dimensions and time)

**5D** Five Dimensional

**A/D** Analog to Digital converter

**AFA** Arbitrary Factor Analysis

**C** Consonant (e.g. CV sequence = consonant-vowel sequence)

**CPU** Central Processing Unit

**CT** Computer Tomography

**D/A** Digital to Analog converter

**DMM** Dynamic Mechanical Model (of VT and glottis)

**EBCT** Electron Beam Computed Tomography

**EMA** Electromagnetic Articulography

**EMG** Electromyography

**EPG** Electropalatography

**F0** Fundamental Frequency

**F1, F2, F3, ...** Formant Frequencies

**FEM** Finite Element Model

**fMRI** functional Magnetic Resonance Imaging

**FOV** Field Of View

**fps** frames per second

**HTML** Hypertext Markup Language

**ICA** Independent Component Analysis

**IPA** International Phonetic Association

**LCA** Linear Component Analysis

**LPC** Linear Predictive Coding

**MR** Magnetic Resonance (as in MR image)

**MRI** Magnetic Resonance Imaging

**NN** Neural Network(s)

**OPG** Optopalatography

**PARAFAC** PARAllel FACtorial analysis

**PCA** Principal Component Analysis

**SNR** Signal to Noice Ratio

**TSE** Turbo Spin Echo (an MR imaging sequence)

**TTS** Text-to-Speech

**V** Vowel (e.g. VCV sequence = vowel-consonant-vowel sequence)

**VF** Vocal Folds

**VT** Vocal Tract

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# Chapter 1

# Introduction

To understand articulatory speech synthesis we have to understand human speech production. This is obvious, if you consider that we are trying to imitate speech production and it is very hard to imitate anything without understanding the object of imitation.

It is also good to know a little bit about speech synthesis in general. After all, articulatory speech synthesis shares many of its methods with other forms of speech synthesis. For these reasons I have written the following two sections as an introduction to someone who has limited experience of either subject.

## 1.1 Human Speech Production

Figure 1.1 shows the human speech production organs along with an idealized model. The picture on the left shows the main articulators - the tongue, the jaw and the lips - as well as other important parts of the vocal tract (VT). The picture on the right shows the model, which is the basis of almost every acoustical model of the vocal tract.

In production of pulmonic sounds breathing muscles act as an energy source while the lungs provide a storage of pressurised air[1]. The lungs are separated from the vocal tract by the vocal folds, which are also known as

[1] In production of non-pulmonic sounds such as clicks the energy source is provided by the tongue muscles as they press an air pocket against other VT structures and then release the pressure producing the click sound.

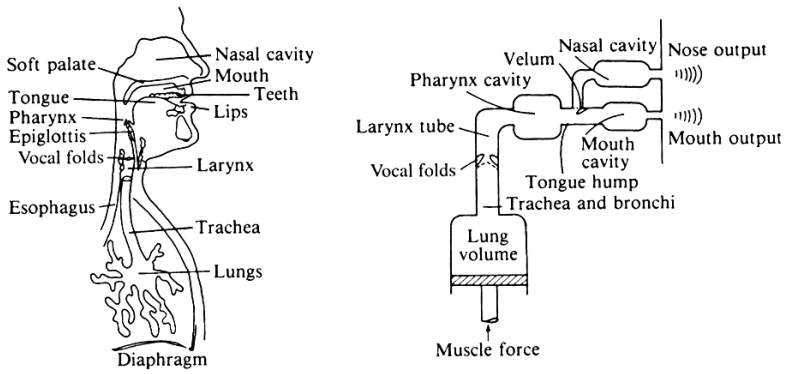



Figure 1.1: The human speech production organs and an idealized model (Rossing et al., 2002, p. 337)

vocal chords. The vocal folds generate a signal, which is then filtered by the vocal tract and finally radiated to the surroundings via the mouth and/or nostrils.

While the above description of the speech production process is fairly accurate for (some stages of) vowel production, it is only a starting point to understanding speech production in general. The neurological process, which leads to articulation movements, is not a part of the described process. It is also outside the scope of this work - and most others - but hopefully in the future a part of a "complete model of speech production".

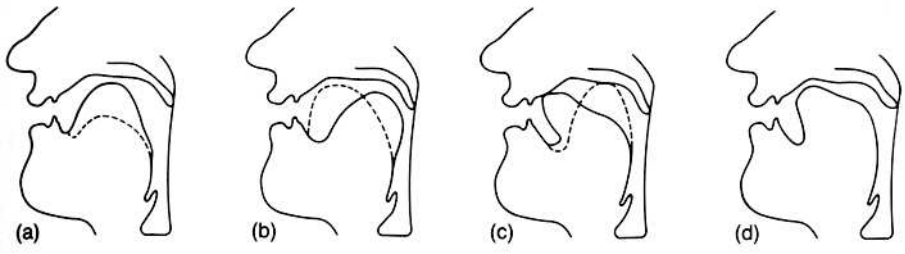



Figure 1.2: Typical articulation positions (O'Shaughnessy, 1987). (a) and (b) illustrate different kinds of vowels, (c) shows possible stop consonant articulations and (d) an alveolar fricative.

However, there are several things that can be added the above description that do fall in the scope of this review. For instance, the possible articulation positions and the physiology of the articulators themselves need to

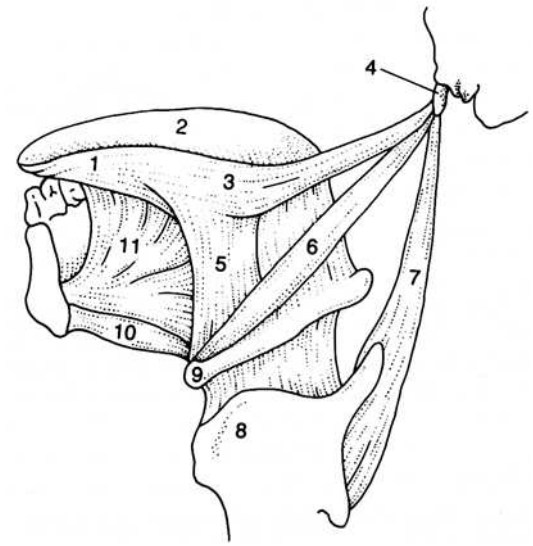



Figure 1.3: Diagram of tongue muscles and some related structures (O'Shaughnessy, 1987). The muscles and structures are: 1: inferior longitudinal muscle, 2: dorsum, 3: styloglossus muscle, 4: styloid process of temporal bone, 5: hyoglossus muscle, 6: stylohyoid muscle, 7: stylopharyngeal muscle, 8: thyroid cartilage, 9: hyoid bone, 10: geniohyoid muscle and 11: genioglossus muscle.

be considered. Figure 1.2 illustrates some typical articulation positions and figure 1.3 gives a glimpse at the complexity of the involved physiology.

Furthermore, as mentioned above, the description is for vowels only. According to Olive et al. (1993) consonants cause all sorts of havoc in such a simple model. Firstly plosives (e.g. [p, t, k]) are based on closing the VT at some point above the vocal folds. Secondly fricatives (e.g. [f, s]) add a turbulent noise source to the system at their point of articulation. Thirdly nasals (e.g. [n, m]) keep the mouth closed while the sound radiates through the nostrils. Fourthly sounds, which are called liquids in American English (e.g. [l, r]), divide the mouth cavity into two tubes instead of one. One could continue the list with the [r] of Finnish and many other sounds, which are more or - usually - less like vowels.

Then one needs to also note that not all sounds are voiced i.e. the vocal folds vibrate only during the production of vowels and voiced consonants. Indeed there are several different modes of operation for the vocal folds. Figure 1.4 illustrates the difference between three of these.


Figure 1.4: The vocal folds. The three shown operation modes are a) breathing (folds stay open) b) normal speech phonation and c) whisper phonation. (from Kahle et al. (1984) in Mixdorff (2002))

Finally, there is still one thing of utmost importance which makes the above description fall somewhat short: Articulation is not a static thing and speech sounds are not produced by isolated incidents of ideal articulation. Rather the articulation of a phone is a function of time and a process known as coarticulation. That is the effect that surrounding phones have on the currently articulated one. For example, [t] in the utterance [ata] is articulated differently from the [t] in the sequence [utu].

## 1.2 Speech Synthesis

A speech synthesis system is by definition a system, which produces synthetic speech. It is implicitly clear, that this involves some sort of input. What is not clear is the type of this input. If the input is plain text, which does not contain additional phonetic and/or phonological information the system may be called a text-to-speech (TTS) system. However, in the strictest sense the term "speech synthesis system" applies only to the part of a TTS system, which produces the speech signal from some sort of a phonetic description. While this strict definition will be used in this thesis, it is useful to take a look at TTS as it is the most common context for speech synthesis.

### 1.2.1 Text-to-Speech Systems

A schematic of the text-to-speech process is shown in figure 1.5.

As seen in the picture the synthesis starts from text input. Nowadays this may be plain text or marked-up text e.g. HTML or something similar. If the text uses some sort of mark-up it may already contain some or all

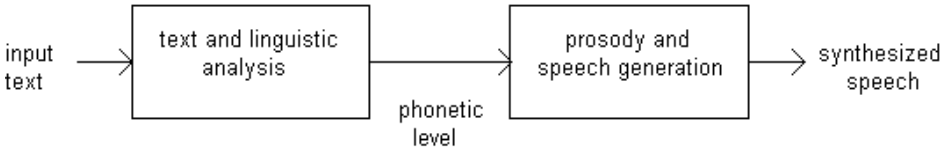



Figure 1.5: Schematic of text-to-speech synthesis citeLemmetty:RSST:1999

of the information made available by the text and linguistic analysis stage. Regardless of the quality of the input text, after this stage we will have a description of the text on the phonetic level. A fairly popular way of representation is again a mark-up of some sort.

The second stage is prosody and third stage speech signal generation. During the prosody stage linguistic information is used to generate F0 contours, timing information for the phones etc. Finally, the synthesized speech itself is generated from these specifications. If we are dealing with normal TTS, the generated speech will take the form of a audio signal, but on the other hand, if we are dealing with text-to-audio-visual-speech then the generated audio will be accompanied by a synchronized video sequence.

### 1.2.2 Speech Synthesis Methods

The above description holds for most TTS systems. The most important differences are usually in the way that the audio (and video) signal is generated. In a way the easiest method to understand is concatenative SS. In essence it is cut-and-paste synthesis. The audio (or video) signal is formed by concatenating prerecorded speech samples. The hard parts are in selecting a good database and selecting the best samples from the database. Usually the database consists of phonemic units or diphones i.e. transitions from one phone to another, but it may also contain longer section such as complete words or shorter ones.

Another method, which is more closely related to articulatory speech synthesis, is formant synthesis. Its goal is to produce an speech by generating with rules a signal, which mimics the formant structure and other spectral properties of natural speech as closely as possible.

### 1.2.3 History of Speech Synthesis

There are several good texts on the history of speech synthesis - Klatt (1987) and applicable parts of Flanagan (1965) to mention just two. Thus here we will only take a brief look at some interesting landmarks.

The history of speech synthesis machines begins at least as early as the 17th century (Fagyal, 2001). Unfortunately, these early attempts did not leave anything but indirect documentary evidence of their existence. If they even ever did exist as anything but plans, they were considered some kind of musical instruments by their designers and users. Nevertheless, the early machines were an omen of things to come. Until the late 19th century SS was based on construction of physical models, which can be considered simple articulatory speech synthesizers.

The beginning of text-to-speech systems can be placed in late 18th century. In 1779, the Academy of Sciences of St Petersburg held a contest for building and demonstrating a speaking machine capable of producing five vowels on its own. The prize went to the well known tube machine of C. G. Kratzenstein. Kratzenstein's machine produced only static vowels, but another contestant - Wolfgang von Kempelen - demonstrated a machine capable of producing dynamic speech. Figure 1.6 shows Wheatstone's reconstruction of von Kempelen's machine[2].

The late 19th century gave birth to all sorts of electrical devices and among them was a electromechanical speech synthesiser by Helmholtz. Another one of the first electrical analogs of the human speech organs was presented by Stewart (1922), who appears not to have known of Helmholtz's earlier vowel synthesis machine. Stewart's device consisted of an interrupted source and two adjustable RCL-resonator circuits. These can be seen to correspond to the vocal folds (the source and interrupter) and the VT (the resonators).

The next great landmark in the history of speech synthesis systems is the boom in 1950's. But this brings us already to the scope of this thesis. Two of the systems from this era are described in section 4.2, page 98.

[2]This reconstruction had an interesting role in the history of speech technology (Deller et al., 2000). Alexander Graham Bell had the chance of seeing Wheatstone's reconstruction as a boy in Edinburgh, Scotland. This event set him on the track, which lead to the U.S. Patent 174465 (the voice telephone).

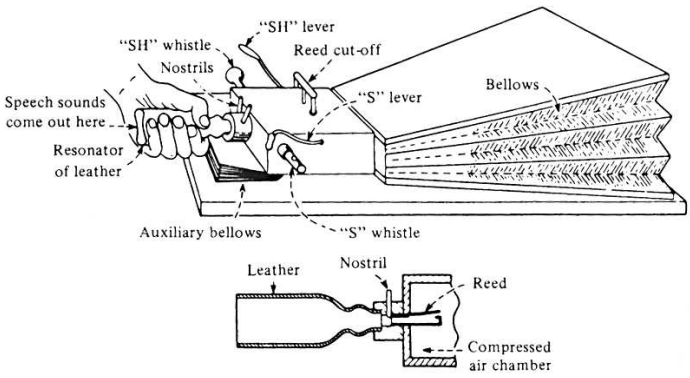



Figure 1.6: Wheatstone's reconstruction of von Kempelen's speech synthesizer (Flanagan, 1965)

## 1.3 Articulatory Speech Synthesis

### 1.3.1 Definition

While this is a master's thesis in engineering rather than in philosophy I still find it useful to give my definition of the topic. Although I have tried to make this definition as precise as possible I still might include material that does not strictly fall under it.

The definition of articulatory synthesis that I decided to adopt for this work is as follows:

"Articulatory speech synthesis models the natural speech production *process* as accurately as possible. This is accomplished by creating a synthetic model of human physiology and making it speak."

This definition gives a set of goals for articulatory speech synthesis:

1. Accuracy of the model in comparison with the speaker(s) on, which it is based.
2. Naturality of the model.
3. Intelligibility of the model.
4. New information and understanding gained from the model.

### 1.3.2 Motives and Applications

Articulatory speech synthesis may be utilised in a project aiming at a finished application e.g. a commercial speech synthesiser. Thus, motives and applications for a speech synthesiser are in principle valid motives and applications for research into articulatory speech synthesis. Nevertheless, it is worth noticing, that if the motivation for a project is for example better quality of synthesised speech within a time frame of few years, articulatory speech synthesis probably is not the answer. Nowadays there are methods like LPC and concatenative synthesis, which are more successful with the quality of output than articulatory speech synthesis. This work, however, concentrates on articulatory synthesis in the context of basic research. And this is the area of application and the motivation where articulatory speech synthesis is very good.

Articulatory speech synthesis may be used as a tool in basic speech research and is in itself a subject of basic speech synthesis research. Articulatory modeling is a good way to gain insights in to the speech production and perception processes. If we are able to understand speech production at the deepest levels, we will be able to construct an articulatory speech synthesiser, which is indistinguishable from a natural speaker. The reverse can be stated with equal certainty: A good enough articulatory speech synthesiser will enable us to understand the speech production process in its smallest details.

On the other hand, it is important to know which properties of the articulatory process have to be modeled in order to achieve high quality speech synthesis. Detailed articulatory speech synthesis would make answering this problem, if not easy, at least fairly straight forward. Of course there would still be a considerable amount of work involved. The necessary perceptual studies would, in contrast, be much simpler with the degree of control that a synthesiser would make possible when compared with a natural speaker.

When considering the naturalness of a synthesiser one important question to consider is whether geometrical modeling is necessary. The fact is that there are some - especially prosodic - features of speech, which are nearly impossible to reproduce with articulatory synthesis without geometrical modeling. Some of them are very hard to extract from the acoustic

signal, but still clearly perceptible to a human listener. Moreover, it can be argued that it is simpler to remove features, that are found to be unnecessary, than it is to add - possibly unknown and unrecognised - features to a model.

Another important aspect to consider is, that articulatory models are not used only for speech production. They are also used for speech recognition (see e.g. (Blackburn, 1996)) as a part of the effort of trying to understand the twin process of speech production and recognition by modeling it as a whole. In this particular context articulatory speech synthesis may be seen as a model for the information encoding in the acoustic speech signal. Figure 1.7 shows a rough sketch of the parts of such a model along with some of the types of data needed for the construction of each part of the model.

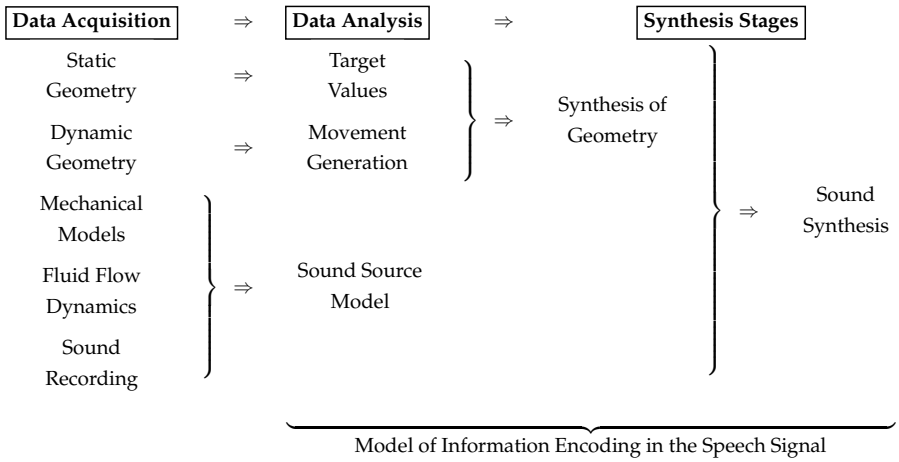



Figure 1.7: A schematic of the construction of an articulatory speech synthesiser and how a such a synthesiser may be considered to contain a model of information encoding in the speech signal.

If we take a wider view of speech research, then two additional reasons for including geometrical synthesis in the model are present. An articulatory speech model could also be used to study speech pathology in the same way as perception models can be used to study perception pathology (see e.g. Eysenck and Keane (2000)). This would obviously benefit from the possibility to view the problematic areas. Moreover, if the geometrical model is a part of the synthesiser, it is simple to generate also video output and enable the model to be used in multimodal research.

Besides the more immediate gains in basic research, it is good to also keep in mind some of the possible uses of an articulatory speech synthesiser as a finished product. These include use as a virtual language tutor (Engwall, Wik, Beskow, & Granström, 2004), in speech therapy, in general purpose audio-visual speech synthesis, in speech encoding (Schroeter, Larar, & Sondhi, 1987), (Schroeter, Larar, & Sondhi, 1988), (Parthasarathy, Schroeter, Coker, & Sondhi, 1989), in imitation of real speakers, as the speech engine of virtual actors and even as a toy.

To summarize, the most important contribution of articulatory models is the same as that of any model in general. They are ways of unifying knowledge and as such act as test beds of theories. Without modeling and comparing the results of the model with the original it is hard to say whether a theory is useful or not.

## 1.4 Purpose and Structure of This Thesis

As the title of this thesis suggests, I will not review the use of articulatory modeling in speech or speaker recognition. Neither will I spend very much time describing synthesis methods, which do not aim at modeling speech with methods based on modeling articulation. A final area of research, which I excluded from the scope of this work, is modeling the vocal folds and/or the glottal signal source. This exclusion was done purely to limit the amount of work, that was necessary in finishing this work. However, none of these exclusions are absolute. Therefore, I will discuss any - and hopefully most of the interesting - research that sheds light on the topic of this thesis.

The disposition of the following chapters follows roughly the modularisation described by figure 1.7. The applicable parts of the figure's first column are the topic of chapter 2, page 12, where I will describe several ways that are or have been used in different projects to obtain information for articulatory synthesis. The focus of this chapter will be the quality of data obtained by a given method and the ethical, technical or other kinds of problems arising from the use of the method.

In chapter 3, page 60 I will describe methods, which have been employed in constructing the geometrical part of articulatory synthesis systems. Different types of static geometries will be reviewed as well as movement parametrisations and movement generation methods. Finally a short look will be taken at handling collisions between articulators.

In chapter 4, page 93 I will review acoustic synthesis methods. The chapter will start with a look at the mathematics of vocal tract acoustics. After that sound synthesis methods will be reviewed. The main focus will be on methods, which have lead into more accurate descriptions of the speech production process. Thus, methods, which have focused on other issues, such as real time synthesis, will be left out.

Chapter 5, page 106 is the concluding chapter. There I will discuss the theories covered as well as some of the ones left out of this thesis. I will also present some directions of future research, which seem to be emerging form the field.

# Chapter 2

# Data Acquisition Methods

The first step in constructing a model is acquiring data on the phenomenon, which is about to be modeled. Accordingly, data acquisition should play an important role also in the construction of an articulatory speech synthesiser.

Important considerations in choosing the data acquisition method for such a project are the application area of a method (which part or parts of the articulatory system it can be used to explore), temporal and spatial accuracy, safety issues, and other issues affecting the usage of the method. In addition, the properties and nature of the raw data obtained in actual studies should be considered. If these considerations are done carefully the next steps of the construction process - building the geometric and acoustic synthesis models - will be made considerably easier.

## 2.1 Static methods

These are methods, which capture static snapshots of articulation. The defining characteristic and main problem of these methods is, that they are unable to represent movement. Instead only isolated samples of articulation may be obtained. Moreover, a fairly common problem is a requirement of prolonged articulation, which may lead to unnatural results. This may be caused by the subject getting tired, but also by their anticipation of prolonging articulation and other, method related, factors.

To minimise the unnaturalness caused by movement between repetitions and during measurement, subject fatigue, and subject's position various

measures can be taken. Very often the articulations can be verified indirectly by comparing the sound produced during data acquisition with sound produced under more natural conditions. During data processing movement artifacts can be removed by fixing the coordinate system to bones instead of subject's position within the measurement device. With some of the methods it is also possible to shorten the acquisition time by sacrificing resolution and thus reduce all of the effects caused by a long acquisition time.

Even though the static methods have some intrinsic problems, they are very important tools. This is due to the fact, that there are currently no real 4D methods. That is, there are no methods, which would gather spatially three dimensional data in real time. Because of this, static methods have to be used to obtain 3D snapshots of articulatory positions. These snapshots can then be used - for example - as the target positions in a coarticulatory model or in a similar manner.

### 2.1.1 Direct Dimensional Measurements

Direct dimensional measurements have been employed in a wide variety of studies to record articulatory data. However, the success of these methods has not always been as great as their ingenuity of design. The single most prominent problem is their interference with the articulation itself. On the other hand they have the advantage of being usually easy to calibrate.

Gauffin and Sundberg (1978) used a fiberscope to obtain pictures of the pharynx. Gauffin and Sundberg (1978) inserted the fiberscope tube through the subject's nose and velar opening. They subsequently took pictures at two levels in the pharynx during sustained phonation of isolated vowels. In order to obtain panoramic views of the pharynx and the vocal cords they swung the fiberscope tip from left to right and afterwards combined the obtained partial views. Two example pictures obtained by Gauffin and Sundberg (1978) can be seen if figure 2.1. An important finding by Gauffin and Sundberg (1978) was that the relative dimensions of the VT differ at higher and lower levels in the pharynx.

While the fiberscope method is quite safe and only causes some discomfort, it is not a very universal method. The methods area of application is restricted to the structures visible beneath the velar opening. Furthermore,

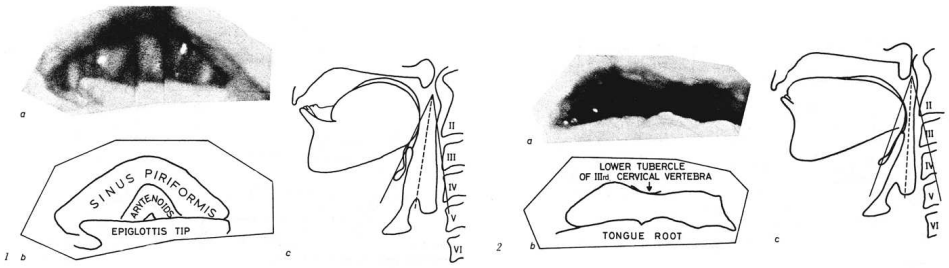



Figure 2.1: Two examples of the pictures along with diagrams of the visible anatomy and the points where the pictures were obtained (Gauffin & Sundberg, 1978).

it would probably be difficult to use the fiberscope method to observe consonant articulation as the articulators would be disturbed by the fiberscope tip and thus the articulation would not remain natural.

As for the nature of the data obtained by the fiberscope method, it is only qualitative, if a scale of reference is not obtained by some other method (Gauffin & Sundberg, 1978). Gauffin and Sundberg (1978) used X-ray tracings to obtain a scale of reference and suggested that the X-rays should be taken simultaneously with the use of the fiberscope to further determine the position of the fiberscope tip.

### Casts of the Vocal Tract

Casts of the VT have been widely employed in speech production research. It has been used to acquire data on the shape of the oral cavity and constriction caused by the teeth. With recent versions of the method, which combine MRI with the more mundane process of taking a dental cast, the method has become something of a standard tool. The taking of the cast itself is a standard procedure of dental medicine and has not usually been varied when using the technique for speech research[1].

Stark, Ericsdotter, Branderud, Sundberg, Lundberg, and Lander (1999) made dental casts of anterior VTs in order to be able to form good conversion rules from cross-dimensions to cross-section areas, ie. $d(x)$ to $A(x)$.

[1]There have, however, been a couple of studies where the technique was employed to produce a cast of the whole oral VT and hence had to be used differently from the standard medical procedure (Ladefoged, Anthony, & Riley, 1971)

They combined the information gained from the dental casts with their main data source of cineradiography to produce estimated sagittal cross-dimension to area functions for the whole VT.

Sundberg, Johansson, Wilbrand, and Ytterbergh (1987) used plaster casts of the subjects palates in a similar way to the use of dental casts by Stark et al. (1999). Instead of making a cast of the whole mouth Sundberg et al. (1987) made a plaster cast only of the palate. Afterwards the cast was filled with plastic. The plastic was then cut up into radial slices. The slices were determined by a circle which was tangent to the horizontal part of the hard palate and the vertical part of the posterior pharynx wall. These slices were used to determine the cross section area of the VT.

To test the effect of the slicing procedure on the results Stark et al. (1999) also made plastic slices with the the angle of cutting being orthogonal to the plane of the teeth. This, however, did not effect the results significantly.

What may be of greater importance, was that Stark et al. (1999) did not have a way of directly relating the amount of plastic to the position of the tongue. Rather they had to do it in a heuristic way using experience and knowledge of the tongue's position in the pharynx gained from their other data source, computed tomography.

Parallel to the use of casts to augment data from cineradiography (Stark et al., 1999) and regular tomography (Stark et al., 1999), they have been employed to augment MRI data. As MRI inherently lacks data on teeth the dental casts are vitally needed.

Baer, Gore, Gracco, and Nye (1991) took dental casts of their subjects. Dental impressions made by using the casts as molds were sliced and the slices digitised by manually tracing them with a graphics tablet. The resulting images were then used to correct corresponding slices obtained with MRI.

In a similar approach Narayanan, Alwan, and Haker (1995) made dental casts of their subjects and measured some relevant distances straight from the casts. In addition they also made dental impressions. The impressions were subsequently sliced and used paper tracings of coronal and sagittal slices to correct MRI results.

The latest innovation is using MRI to measure the casts and hence easing the task of fitting the teeth with the MR images. Engwall and Badin (1999)

gives a very detailed description of the required procedures. In short, the casts are immersed in a water tank, which is then imaged with the same scanner, which is used for imaging the VT. Afterwards, the resulting images are fitted together to form images of the VT with the teeth in place. The same procedure has also been used by Badin, Bailly, Raybaudi, and Segebarth (2002).

### 2.1.2 Computed Tomography (CT)

Computed tomography is a 3D imaging method which is based on X-ray imaging. An intermediate step in the historical development of computed tomography was ordinary tomography. It was a method of producing tomograms - pictures of a slice through the imaged object - on regular X-ray film. X-ray imaging and ordinary tomography have been used in articulatory studies and are discussed in section 2.2.1, page 31.

Like ordinary tomography, CT produces 2D slices of the subject. The slices are the result of computing the tissue density by solving the inversion problem for X-ray absorption data. The absorption data is gathered by rotating an X-ray source around the subject and registering the amount of radiation passing through the subject on sensors positioned opposite to the X-ray source. 3D images are produced by obtaining several parallel slices of the same subject.

While early instruments were naturally slower more modern ones have a fairly fast slice acquisition time. Even so, they have been far from being instantaneous in acquiring a 3D image. This is changing with the introduction of multislice-CT, which can at present acquire up to 16 slices in one pass.

CT produces images with a good contrast between air and bodily tissues in general. For the purposes of imaging the VT this is especially good as the walls of the VT will be clearly defined and easy to extract.

Moreover, CT has good resolution. Modern equipment are capable of producing a slice thickness of 1mm and plane pixel size of 0.2mm. These are, however, parameters for clinical use of CT. If used for non-medical purposes - such as speech production studies, it may be necessary to be more strict about the radiation doses and thus accept lower resolutions.

CT's main flaw is its use of X-rays. There will inevitably be an amount of ionising radiation absorbed by the subject. The obvious need to keep the radiation dose for the subject in acceptable limits restricts the number of images that can be acquired from one subject.

### Studies

Sundberg et al. (1987) took computed tomograms of two subjects (a male and a female) in an effort to establish the area function $A(x)$ relating the VT's sagittal cross-dimension to its cross-sectional area. The study was limited to tomograms at four layers of the pharynx during the sustained production of four vowels. The tomogram layers were 2mm thick and required a complete set of four required an acquisition time of 3.2 seconds. During the whole study the maximum absorbed dose of radiation for the subjects was 70 mGy.

As can bee seen in figure 2.2 the images were of good quality with a clear VT contour. Figure 2.3 shows the VT contours for the male subject.

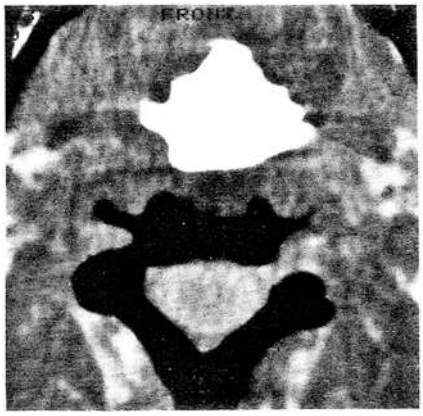



Figure 2.2: A CT cross section of a male pharynx 5.7 cm above the glottis while pronouncing [i:] (Sundberg et al., 1987). Air is represented by white and bone and tissues by other shades.

Interesting results from Sundberg et al. (1987) include the following: The VT's sagittal cross distance cannot be mapped to the lateral width by means of simple linear regression and, furthermore, the mapping varies between subjects. In contrast, such a mapping will produce good results, if it concerns only one subject and a specific level in the pharynx.

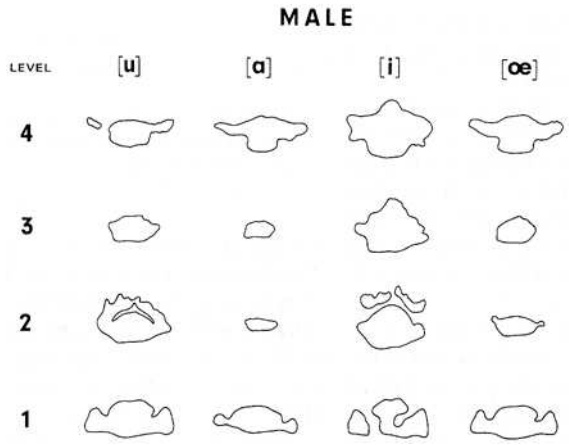



Figure 2.3: Cross-sectional contours of a male pharynx obtained by CT Sundberg et al. (1987). The cross-sections were all taken in the axial plane with levels ranging from the glottal region (level 1) to upper pharynx (level 4). All imaging levels were separated by 1.9 cm.

A more recent study by Tom, Titze, Hoffman, and Story (2001) (see also (Tom, Titze, Hoffman, & Story, 1999)) had as its objectives the acquisition of high resolution 3D images of the VT during phonation at variable pitch, loudness and register (chest or falsetto) conditions. After consideration they decided to use Electron Beam Computed Tomography (EBCT). The main reasons were EBCT's fast image acquisition time and good contrast between bodily tissues and air when compared with the then current MRI devices.

With the device used by Tom et al. (2001) the slice acquisition time was 100 ms, which, along with the slice thickness of 3 mm, made it possible to image the whole VT in 12 to 18 seconds. The 3D image thus acquired consisted of 60 slices with a resolution of 512x512 pixels each and a pixel size of 0.410 mm. Known error from a test with a phantom VT was 1.8-2.0 %.

Further processing of the acquired image set was performed with an image display and measurement tool called VIDA$^{TM}$. The VT region was identified in the images by a seeded region growing method. The 3D shape of the VT was then reconstructed with shape-based interpolation with the resulting image set having the same 0.410 mm resolution in all directions. The cross-sectional areas $A(x)$ were measured from the final set.

### 2.1.3 Ultrasound

Ultrasound is also called pulsed echo ultrasound. Pulsed echo ultrasound uses short bursts of ultrasound to study the acoustic impedance properties of a medium. The bursts are produced by a transducer (or several) and directed in a tight beam into the medium. As the wave encounters changes in the acoustic impedance of the medium or even a change in medium some of the wave's energy is reflected back towards the ultrasound device. This returning wave is then detected as an echo and the point of reflection is calculated by the time the wave took to travel to the point of reflection and back. If the wave is reflected by a boundary, where a dramatic change in acoustic impedance occurs, then most of the energy will be reflected back. This means that it will be practically impossible to gain any information from beyond that boundary.

Ultrasound has three technical handicaps as a tool for articulatory data acquisition. Firstly, it cannot record practically anything through an air gap. This means, that it is impossible to get a reading on the opposite side of the VT while recording the other side. Also, in some cases, when recording the tongue from beneath, it is not possible to get a reading on some parts of the tongue due to the tip being raised (for example [l] and [ɪ]). Because of this handicap ultrasound is used mostly to record the tongue's position.

The second handicap is that ultrasound cannot record oblique surfaces. That is surfaces with a steep slope in relation to the recording ultrasound beam. For example cases where the surface of the tongue is steeply curved can be problematic. This handicap may be overcome for the most part by careful positioning of the recording device.

The last handicap is ultrasound's inability to penetrate bone. This is related to the problem with air gaps: Any surface where the density of matter changes very radically tends to reflect most of the incident sound energy and thus makes it very hard to image anything behind it. Luckily, with the limitations posed by the previous two handicaps, this one is not a significant problem.

The greatest benefit of using ultrasound is that there are no known health risks given that the intensities used are in the diagnostic range. At the same

time, the data can be quite interesting and even relatively comprehensive, when modern methods are used.

### Studies

Minifie, Kelsey, and Zagzebski (1971) were among the first to use ultrasound to record articulation. They used it to study sustained vowels and fricative consonants. The device they used was essentially a 1D device. It required an acquisition time of 3 seconds for one sagittal slice and provided meaningful information only on tissue/air boundaries. The main result from this study was a confirmation of the usefulness of ultrasound as a research method.

Sonies, Shawker, Hall, Gerber, and Leighton (1981) developed a real-time ultrasound imaging system specifically for visualizing tongue movement during continuous speech and used it to record VV and VC sequences. The system incorporated a sector (2D) scanner with a sector sweep rate of 30 Hz. They did several simultaneous recordings with a sitting subject. Besides the ultrasound data timing information, a soundtrack, a low passed (100 Hz cut-off) sound signal form, and a frontal and lateral video view were recorded.

The ultrasound device's axial resolution (resolution along the sweeping sound beam) was 1.5 mm at a depth of 8 cm and its lateral resolution was 10 mm at a depth of 5 cm. During recording the subject's head position remained natural as they sat in a custom fitted wheelchair with the head position stabilised with a headrest and a velcro headband attaching the head to the headrest.

Sonies et al. (1981) validated the method with a simultaneous recording of cineradiography and ultrasound films. Two lead pellets were used to mark the tongue surface. Furthermore, a series of smaller pellets were attached to the ultrasound setup to indicate the direction of the ultrasound beam. Sonies et al. (1981) reported very good agreement of results with deviation between ultrasound and cineradiography data being explainable with intrinsic X-ray image distortion.

Keller and Ostry (1983) used an A-scan (1D) device with attached computer equipment to measure tongue dorsum movement to study CV and

CVCVCV sequences (only plosive consonants). The study was mostly aimed at developing and validating the technique. In a related article Keller (1987) reports use of the same equipment to study different kinds of speech impairments and it has also been used by Parush, Ostry, and Munhall (1983) to study coarticulation in VCV sequences.

In the setup described by Keller and Ostry (1983) the transducer was held in position under the chin with an adjustable head harness. The harness' effect on jaw movement was assessed with videotapes where the subject repeated the same material with either the harness on or off. The differences in articulation were found statistically insignificant.

Keller and Ostry (1983) further validated their results with comparison to results from previous X-ray studies and a one of their own. In comparison they found agreement of results, but with greater in-method variability in the lateral X-ray results.

Stone and Lundberg (1996) and Lundberg and Stone (1999) used 3D ultrasound to record tongue surfaces. In the first paper they report recording data on sustained vowels (in a $[\mathrm{pV(p)}]$ context) and consonants (in a $[\mathrm{aC}]$ context). The aim of the initial study was to find out whether the 3D surface shapes differed categorically between vowels and consonants.

Stone and Lundberg (1996) used an ultrasound device with a curvilinear array of 128 crystals mounted to scan a $90\,^\circ$ slice. One slice was acquired in a time of 3.3 ms. To get 3D images the device had a motor which pivoted the transducer array through an arc of $60\,^\circ$ with $1\,^\circ$ increments. The acquisition time for one 3D image of 60 slices was thus about 10 s.

The images, while of good quality, were limited by some of the usual problems with ultrasound. The tongue's tip and lateral margins were often behind an air gap and thus invisible to the device. Similarly the tongue root was at times obscured by the hyoid bone.

Based on the data they gathered Stone and Lundberg (1996) were able to categorise the studied material into four shape classes: front raising, complete groove, back raising and two-point displacement. The last class was involved only in consonant production. Nevertheless, vowels and consonants were not found categorically different in their methods of production (see the section on EPG (2.2.4, page 41) for more on this subject).

In the latter paper (Lundberg & Stone, 1999) they describe a method for reconstructing the tongue surface from a sparse coronal slice set. They achieved at best an approximate 90 % coverage of the tongue surface by selecting an optimal set of six coronal slices for tongue reconstruction. As they used the data from the previous study, they had a maximum of 60 slices to choose from with fewer slices for some speech sounds as a result of the tongue's position in relation to the fixed slice planes.

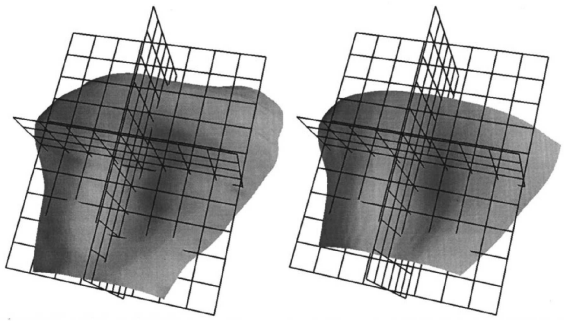


Figure 2.4: Tongue surfaces for [l] reconstructed from a dense (left) and sparse (right) set of ultrasound slices (Lundberg & Stone, 1999).

Lundberg and Stone (1999) used two sets of slices for the sparse reconstructions. The first one was determined by the six optimal points for reconstructing the sagittal contour, while the other set was determined by the optimal slices for reconstructing of the whole surface. Their overall results were good in three out of four of their criteria: Local depressions or dimples of the tongue surface, steep slopes and fricative constrictions were captured well by the sparse reconstructions. However, capturing tongue asymmetries (asymmetrical articulation) needed improvement.

### 2.1.4 Magnetic Resonance Imaging (MRI)

Magnetic resonance imaging or nuclear magnetic resonance imaging (NMR imaging) to give its full name[2] utilizes magnetic resonance of hydrogen nuclei to produce a tomographic image of the object being studied. In order to produce meaningful data the spins of these nuclei have to be aligned in the

[2]"Nuclear" was dropped out of use in medical contexts and hence articulation study contexts. This was done as the use of "nuclear" falsely suggested to patients and subjects, that the method would use ionizing radiation.

same directions. This is accomplished by the application of a very strong magnetic field. The actual imaging procedure involves radio frequency oscillations of the magnetic field in order to produce a measurable echo field from the nuclei. The gathered data is then processed with mathematical inversion methods to produce the tomogram.

MRI does have some drawbacks most of which can be overcome with careful planning or by the use of state-of-the-art equipment. First and foremost should be mentioned, that the subject's safety has to be considered carefully. As the method involves very strong alternating magnetic fields the patient should not have any metallic implants or extraneous metallic material within his or her body. The metallic material may start to heat up and cause injuries or, in the case of small particles, it may even move back and forth with the changes in the magnetic field and thus damage the tissue it moves through. These same restriction apply to any extra equipment used within the magnetically shielded scanning room. While the no metal restriction is not absolute it has been found that for example the wires for a normal microphone tend to pick up noise from the oscillating magnetic field, which often renders the recordings useless.

Another problem is the acoustic noise level within the scanning room. Modern MRI scanners produce noise through the magneto strictive effect during rapid field changes which occur during the scanning procedure. This poses an other problem for simultaneous sound recording. (See below in this section 2.1.4, page 30 for possible solutions to this problem.)

Finally there are three problems with the imaging itself. These are the typically long acquisition times needed for full 3D scans of the VT, the occasionally poor air-tissue contrast and the fact, that MRI does not produce practically any signal from calcified structures such as bones or teeth. The acquisition time and the air-tissue contrast problems are disappearing with advances in equipment technology and imaging protocols. In contrast, the problem with bones and teeth is inherent to the way MRI works. It stems from there being very little hydrogen, which would be capable of producing the necessary echo, in the calcified tissues.

However, when used correctly MRI does not cause any known health risks. Indeed, when compared with the cineradiographic techniques which have been very popular until late 1980's, MRI has two very important advantages. Despite its original name it does not use ionizing radiation. It

can, therefore, be used for large corpus studies and for repeated measurements with the same subject. In addition, MRI is a volumetric imaging technique, which is able to produce images with good spatial resolution at least with modern equipment. A further bonus is the possibility of choosing the angle and plane of the tomograms fairly freely.

### Studies

Baer, Gore, Boyce, and Nye (1987) and Baer et al. (1991) report studies with two scanners: an experimental one (both papers) and a commercially available one (the latter paper). The experimental scanner was a whole body scanner with a 0.15 T (Tesla) resistive magnet. The studied material Baer et al. (1987) was sustained vowels with recordings of two subjects in supine position.

Baer et al. (1987) acquired transaxial and coronal tomograms (slices) during different sessions, but the coronal slices were found to be of too poor quality to be useful. For the transaxial set the slice thickness was 8 mm. The set consisted of slices in a range of 9.5 cm with 0.5cm between each slice - 19 images in all. With later improvements (Baer et al., 1991) the same device was used to record also sets of coronal images for the same subjects. This set consisted of 18 images with same specifications as for the transaxial one. In both cases the image resolution was 256x256[3]. The acquisition time for one slice 3.4 minutes. This meant that the whole VT was imaged in 3 to 4 hours.

As the scanner was an experimental one, the experimenters were able to do some customisation. They used a head mold to minimise head movements and to support the registration coils of the scanner. Two other custom fittings were possible, since the the low field experimental scanner did not produce too much acoustic noise. It enabled the use of a regular microphone to record the sounds produced by the subjects. The microphone was connected to a battery powered amplifier inside the scanner room which in turn was connected to the recording equipment outside of the scanner room. A canonical target sound was also played to the subjects through earphones. The sound was carried to the subjects along a hollow plastic tube from outside of the scanner room.

[3]Field of view (FOV) was not reported.

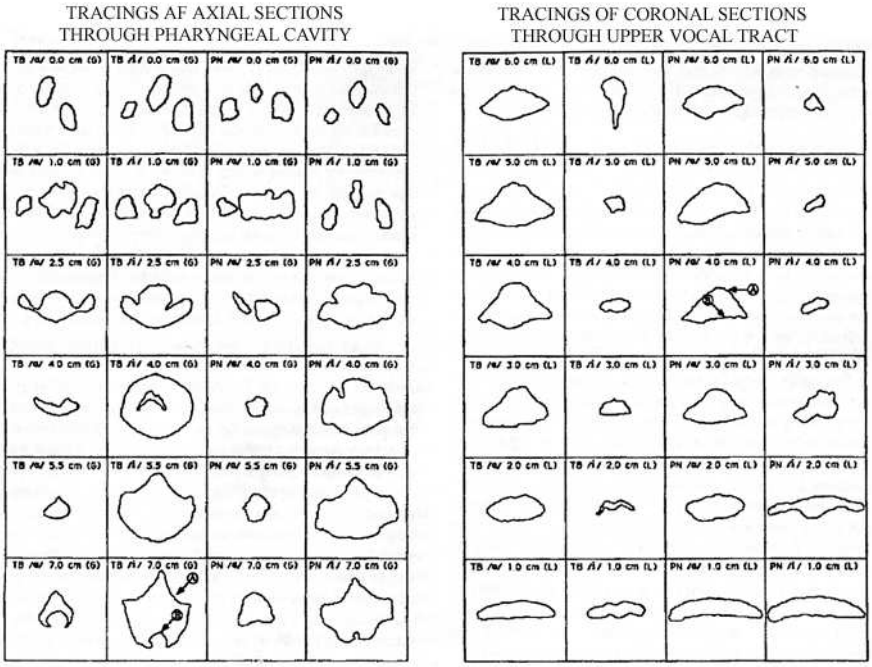



Figure 2.5: An example of the VT contours obtained by experimental MRI (Baer et al., 1991). The tracings have either the anterior direction (on the left side) or the vault of the palate (on the right side) on top. The distances indicated on the left are approximate distances from glottis and on the right approximate distances from the lips. In the picture on left, the columns are from left to right: 1. subject TB pronouncing [a], 2. TB [i], 3. PN [a], 4 PN [i] and the arrows indicate (A) tongue grooving and (B) the uvula. In the picture on the right, the columns are in the order as on the left and the arrows indicate (A) dome of the palate and (B) the surface of the tongue.

The VT was traced manually from the tomograms. Figure 2.5 shows examples of the tracing results. Teeth were taken into account manually with help from X-ray data and visual observation.

In Baer et al. (1991) a General Electric Signa scanner was used for a second study. The machine had a 1.5 T superconducting magnet and hence quite different conditions and properties when compared with the experimental scanner. The whole VT could be imaged in less than 30 minutes with a multislice acquisition procedure (17 tomograms in less than 3.4 minutes), but on the downside this generated a loud acoustic noise. Consequently the produced sounds could not be recorded and neither could a

canonical target be played to the subjects. Image accuracy was roughly the same as with the experimental scanner with a field of view of 20 cm and resolution of 256x256 pixels. The mid-sagittal plane and the two orthogonal planes were imaged with slice thickness of 8 mm and slice interval of 5 mm.

Worse signal to noise ratio at extremes of the imaging area resulted in a $\pm$6 mm uncertainty of VT length at both ends. With calibration from X-ray images this was reduced to 1-2 mm. The cross-sectional areas of the oral cavity were also checked with dental casts. See above (2.1.1, page 14) for more details.

Narayanan et al. (1995), Bangayan, Alwan, and Narayanan (1996), Alwan, Narayanan, and Haker (1997), and Narayanan, Alwan, and Haker (1997) studied fricatives, laterals and rhotics with a General Electric 1.5 T Signa scanner.

In the 1995 study they used a fast radio frequency spoiled GRASS protocol and a special head-neck receiver coil (by Medical Advances) in data acquisition . The data was recorded on two male and two female subjects. Image resolution was 256x256 with a pixel resolution of at worst about 0.94 mm/pixel. The FOV was either 24 or 20 cm. (With 20 cm resulting in better pixel resolution.) The slices were 3mm thick and were recorded without interslice spacing.

The whole VT was imaged in coronal, axial and sagittal planes. The image set consisted of 28 to 35 images in the sagittal plane and 40 to 45 images in the two other planes. Each set of each subject was recorded in a separate session of 1.5 to 2 hours. The subjects sustained the sounds for 13-16 seconds to facilitate the acquisition of 4-5 slices (about 3.2 seconds each). An image set (sagittal or coronal) of the whole VT was thus obtained with 6-9 repetitions.

Automatic contour extraction was followed by manual correction of extracted contours. To correctly include the decrease in area caused by teeth Narayanan et al. (1995) tried several methods aimed at making the them visible in the MR images. As all of these (application of mineral oil, paraffin wax or EZ-paste) proved unsatisfactory, they used instead data obtained from dental casts. See section 2.1.1, page 14 for more details.

In Bangayan et al. (1996) a study of [ɬ] and [l] is reported. The same setup and equipment as in Narayanan et al. (1995) was used. Narayanan et al. (1997) reports the study in further detail and Alwan et al. (1997) adds material on [ɹ].

Badin, Bailly, Raybaudi, and Segebarth (1998) and Badin et al. (2002) (see Badin, Bailly, Elisei, and Odisio (2003) for a summary of the latter study) report two similar studies with essentially the same MRI setup. They studied consonants (in a [VCV] context) and vowels ([V]) with a 1.0 T Phillips Gyroscan T10-NT scanner. With it 55 (53 in the 2002 study) slices were obtained to image the whole VT. Each image set consisted of a coronal subset in the oral region, a oblique subset in the pharyngeal region and a axial subset in the laryngeal region. The slices had a thickness of 3.6 mm and were sampled every 4.0 mm. Final pixel resolution was 1 mm.

To obtain a complete set of the VT the subject had to artificially sustain the articulations for 43 seconds. This was done either in full apnoea or by breathing out very slowly.

In the 1998 study (Badin et al., 1998) the image data was transformed to the usual semi-polar system. The transformed stack can be seen in figure 2.6. After this the VT contours were detected automatically with a threshold operation. The resulting contours were then sampled with 51 points and low-passed in x- and y-directions in order to smooth them.

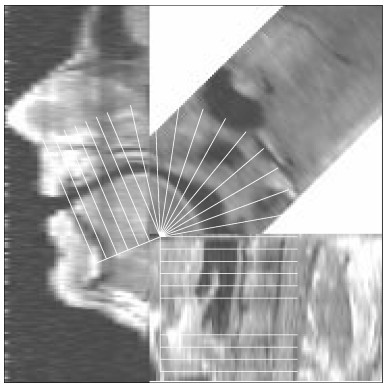


Figure 2.6: An example of a reconstructed semi-polar image grid position displayed on a mid-sagittal image composited from the original image stacks (Badin et al., 1998).

In the study reported in Badin et al. (2002) the image data was transformed similarly as in the previous study, but this time the grid was dynamically adjustable (see Beautemps, Badin, and Bailly (2001) for details). In addition, the VT contours were traced and divided into functional parts (tongue, epiglottis etc.) manually with a B-spline editor. Teeth were included in the model with a combination technique involving dental casts and MRI (see section 2.1.1, page 14). After fitting the image stacks together they were resampled evenly with 80 points along the tongue contour and 22 slices altogether.

Engwall and Badin (1999) recorded consonants ([vCv]) and vowels ([V]) with the scanner used by Badin et al. (1998). The whole VT was imaged in 43 seconds and the mid-sagittal plane in 11 seconds. This involved sustaining the sounds in either full apnoea (vowels and stop consonants) or by breathing out very slowly (fricatives). The entire corpus was recorded in one session.

Amount of images used to record the whole VT as well as their orientation was identical with Badin et al. (1998) and Badin et al. (2002). The image orientation was similar to the one shown in figure 2.6. The teeth were included by imaging dental casts with MRI (see 2.1.1, page 14).

The 3D images were analysed mostly as in Badin et al. (1998) and Badin et al. (2002). Indeed, the only truly notable differences with the above studies by Badin et al. were that after a thresholding operation the boundaries were extracted with chained pixel search and finally defined as a Bézier controlled splines. The contours were checked manually to correct errors resulting from a slightly varying brightness in the original pictures.

The sagittal images were processed much in the same way as the 3D images. The thresholded sagittal images were subjected to a Matlab tool to extract VT contours. Similarly to the 3D images the sagittal images were subsequently hand corrected.

#### Validity of Sustained Articulations

Engwall has evaluated the validity of MRI measurements in two studies (Engwall, 2000b), (Engwall, 2003). See also (Engwall, 2006) for a more recent treatment. In the first study Engwall compared static MRI data with

EMA and EPG data (the first acquisition is treated above (Engwall & Badin, 1999) and the second and third are detailed below in sections 2.2.3, page 37 and 2.2.4, page 41, respectively). He found that the sound were hyperarticulated during static recordings and coarticulation had a diminished effect on the tongue. Further, the static articulations had greater lip protrusion and jaw opening along with a more neutral tongue shape. In the paper reporting the recording of the MRI data he also mentions that - quite naturally - the velum was lowered due to apnoea (Engwall & Badin, 1999).

In the latter paper (Engwall, 2003) results from MRI recordings of artificially sustained, normally sustained and regular articulation are compared. Artificially sustained articulation involves full apnoea while normally sustained articulation is normal phonated speech which is only prolonged. Both of the sustained articulation conditions could be imaged in full 3D, but regular articulation had to be obtained in only 2D to make real-time MRI feasible.

The acquisition oft the data set on artificially sustained articulation is detailed in Engwall and Badin (1999)(see this section above), while details on the acquisition of the normally sustained and real-time data sets are in the paper itself (see section 2.2.6, page 45 below).

Effects of artificial sustaining were tested with long isolated vowels and voiceless fricatives in a vowel ([vCv]) context. Differences between real-time and sustained articulations were tested with the latter type of utterances. Gravitational effects were tested with isolated long vowels and real-time utterance of [aiu]. To obtain the gravitational test data the subject was recorded while lying face up and face down.

Significant effects were found in relation to artificial sustaining, coarticulation and gravitation. Specifically, artificial sustaining caused supralaryngeal VT and lower pharynx to be much narrower, made the difference in advancing of the tongue smaller between vowels and made the oral part of the tongue contour less varied.

Similarly, there were two major differences in coarticulation between sustained and real-time data. Firstly, tongue coarticulation was much greater in real-time, whereas, jaw and lip coarticulation was smaller in real-time data.

Gravitation was found to effect pharynx width in two of the studied vowels [ɑ, u]. In addition, some differences in the position the tip of the tongue were observed in [ɑ].

Engwall (2003) concluded, that artificial sustaining of articulations caused hyperarticulation and that the articulations were difficult to maintain correctly. Furthermore, sustained articulation caused hyperarticulation in general and hypercoarticulation in particular. Finally, the supine position was found to have an effect, which was made more pronounced by the degree of sustaining. However, static MRI recordings can be used as target articulations with fair confidence (Engwall, 2000b). Hence, they should be considered useful in spite of the unwanted effects.

#### Sound Recording during MRI

Regular sound recording is an obvious part of at least data gathering for confirmatory tests of acoustic quality of produced speech. As seen in for example Engwall and Badin (1999) sound can be recorded before and after each MR image acquisition with a microphone, if a screened cable is used.

Nevertheless, sound recording during MRI does pose some challenges. In particular, if speech should be recorded during an MRI sequence, the over all conditions are quite difficult (Demolin, Metens, & Soquet, 2000). Firstly, there is a high level of noise from the MRI device itself. Secondly, nothing of ferromagnetic nature may be present in the room, where the device is housed.

The first restriction means, that the sound recording system has to either be able to separate the background noise from the speech and be directional enough not to record too much of the noise in the first place. The second restriction makes the use of regular microphones impossible. Two possible solutions are optical microphones or a regular microphone used on the outside of the device's field, with a sound carrier tube carrying the sound from a pneumatic mask worn by the speaker. The use of a pneumatic mask has been proposed by at least Demolin, George, Lecuit, Metens, Soquet, and Raeymaekers (1997).

However, the most promising results to date come from an optical differential microphone system, which has been put to actual research use

by Ericsdotter (2005). The system consisted of a light reflecting membrane, whose vibration was measured optically. The optical signal was then transduced to an electrical one at a safe distance from the MRI scanner and after amplification recorded with a DAT recorder. After a post recording noise filtering stage a vowel's fundamental frequency and strongest formants could be measured.

## 2.2 Dynamic Methods

These are methods capable of capturing articulatory movement. This ability comes with a cost in spatial resolution or in the necessity to restrict the method to only two dimensions or even to a set of points.

Some of the considerations relating to static methods (see page 12) are also valid for dynamic methods. In particular, subject fatigue and undesired movement of subject during acquisition are universal problems in data acquisition. In contrast, some error sources are more common with the dynamic methods than static ones. One of the most important is unnatural articulation resulting from equipment, which has to be placed within the subjects oral cavity.

### 2.2.1 Cineradiography

Until recently, as can be seen from the bibliography by Dart (1987), cineradiography has been the method of choice for studying speech movements. Cineradiography uses ordinary X-ray apparatus. Instead of taking still pictures the apparatus is used to record a cinematic sequence of X-ray pictures with a specialist camera[4]. While the procedure can produce interesting data, it has some serious drawbacks.

The most important drawback and the reason for cineradiography's decrease in popularity is radiation. As awareness of the long term effects of ionising radiation has increased, the regulations on the maximum allowed dose have become stricter. This can be considered only a good thing, since the safety of experimental subjects should always be a prime consideration.

[4]As cineradiography has its roots in ordinary X-ray studies of still pictures I have included some of those studies under this heading.

It does, however, make it very hard to gather a large corpus of data with cineradiography.

Another important drawback is the fact that cineradiography is a 2D method. Hence, the data is customarily recorded only in the mid-sagittal plane. This leaves any asymmetries out of the data as well as making it hard to judge changes in the coronal direction such as grooving. This adds to the difficulty of modeling the articulatory movements of certain sounds such as [l]. Moreover, as the teeth and certain parts of the tongue are quite often projected on top of each other in a mid-sagittal projection, it is sometimes hard to make out the correct contour of the tongue from the images.

On the positive side, the recording position in cineradiography is usually very natural and obviously the method is dynamic in nature. Thus, the gathered data should be quite relevant. It is also good to note, that while new studies may not be undertaken lightly data from old studies is often available in a useful form (see eg. (Munhall, Vatikiotis-Bateson, & Tokhura, 1995)). Furthermore, advances in imaging techniques facilitate gathering of a larger corpus of data than before with the same radiation dose.

#### Studies

In a classic study Fant (1970) used radiographic data obtained by MacMillan and Kelemen (1952). The subject was a male speaker of Russian and the corpus included pictures of the articulation for sounds produced in the manner of "Say [s] as in [sat]".

A barium paste (of barium, water and mucilage of acacia) was applied to the subjects tongue and palate to enhance contrast. The subject's position was ascertained with the use the equipment in figure 2.7. The subjects lip shape was obtained from photographs taken simultaneously to the X-ray recording and the position of vowel formants was verified from a simultaneous sound recording. Finally, the X-ray information was supplemented with slices of plaster casts of the subjects mouth cavity.

Mermelstein (1973) used data obtained by Perkell (1969). It consisted of sagittal cineradiography data on [həCV] material. The corpus included about 20 frames each on 8 utterances and one sentence (2.5 seconds) of

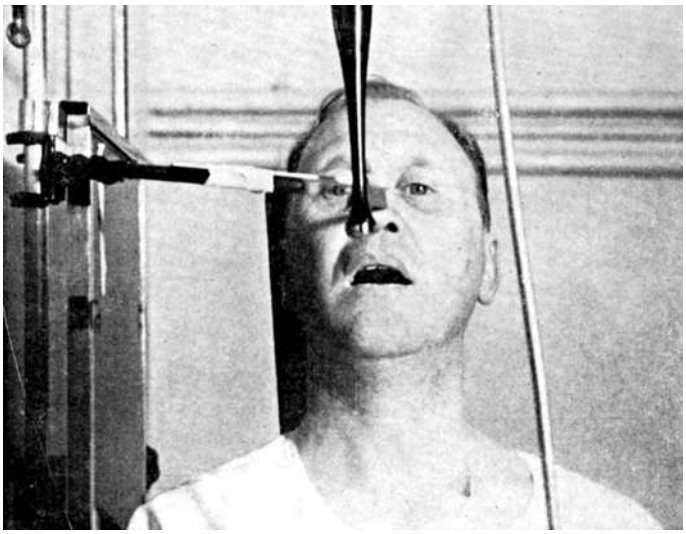


Figure 2.7: Equipment used to position a speaker for X-ray (Fant, 1970).

120 frames. This study was one of the first to use a semi-polar grid in converting the mid-sagittal cross-distance to cross-sectional area.

Gay (1977) studied [CVCVC] material with two male subjects. Gay used in his study a cinecamera at 60 frames per second and had an X-ray generator produce 1 ms pulses. In order to enhance the quality of the obtained data Gay used 2.5 mm lead pellets to tag six points on the lips and the tongue and an additional reference point on the upper incisors. This experimental setup has obvious similarities with the X-ray microbeam (see section 2.2.2, page 35 below) and electromagnetic articulography (see section 2.2.3, page 37 below) methods.

Harshman, Ladefoged, and Goldstein (1977) applied PARAFAC analysis (Harsman, 1970) to tongue shape data. The raw data was obtained as cine-fluorograms (x-rays) of five speakers uttering English vowels. The VT shapes were then traced from the cine-fluorograms chosen to represent the vowels. The next step was to fit a reference grid to each speaker and use it to measure the VT cross-sections at 18 points in the sagittal plain.

Maeda (1990) gathered simultaneous cineradiography, labiofilm and sound data in a study, which involved two female subjects. The corpus included ten French sentences. The VT was manually traced from the films and sampled for analysis with the semipolar grid in figure 2.9 (a). Next the grid and wall intersections were detected automatically. As a final step

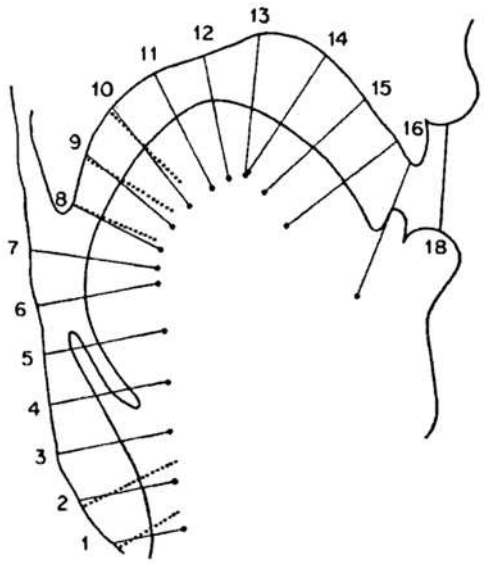



Figure 2.8: Semipolar grid used by Harshman et al. (1977) to measure the VT's cross-dimension from cineradiographic data.

of the preprocessing the position of the end of larynx and lip shape were processed separately as the basic grid is not applicable to these parts.

Stark et al. (1999) used cineradiography in synchrony with sound recording. They used a digital X-ray facility which had a frame rate of 50 per second and an acquisition time of 3 ms/frame and an image resolution of 0.3 mm.

Stark et al. (1999) used a customised setup to minimise radiation to the subject. This involved a prespecified pediatric program and removal of the image intensifier scatter-radiation grid from the device. Dosimeters were used to control actual radiation levels. These safety measures resulted in an absorbed dose of less than 4 mGy in the most exposed organ - less than 0.1 mSv as an effective dose (tenth part of annual background dose).

By using tagging measures Stark et al. (1999) achieved a high image quality. They attached a copper wire to the subject's hard palate and applied contrast paste to the lips and the mouth region (see figure 2.10). VT contours were extracted manually with the aid of medical software, but the images were first automatically rescaled, and corrected for optical distortion and head movement.

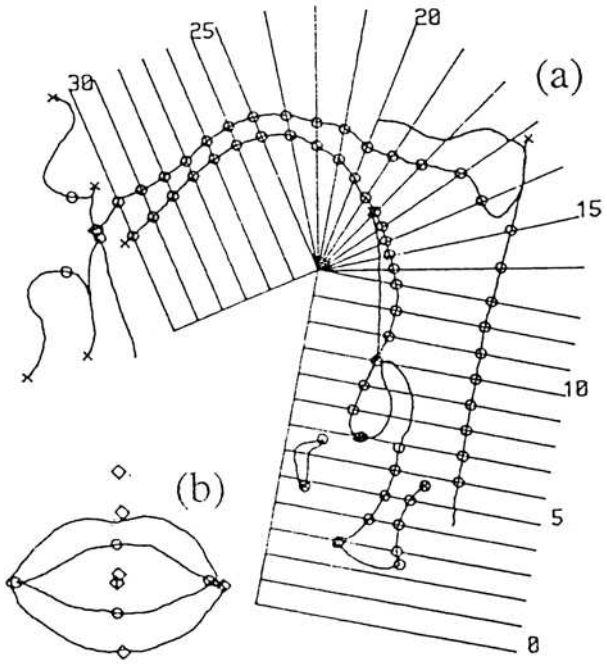



Figure 2.9: Semipolar cineradiography grid (a) and lip measurement points (b) from Maeda (1990).

In a study resembling that of Maeda (1990), Beautemps et al. (2001) recorded synchronous cineradiographic labiofilm and sound data. The corpus included [VCV] (voiced consonants) and [...VVV...] material. Both filming methods had a framerate of 50 images per second.

The X-ray images were first drawn on paper by hand from a projection and subsequently digitized and split into eleven subcontours. The analysis was facilitated with the usual semipolar grid (see figure 2.11), which was extended with dynamical modification possibilities for the lips and larynx region.

### 2.2.2 X-ray Microbeam

X-ray microbeam is a point tracking method developed originally by Kiritani, Itoh, and Fujimura (1975). It is based on ordinary X-ray technology. However, instead of imaging the whole VT only small lead or gold pellets are tracked. The tracking is performed with a very small X-ray beam. The

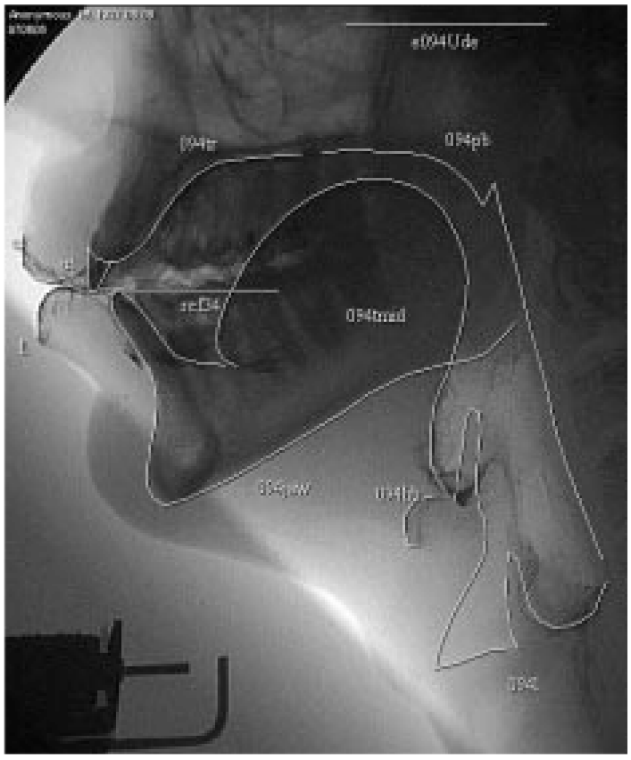


Figure 2.10: Sample picture with traced VT from Stark et al. (1999) during articulation of [u].

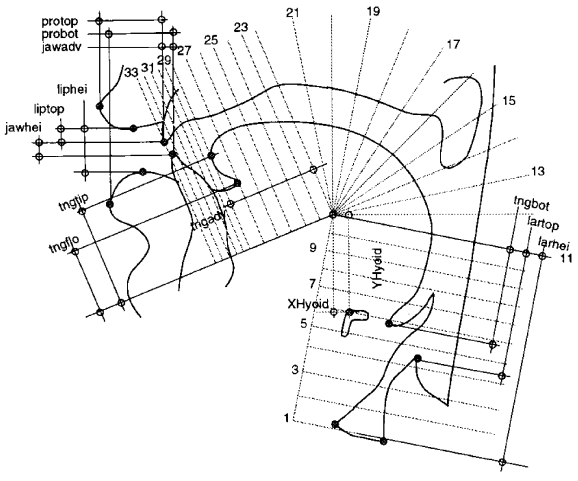



Figure 2.11: Dynamical semipolar mid-sagittal grid used by Beautemps et al. (2001) with relevant lip, jaw, tongue, hyoid etc. parameters.

beam's penetration of the tissues and the pellets is registered with a digital detector array on the other side of the object as seen in figure 2.12.

Before tracking begins the pellets are first located by scanning the whole image area. In contrast, during tracking only a neighbourhood around the pellets' last locations is sampled. This reduces the amount of radiation exposure greatly. The only problem with this is that specifically tissues around the pellet and its projection absorb most of the radiation. Even so, the amount stays quite small even during a recording of a comparatively large corpus of data.

The early system had a spatial resolution of 1mm and an effective sample rate of 100 Hz (Kiritani et al., 1975). The pellet diameter was originally 3 mm, but dropped to 2.5 mm as the material changed from lead to gold (Fujimura, 1991).

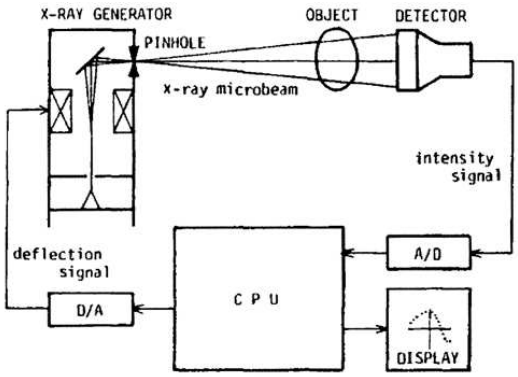



Figure 2.12: X-ray microbeam system diagram (Kiritani et al., 1975).

### 2.2.3 Electromagnetic Articulography (EMA)

Electromagnetic Articulography is a point tracking method capable of tracking several tagged points on the face, teeth and tongue of a test subject. The tracking is based on measuring changing magnetic fields at the tracked points. The fields are produced by transmitter coils, each of which has its own field frequency. The position and movement of the tracked points can be solved by measuring the signals produced by induction in small receiver coils, which are used to tag the points.

EMA works by estimating from the measured current strength the distance from the receiver coil to the transmitter coil. The process involves estimating the local field strength at the receiver from the current strength and computing the distance thereafter from the field strength. Problems arise if the estimated field strength is incorrect, which, in turn, is caused by misalignment of the receiver. To work well, EMA requires that the receiver and transmitter coil's main axis be parallel as in figure 2.13. If this is not the case, the distance will be overestimated.

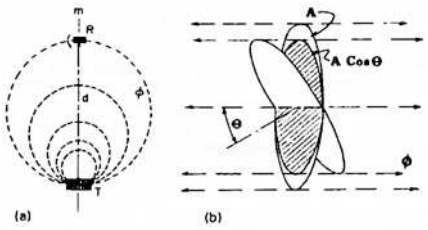



Figure 2.13: The operation principle of EMA (Perkell et al., 1992). The magnetic coupling of two solenoidal coils is illustrated in (a) and (b) shows how misalignment with the magnetic field effects one winding of a transducer coil. Here T is the transmitter coil, R the transducer coil, $d$ the distance between them, $\Phi$ the magnetic flux, $A$ the area of a correctly aligned winding, and $\theta$ the angle of misalignment. In (b) the effective area of the misaligned winding is thus only $A \cos \theta$ instead of $A$.

The greatest shortcomings of EMA lie with what can be measured: EMA is usually confined to the mid-sagittal plane and tracks only points. With the traditional devices capable of measurement in one plane the method is also susceptible to error caused by tilting of the receiving coils in relation to the measurement plane. While the restriction to the mid-sagittal plane is becoming a thing of the past, EMA is by nature a point-wise measurement method an this is unlikely to change. Furthermore, the method cannot be used very deep inside the vocal tract as the coils and wires involved would trigger the subject's gag reflex.

Other significant problems are that in order to work the receiver coils have to be connected with wires to an analysis unit. This means, that when measuring intra-oral articulation, there will be wires passing into the subject's mouth. In addition, the articulation's naturalness may be affected by the receiver coils themselves. This is evident from audible deviations on

some sounds, for example [s] (Engwall, personal communication).

Nevertheless, EMA is a very popular method for measuring articulation. The main causes for its popularity are its safety, its dynamical nature and the fact that after the initial cost of acquiring the equipment it uses is cheap. In addition, EMA is well used and hence well understood method with other less obvious good qualities. For example it can be used simultaneously with EPG (see section 2.2.4, page 41 below).

### Development of the Method

Probably the first reported attempt to build an EMA device is that of Hixon (1971). The device worked inversely in comparison to the modern ones: The magnetic field was generated at the point of interest and measured with static coils. The coils were very clumsy with a length of 2 cm and diameter of 5 mm. The device was capable of 1- and 2-dimensional measurements.

In contrast, Lance and van der Giet (1974) reported an experimental EMA system with two pairs of transmitting static coils and one tracked receiving coil. The tracked coil in this system was already considerably smaller with a length 4 mm and a diameter of 2.8 mm. The system was capable of tracking the position of the receiving coil in real time.

One of the first commercially available systems - called Movetrack - is described by Branderud (1985). The dimensions of the transducers (receiving tracked coils) are 1.5 mm $\times$ 4 mm. The two transmitter coils are fixed to a special helmet. With the original setup the system tracked three transducers in the mid-sagittal plane, but had a theoretical maximum of 12 transducers. A Movetrack system used simultaneously with an EPG system can be seen in figure 2.16, page 43 below.

Perkell et al. (1992) describe two experimental EMA devices: The other with two transmitters and the other with three. The two experimental devices are evaluated and compared with two commercially available systems.

The experimental devices took two different approaches in dealing with the misalignment problem: The three transmitter device used the extra

transmitter's signal for alignment correction while the two transmitter device used biaxial receiver coils to actually measure the alignment. The biaxial coils consisted of two normal EMA coils mounted at right angles to each other. In tests the three transmitter was found to be preferable.

The preference was due to a more difficult calibration process, higher price and higher field strength associated with the two transmitter device. In general terms the experimental devices had low errors: maximum spatial distortion was 0.5 mm and maximum rotational distortion was less than 0.9 mm for the two transmitter device and less than 2.0 mm for the three transmitter device. Both devices operated at a sample rate of 100 Hz.

The commercial systems tested by Perkell et al. (1992) were AG100 from Carstens Medizinelektronik and Movetrack from Special Instrument AB. Either one was found to be a good choice depending on the choosing criteria: While Movetrack was cheaper, it required more work before it could be used for experiments as it did not have analysis software bundled with it.

Recently, an EMA device - called Carstens AG500 - which is not restricted to the mid-sagittal plane has been in development (Zierdt, Hoole, Honda, Kaburagi, & Tillman, 2000), (Hoole, Zierdt, & Geng, 2003). Figure 2.14 shows the transmitter coil setup of the system and an actual experimental situation. With each transmitter transmitting a different frequency the system is able to track each receiver in 3D and provide two rotational coordinates for them as well.

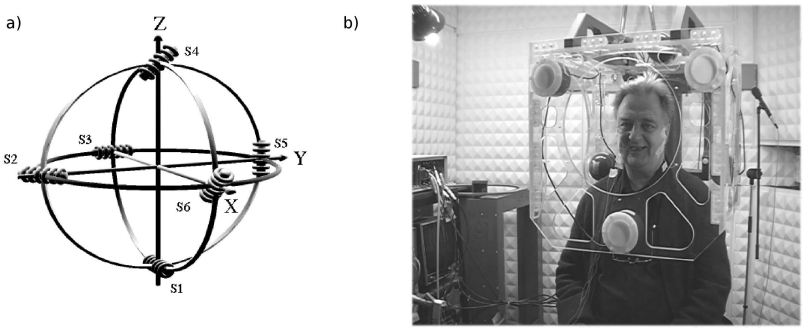



Figure 2.14: 5D EMA setup: a) schematic picture of the transmitter coil setup and b) the actual experimental setup. Both from Zierdt et al. (2000).

The prototype is reported to have a spatial resolution which is better than 1 mm and a rotation resolution of about 1°. The prototype was at first able to track 4 receiver coils (same as the ones used with Carstens AG100), and later reported to track 5 (Hoole et al., 2003). As the head moves freely inside the measurement area, the head position has to be accounted for by reference coils.

An interesting resource of EMA data - the MOCHA database is described by Wrench and Hardcastle (2000). MOCHA (Multi-CHannel Articulatory database) contains EMA, EPG and laryngograph data from 40 English speakers reading a maximum of 460 TIMIT sentences. In particular, MOCHA contains EMA data from the subjets' lips (two coils), jaw (one coil), tongue (three coils), and soft palate (one coil). The EMA data has been recorded at 500 Hz. See section 2.2.4, page 41 for details of the EPG data.

### 2.2.4 Electropalatography (EPG)

Electropalatography is a method for dynamically measuring the tongue-palate contact. The measurement is carried out with a thin artificial palate, which contains the pressure sensors, that detect tongue contact. The contact pattern can be sampled with a high frequency by attached analysis equipment. An example of an artificial palate is shown in figure 2.15. (Do note, that different manufacturers of EPG equipment use different sensor configurations.)

EPG is limited by the fact, that subjects have to get used to wearing the artificial palate before reliable measurements can be made. This adds to the cumbersomeness caused by custom fitting an artificial palate for each subject - EPG is certainly not a spur-of-the-moment type of investigation method. In terms of acquired data, EPG is limited only to contact information. No information is gained on the tongue-palate distance or movements of other articulators.

The main good qualities of EPG are its safety and fast sampling rate - usually either 100 Hz or 200 Hz. The spatial sampling is also quite dense with electrode numbers ranging from 60 to almost 100. Like EMA, it is also a well used and well understood method, which produces results, which

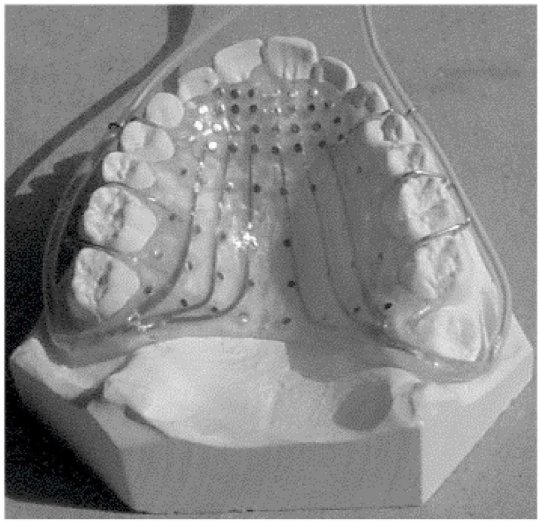


Figure 2.15: An artificial EPG palate (produced by Reading) mounted on a dental cast. (Engwall, 2000c).

facilitate comparison with other studies. Also as mentioned above (see 2.2.4, page 42 for further details) it can be used simultaneously with EMA.

When studying general linguistic phenomena such as coarticulation, EPG data can be reduced and simplified in many ways (see for example (Hardcastle, Gibbon, & Nicolaidis, 1991)). In contrast, when using EPG as a data source for articulatory speech synthesis its greatest value often lies in fitting artificial contact data from the model with the real EPG data (for example (Cohen, Beskow, & Massaro, 1998) and (Engwall, 2001b)).

As mentioned above in section 2.2.3 the MOCHA database described by Wrench and Hardcastle (2000) contains also EPG data. The EPG part of the database has been recorded with 62 sensor artificial palates with a sampling frequency of 200 Hz.

#### Combining EMA with EPG

Simultaneous EMA and EPG recording has been firmly established as a good method for obtaining dynamic data on articulation (Hoole, Nguyen-Trong, & Hardcastle, 1993), (Engwall, 2000c), Fuchs and Perrier (1999). The combination has been found a good match as EMA provides data on the constriction size and EPG tells whether the situation in the mid-sagittal

plane extends to the sides of the cavity as well. The combination is particularly useful in studying vowel-consonant coarticulation.

The reservations of both methods have to be taken into account and an additional consideration is in order when using EMA and EPG concurrently (Engwall, 2000c): Even though it is considered quite unlikely, the data may contain interference from one method to the other. Palate contact may move the EMA coils or coil contact register as in EPG even if normally (without simultaneous EMA) no contact would be found. These problems can be ruled out by separate control registrations with both methods by themselves.

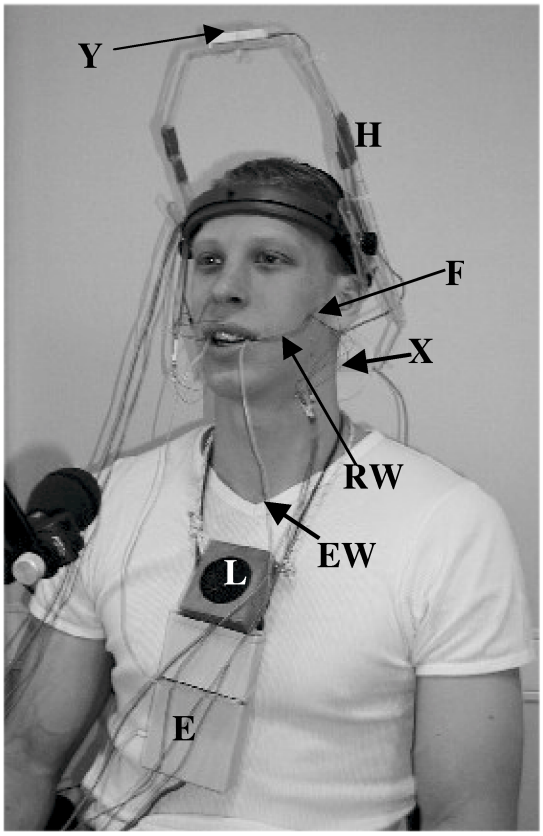



Figure 2.16: Concurrent recording of EMA and EPG (Engwall, 2000c). The labels are as follows: X = X-direction transmitter, Y = Y-direction transmitter, H = Head mount, RW = Receiver coil wire, E = EPG unit, EW = EPG wire, L = Loudspeaker for synchronisation signal, and F = Fastening of EMA wires with tape.

### 2.2.5 Optopalatography (OPG)

The use of optical distance sensing equipment to study intra-oral articulation was first suggested by C-K. and Wang (1978) and later developed under the name Glossometer by Fletcher, Dagenais, and Critz-Crosby (1991). Optopalatography aims to broaden data gained from palatographic measurements by providing tongue-palate distance data as well as readings on the firmness of tongue-palate contact.

Wrench et al. have developed a working prototype OPG device (Wrench, McIntosh, & Hardcastle, 1996), (Wrench, McIntosh, & Hardcastle, 1997), (Wrench, McIntosh, Watson, & Hardcastle, 1998). A schematic of the general system is shown in figure 2.17.

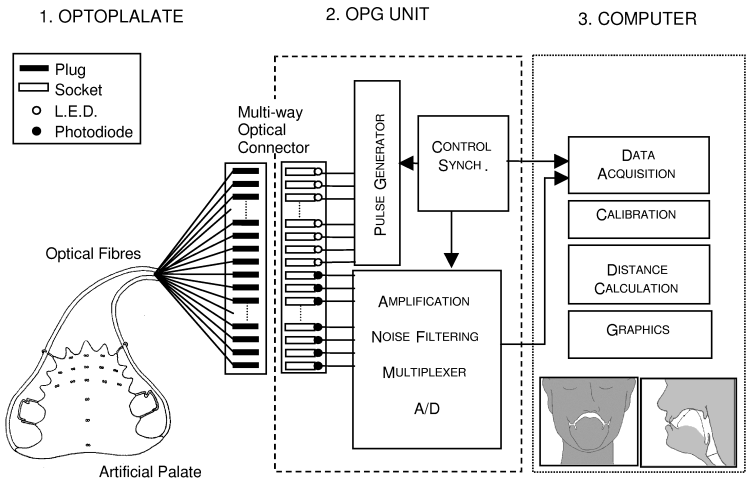



Figure 2.17: The OPG setup (Wrench et al., 1998).

As is evident from the name OPG is a descendant of EPG. However, instead of mounting contact sensors on the artificial palate, optical fibres have been mounted there. These are used in pairs: One fibre carries the light produced by the OPG unit to the palate and sends it towards the tongue and the other catches returning light and transfers it back to the OPG unit for analysis.

The most important limitations for OPG system design are caused by the qualities of the optical fibres. Fibre diameter limits the number of possible measurement points as the size of fibre bunch, that can be comfortably relayed out from the corners of a subject's mouth is quite limited. On the

other hand fibres with a very small diameter can not be used as they have to be bent 90° within the thickness of the artificial palate.

Another important limitation is the signal to noise ratio (SNR). SNR is affected by light source strength, fibre qualities and possibly by the proximity of the sensor/source pairs in the artificial palate. The proximity problem can be avoided by operating the points in sequence with only one sensor/source pair active at any given time.

When compared with EPG, OPG has two additional good qualities: It produces tongue-palate distance measurements and can be used for force measurements, since all of the light is not actually reflected by the surface of the tongue, but also from within it. Thus the firmer the contact - and the greater the force - the stronger the reflection. In addition, when compared with the Glossometer, OPG clearly facilitates more measurement points.

The latest prototype (Wrench et al., 1998) had 16 sensors built from 0.5 mm plastic optical fibres. The system had a measurement range of 20 mm and sample rate of 100 Hz. The light sources were infra-red LED sources and they were used in sequence.

### 2.2.6 Fast MRI

Fast MRI differs from regular MRI in terms of imaged area and temporal resolution. The imaged area is always smaller in fast MRI than in static MRI - usually it is restricted to a single mid-sagittal slice of the VT. With this reduction comes the advantage of speed: Instead of using several seconds or even tens of seconds to image the whole VT, one slice can be acquired in less than a second and with suitable imaging protocols even at rates approaching 10 Hz. On the other hand by averaging several repetitions of the same utterance even greater temporal resolution is achievable.

#### Studies - Real Time MRI

In two similar studies - Demolin et al. (1997) and Demolin et al. (2000) - an ultrafast implementation of Turbo Spin Echo (TSE) sequence was applied to achieve MR imaging of the VT with one mid-sagittal, 6 mm thick slice at 4 Hz. The sequence was previously known as TSE Lolo (Local Look), but currently called TSE Zoom. In the first study the FOV was $300 \times 150$ mm$^2$

and the pixel matrix 32 × 128 pixels, and in the second study the FOV = 125 × 250 mm$^2$. Figure 2.18 shows two pictures acquired in the second study.

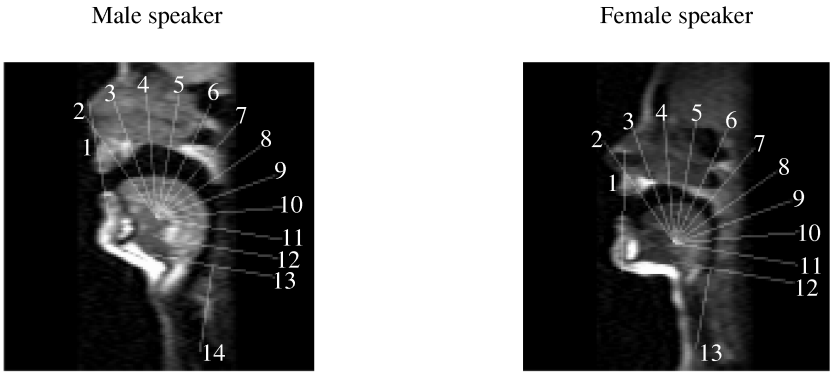



Figure 2.18: Pictures of a male and a female speaker from real time MRI with grids used in VT cross-dimension measurement.(Demolin et al., 2000).

Recently, Engwall (2004) studied tongue movements with the TSE Zoom sequence. He was able to acquire one mid-sagittal 6 mm thick slice of the VT at a rate of 9 Hz. The scanner and equipment were as in (Demolin et al., 2000), but the pixel matrix size was 128 × 128 and spatial resolution 2.34375 × 2.34375 mm.

Narayanan, Nayak, Lee, Sethy, and Byrd (2004) report a novel approach to imaging speech production with MRI. They used a spiral sampling strategy in gradient echo imaging. This means that instead of the normal planar acquisition of images they acquired images in interleaved spirals resulting in picture as seen in figure 2.19 and, more importantly, a image acquisition rate of 8-9 fps and a high reconstruction rate of 20-24 fps.

The experiments were carried out with a GE Signa 1.5 T scanner. The receiver coil was a general purpose head coil. Images were acquired in one mid-sagittal plane of 5 mm thickness. The resulting effective image resolution was 2.7 mm with a FOV of 30 cm. Since the study was first with this imaging sequence, Narayanan et al. (2004) expect to make improvements with image quality and resolution.

Figure 2.19: A picture from real time spiral echo MRI (Tamil retroflex [ɭ]) (Narayanan et al., 2004).

**Studies - Other Fast MRI Methods**

As MRI is still a developing method it is probable, that the temporal and spatial resolution of the scanners and the imaging sequences will continue to develop further. Furthermore, new imaging methods may well emerge. Two methods, which have done so, are stroboscopic MRI and tagged MRI.

Mathiak, Klose, Ackerman, Hertrich, Kincses, and Grod (2000) have performed stroboscopic MRI. The principle of stroboscopic MRI is to achieve a high time resolution by sampling several repetitions of the same utterance. This obviously requires measures to be taken to ensure that the subject repeats the utterance with very little variation between the repetitions and that, specially, the any movement artefacts are avoided.

The study concentrated on [gVŋ] material, where the vowel was one of [o, i, a, e]. Each word had to be repeated 170 times by each of the three subjects to facilitate reconstruction. The subjects synchronised their productions with the imaging sequence by listening to the MRI scanner noise.

The scanner used in the study was a 1.5 T Siemens scanner and the imaging sequence was Fast Low Angle Shot (FLASH). FOV was 200 $\times$ 200 mm$^2$ and pixel resolution 256 $\times$ 256. With a slice thickness of 8 mm the resulting voxels were 8 $\times$ 0.8 $\times$ 0.8 mm$^3$. Only one mid sagittal slice was recorded during each of the repetitions.

The final temporal resolution of the averaged sequence was 40 Hz, when the whole mid-sagittal view of the VT was reconstructed. But when only

a section through some important features, such as the uvula, was reconstructed the temporal resolution reached 120 Hz.

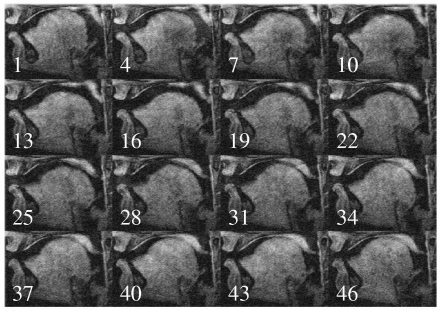



Figure 2.20: Pictures from stroboscopic MRI during the production of [gaŋ] (Mathiak et al., 2000). Each of the pictures is a result of averaging over 3 successive frames.

Tagged MRI is another method, which relies on averaging over several repetitions. It originates from cardiology. Unlike regular MRI, which excites the imaged tissues evenly, tagged MRI excites the tissues in layers called tags. As the excited areas then move and deform during movement of the tissues, a time varying, dynamic sequence can be reconstructed. If the same utterance is imaged several times, while the direction of tagging is varied the movement of flesh points inside for example the tongue can be followed. However, the tags do not last for very long and hence the utterance length is quite limited.

In an example study by Stone, Dick, Douglas, Davis, and Ozturk (2000), the tags decayed in 500-600 ms, and thus allowed imaging of quite short utterances. An example of the resulting images is shown in figure 2.21. The study consisted of two experiments: The first one aimed at acquiring data for a 2D tongue model and the second one for a 3D model.

In the first experiment the subject repeated one syllable 32 times. The imaging was performed in three sagittal planes, which each were 7 mm thick. The planes had a spatial resolution of 2.4 $\times$ 2.4 mm$^2$. 40 tagged points were tracked.

In the second experiment the subject repeated one syllable 18 times per slice, which were as follows: 5 sagittal slices repeated with horizontal and vertical tag planes and 10 axial slices with anterior to posterior tag planes.

The spatial resolution was 1.2 $\times$ 2.4 mm$^2$ and slice thickness 7mm. The resulting sequence had a frame rate of 24 fps (frames per second).

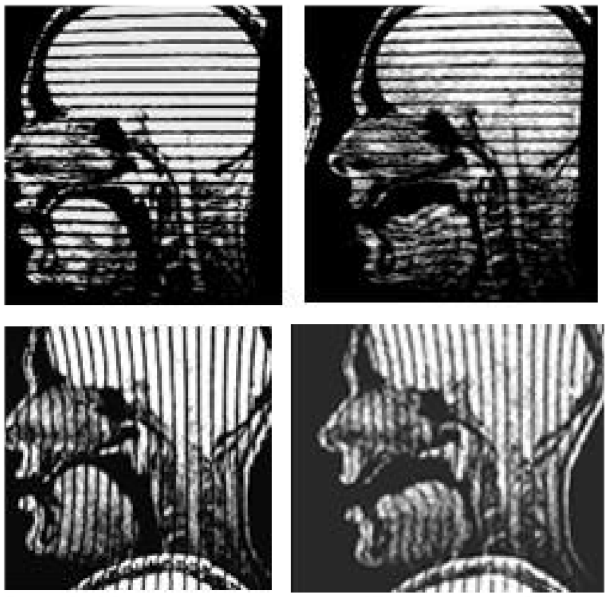


Figure 2.21: Mid-sagittal pictures from tagged MRI (Stone et al., 2000). The black lines are the tag planes, which have been initially before speech movements either horizontal (top) or vertical (bottom). Clear internal deformations of the tongue are visible in the images in the right side column.

### 2.2.7 Motion Capture

Motion capture methods are usually based on tracking markers. The markers are tracked with one or more cameras and so the resulting data describes the movement of the markers either in two or three spatial dimensions. Because the method does not utilise strong magnetic fields or ionising radiation, it is completely safe for the subject.

However, some restrictions apply to the quality of data obtained by motion capture methods. If data from only one camera is available, the resulting data is necessarily restricted to two dimensions. Furthermore, as the system uses either cameras operating in the visible range or in the infrared range, only visible articulation can be recorded. In marker based systems the inner lip boundary is a problem, since markers can not be placed there

due to obvious practical problems and problems with naturality of articulation.

On the other hand, motion capture produces data on potentially very natural articulation as the subject usually sits during recording. Further, simultaneous intraoral data acquisition is possible. An example can be seen in figure 2.23.

### Studies

In a study by (Honda, Kurita, Kakita, & Maeda, 1995) a general purpose marker tracking system was used. The left side of the outline of the lips was tagged with seven markers. These were tracked at a sample rate of 60 Hz with a gray-scale video system the signal of which was automatically precessed to retrieve the markers' centroid coordinates. The markers were hemispherical and had a diameter of 5 mm.

Beautemps et al. (2001) recorded concurrent cineradiography, labiofilm and sound data. The lips were painted blue and later extracted automatically. The extraction actually yielded the inner contour of the lips as it was based on the color information rather than the use of markers. The labiofilm was recorded with one camera at 50 fps (frames per second) with synchronous sound.

Badin et al. (2002) have recorded video images for motion capture purposes. Figure 2.22 shows examples of the recorded images and the resultant 3D mesh. The system is reported to be cheap to implement in comparison with commercial motion capture systems.

As seen in the picture the subject's face was marked with a grid and with 32 plastic beads. Further, his lips were painted blue to facilitate extraction of lip shape. A side view was provided by a mirror. Thus both the frontal and side view were recorded with the same camera.

3D coordinates of the 32 flesh markers were extracted by camera perspective models, which were calibrated by measuring a know object. In contrast, the lip shape (as defined by 30 points) was adjusted manually.

Beskow, Engwall, and Granstrom (2003) (see also (Engwall & Beskow, 2003b), (Engwall & Beskow, 2003a)) recorded concurrent EMA (Movetrack)

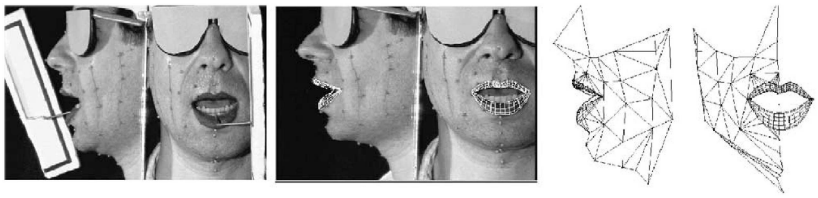


Figure 2.22: Motion capture with lips painted blue. First image on the left shows a video image of the subject with a jaw splint, the middle on shows the lip mesh superposed on the image and the final pair on the right shows the complete mesh as extracted in 3D from the video images (Badin et al., 2002).

and motion capture data (Qualisys) with commercial systems. The measurement setup is shown in figure 2.23.

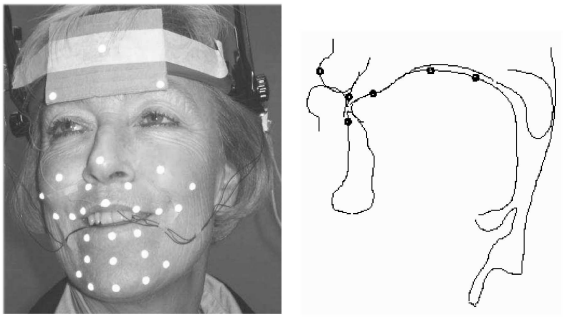


Figure 2.23: Simultaneous EMA and motion capture. Motion capture marker positions on the left and EMA coil positions on the right (Beskow et al., 2003).

The recorded material was spoken by one Swedish speaking female subject in sets of one minute. The material consisted of 270 sentences, 138 [VCV] & [$VC_1C_2\{C_3\}V$] words with symmetric vowels and 41 [$C_1VC_2$] words.

The motion capture system used 28 reflective markers (diameter 4 mm) and four infrared cameras. The markers were positioned on the face and the Movetrack unit. Furthermore, the lip coil was fitted with one of the markers. The Movetrack markers and the lip marker were used to align the data sets. The system extracted the markers' 3D coordinates at a rate of 60 fps.

In a study comparing slow, normal, rapid and hyperarticulated speech Maeda and Toda (2003) recorded motion capture data with a Vicon Motion Capture device. The system used 6 infrared cameras to track 3D coordinates of over 60 markers. The data was recorded at 120 fps.

The recordings were done with one American male and two French female subjects. Each subject spoke [VCV] nonsense material and a short text (in English or French according to nationality). The subjects were instructed to hyperarticulate the nonsense material and repeat the text with slow, normal and rapid rate of speech.

### 2.2.8 Electromyography (EMG)

Electromyography measures electrical potential caused by the contraction of muscles in response to nerve stimulation. Some examples of EMG data patterns in relation to the corresponding speech envelope can be seen in figure 2.24.

Electromyography is performed either with surface electrodes or with needle electrodes. The needle electrodes are actually thin needles, which are inserted into the monitored muscle. They are quite uncomfortable. In contrast, the surface electrodes are often similar to ones used in electrocardiography and cause only mild discomfort at most.

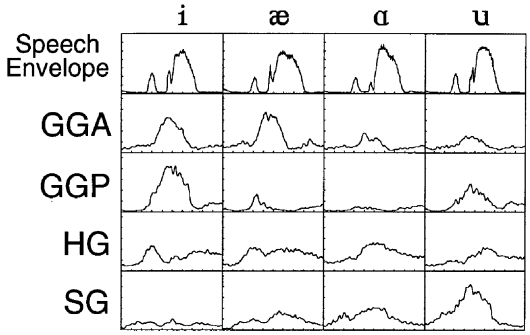



Figure 2.24: Examples of EMG data with a corresponding speech envelope. GGA is anterior part of genioglossus , GGP is posterior part of genioglossus, HG is hyoglossus and SG is styloglossus (Honda, 1996).

However, there is a problem with using surface electrodes. As they are not very selective the recorded signal may contain interference from other

muscles. On the other hand, needle electrodes are very selective even down to the level of a few muscle fibres. In other words, if complex muscle groups are involved it may be necessary to use needle electrodes.

No matter which electrode type is used, recording deep inside the VT or from intrinsic muscles is very difficult. The restriction on depth of the recording point inside the VT is caused by the same factors as the same restriction on EMA. As for the problem with intrinsic muscles, since these muscles are covered by other muscles surface electrodes are clearly out of the question, but the possibility of using very long needles is neither a good solution.

Despite the inherent difficulties with EMG, interesting data on motor control involved in speech production can be gained from its use. Indeed, EMG provides very useful data on the physiological mechanisms of speech production.

Honda et al. (1995) recorded EMG data from muscles, which control lip movement. Recorded speech material consisted of [ebVbe] and [ewVwe] sequences where the vowel was one of [i, a, u].

The recording targeted the following lip closing muscles: the orbicularis oris superior and inferior, the levator anguli oris, the depressor anguli oris, and the mentalis and the following lip opening muscles: depressor labii inferioris, and the levator labii superioris.

The data was recorded with two different types of surface electrodes. The first type was a disc electrode with a diameter between 4 and 6 mm. These were made from electrocardiography electrode material. The second type was a miniature cup electrode with Ag-AgCl discs of 2 mm diameter surrounded by electrode paste.

Hirayama, Vatikiotis-Bateson, and Kawato (1993) recorded EMG data from eight orofacial and extrinsic tongue muscles controlling jaw and lip closing and opening and tongue lowering and raising. The recording was done with surface and hooked-wire electrodes. The recorded material included real English speech and nonsense sequences like [asisuseso] where the consonants [s, t, p] changed place with each other. The recording was done at a sample rate of 2000 Hz, which was rectified and integrated at 100 Hz to enable its use in a neural network model of articulation.

## 2.3 Aeroacoustic Measurements

The term aeroacoustic measurements is used here to refer to measuring air flow and acoustic effects of air flow within the VT. The acoustic effects of airflow are very minimal in vowel and nasal sound production - if we consider only the VT and exclude the glottal and subglottal systems. The situation is quite different during the production of sounds, which rely on sound being produced in the supraglottal system. The most prominent example of this class of sounds are the fricative sounds, which are produced with a noise source provided by turbulent flow through a tight constriction within the VT. Detailed aeroacoustic studies are necessary to validate and develop theory on the mechanisms involved.

Airflow can be measured directly both with living subjects and with mechanical models. It would obviously be preferable to make all of the measurements within the VTs of real humans, but just as obviously this can not be done in most cases. Consequently, mechanical models are needed to enable measurements, which would otherwise be either completely impossible without compromising naturalness of articulation or without compromising the subject's safety.

Indeed, it is hard to design a setup to measure flow in a fricative constriction, without interfering with the articulation. In this light it is not very surprising that in vivo measurements are usually carried out during vowel production and more constricted articulations are measured with mechanical models.

### 2.3.1 Airflow Measurements with Living Subjects

Teager (1980) analysed the characteristics of airflow in the mouth cavity during phonation. His instrumentation included three anemometers (hot wire flow censors) and a microphone. All of these were mounted as an array. Two of the anemometers were mounted parallel to each other above the third one. This array was then moved across the mouth during a one second sustained phonation of [i].

The results were briefly as follows:

1. The sound measurement was more out of phase with the flow measurements (both upper and lower) then is possible to explain by the construction of the censor array (see figure 2.25).

2. The airflow measurements do not agree with laminar flow. Rather, they are consistent with separated flows. This means, that instead of a unified volume flow, there is a different kind of a flow present in the oral cavity.

3. This seems to mean that sound is actively generated in the mouth cavity during the described experiment.

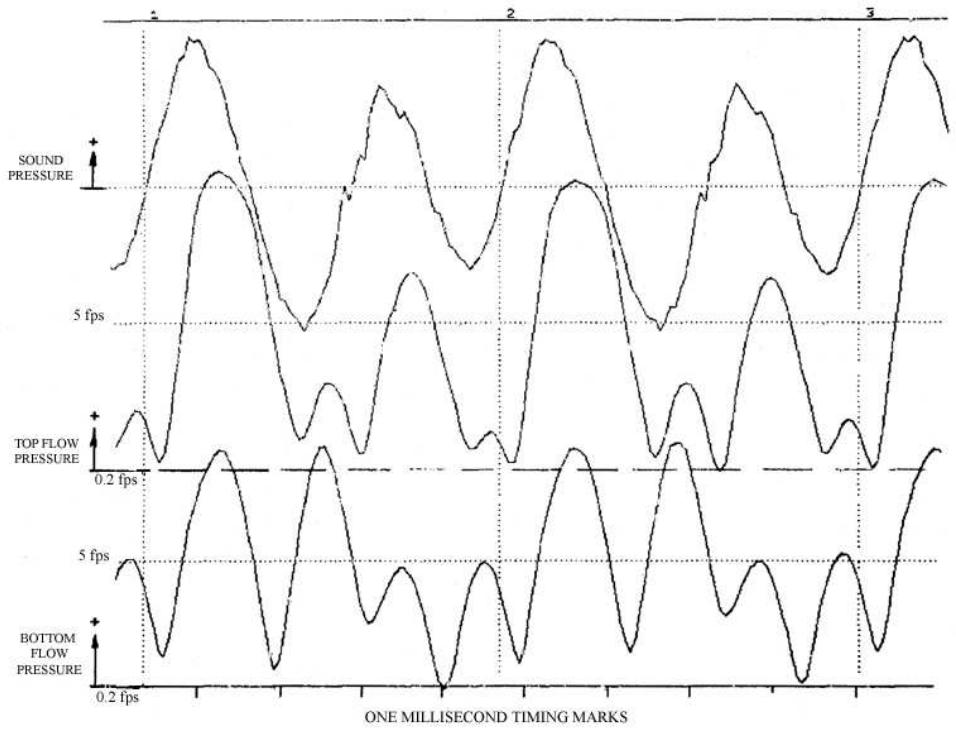



Figure 2.25: Flow measurements in the mouth cavity (Teager, 1980).

Anemometers record the local flow velocity very accurately and the direction of the measured flow can be easily controlled (Teager, 1980). There is, however, one problem with the use of anemometers in speech production studies. When recording with a live subject an anemometer can not be used at positions deep within the VT because of the risk of causing injury to the subject with the hot wire.

### 2.3.2 Mechanical Models

Mechanical modeling of articulation is probably the oldest method of speech synthesis. Over time methods have evolved and currently circuit models and computational models are more popular. However, in the context of airflow estimation, mechanical models have recently seen an increase in popularity. As everything can not be measured with a real speaker the construction of a more or less flexible mechanical speaker is often the best alternative.

#### Studies

Barney, Shadle, and Davies (1999) constructed a Dynamic Mechanical Model (DMM) of the glottis and the VT. A schematic of the model is shown in figure 2.26. The DMM consisted of a straight plastic tube of approximately the same dimensions as a human male VT. At an appropriate distance from the exit corresponding to the mouth the tube had a pair of electromechanically driven shutters to act as the vocal folds.

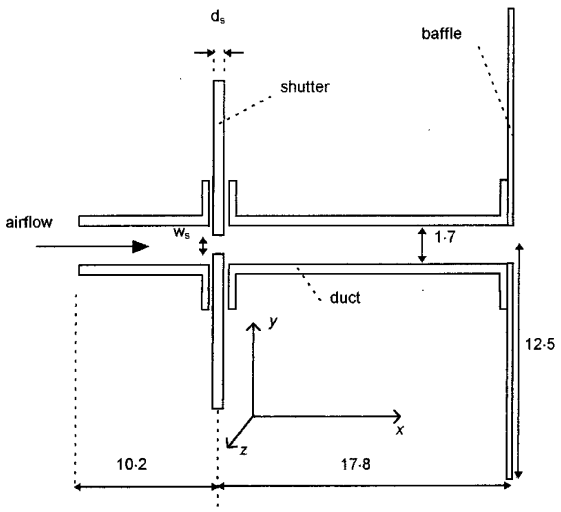



Figure 2.26: Dynamic Mechanical Model of the VT (Barney et al., 1999). The dimensions are given in centimetres. (The image is not to scale).

Barney et al. (1999) measured the flow field within the DMM with a with a constant temperature hot wire anemometer inserted through the artificial tract's wall. In the measurements they found, that the flow immediately

downstream of the shutters was very irregular. However, by the time the flow was close to the artificial tract's exit it had assumed an almost uniform cyclic character. This behaviour was considered to be caused by shear layers forming at the shutter exit with the development of more regular and even vortices taking place as the flow passes down stream along the tract. In spite of the fluctuating and, at certain points, irregular nature of the flow, a mean flow was observed at all measured positions even when the shutters were closed.

Barney et al. (1999) found that the radiated far-field sound pressure could be explained only in part by the traditional assumption that the VT acts as a passive filter for the sound generated at the glottis during phonation. They had better fit with a model, which included also sound generated by non-acoustic velocity fluctuation (i.e. vorticity) in the VT.

In the second part of the article Shadle, Barney, and Davies (1999) describe hot wire anemometer measurements performed in the upper VTs of four male subjects. The subjects and the recorded speech sound - [ʌ] - were chosen to correspond closely with the DMM. In addition, the same hot wire sensors were used as in the DMM measurements.

The hot wire signal was measured with the sensor placed 1 cm behind the front teeth in the middle of the VT, while the subjects phonated the vowel [ʌ] for 2 seconds. The signal was sampled at 5 kHz and low-passed at 2 kHz.

Static flow magnitude and dynamic variation in humans were found to be similar to the corresponding values in the DMM. Still, the values of the human subjects were a lot more varied. As the DMM was found to contain conducting vortices during simulated phonation, the similarity of measurement results was taken as evidence or similar phenomena in the human VT. Differences in the results were accounted for mainly with several unideal characteristics of the DMM.

In his Ph.D. thesis Sinder (1999) used a mechanical model, which is shown in figure 2.27, to validate his fricative production model (see section 4.3, page 103 for more details). The measurement facility consisted of a straight metal tube, which had cross dimensions of the same order as the human VT. It had a flow source at one end, feeding a jet flow into the tube. An obstruction of Gaussian form was placed within the tube at a distance of

5cm ($x_c$ in the figure) from the flow source. The flow velocity and average or time varying pressures could be measured at different points within the tube by inserting a hot wire anemometer or a pressure transducer through the wall.

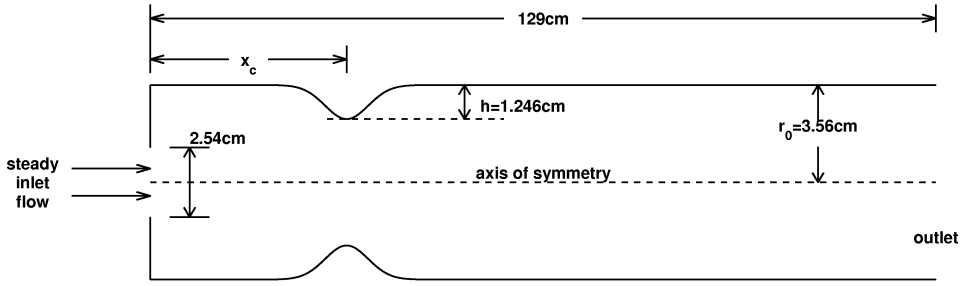
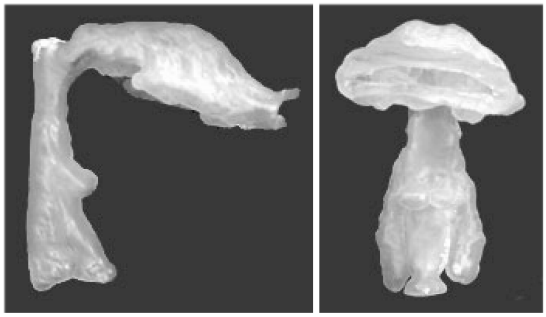



Figure 2.27: A diagram of the flow measurement facility used by Sinder (1999).

An interesting approach to VT modeling has been taken by Kitamura, Fujita, Honda, and Nishimoto (2004), who have made resin reconstructions of the VT based on MRI data. An example is shown in figure 2.28. The reconstructions are going to be used to measure VT transfer functions for vowels.



(a) lateral view (b) front view

Figure 2.28: A resin reconstruction of the VT based on MRI data (Kitamura et al., 2004).

## 2.4 Summary

Several choices of data acquisition methods or combinations thereof can be argued for quite convincingly. Besides the data collection properties of the

methods factors such as availability and cost of equipment have to be taken into account. In any case, one of the chosen methods should be a dynamic one. Furthermore, if the goal is not 2D but 3D synthesis another one should probably be a static 3D method.

The choice of measurement methods is naturally interconnected with implementation issues of geometric synthesis. These are the topic of the next chapter.

# Chapter 3

# Models of Vocal Tract Geometry

Geometrical models of the VT can be seen as a sum of three parts: The base geometry, movement parameters and movement generation mechanism. The first part is obviously based on static data and, as obviously, the third part has to based on dynamic data. In contrast, the case for movement parameters is not clear cut. They may be derived based on either static articulations or on dynamic movement data.

In constructing a model of VT geometry, each of these parts can be varied fairly independently of the others. For instance, a choice of either two or three dimensional base geometry does not exclude the use of concatenation or coarticulatory modeling to generate movement.

## 3.1 Modeling Vocal Tract's Base Geometry

VT's base geometry can be modelled in two or three spatial dimensions or in a way that effectively falls between them. The first and last cases have been handled together below, while the second case is under its own heading.

All of these models are usually separated into functional or physiological parts. For example, a very common division is to consider tongue, velum,

jaw, and lips as separate articulators with little or no interaction. Such separation has the benefits of making the model easier to manage from an implementation point of view as well as rendering its functioning easier to interpret in terms of theories of speech production.

### 3.1.1 Two Dimensional Models

A two dimensional description of VT geometry is required for traditional source filter methods of speech synthesis with an articulatory model (see chapter 4). This 2D description is in terms of distance from glottis $x$ and the cross-sectional area $A$.

The simplest 2D models of the VT model the cross sectional area $A$ directly as a function of the distance from glottis $x$. That is, they are of the form $A(x)$. A somewhat more complex class of models, model the area as a function of the mid-sagittal cross-sectional distance $d$ in a piece wise manner. This means, that they define a different area function of the form $A(d)$ for different regions of the VT.

Furthermore, two parts of the VT are usually modelled separately from the rest. These parts are the lips and the nasal tract. The separate modeling of lips follows from the fact that the region of the VT in question changes not only its area but also its length. In contrast, the nasal tract is considered a separate part in many models, since it does not change its characteristics significantly over time, but rather only its coupling with the main tract.

#### The $\alpha\beta$-model

The most used way to define the relation between the mid-sagittal dimension of the VT and its cross-sectional area first appears in a paper by Heinz and Stevens (1964). It has become to be known as the $\alpha\beta$-model after its parameters. It relates the sagittal cross-sectional distance $d$ to the cross-sectional area $A$ simply as follows:

$$A = \alpha d^{\beta} \tag{3.1}$$

Here $\alpha$ and $\beta$ are parameters, which depend on the position in the VT. Examples of the use of this model can be found in the next section.

#### Models

Lindblom and Sundberg (1971) used a version of the $\alpha\beta$-model with parameters shown in table 3.1. The model was reported to be inaccurate for modeling pharyngeal VT area if the cross-sectional distance $d$ exceeds 20 mm, but otherwise achieved reasonable accuracy.

The values for $\alpha$ and $\beta$ used by Lindblom and Sundberg (1971) are listed in table 3.1. There are no values for lips because they were modeled separately with a different type of model (see section 3.2.2, page 66).

Table 3.1: Values of $\alpha$ and $\beta$ used by Lindblom and Sundberg (1971)

| Region | $\alpha$ | $\beta$ |
|---|---|---|
| Mouth | 2.2 | 1.38 |
| Upper pharynx | 0.68 | 1.9 |
| Lower pharynx | 1.1 | 2.21 |

A complex implementation of the $\alpha\beta$-model was used by (Mermelstein, 1973). Mermelstein's definitions for $A(d)$ are shown in table 3.2. The pharyngeal region follows Heinz and Stevens (1964) while the oral region is an approximation of data by Ladefoged et al. (1971).

### 3.1.2 Three Dimensional Models

Modeling the VT as an essentially 2D entity has several problems, which can be solved by constructing the model as three rather than two dimensional. $\alpha\beta$-type models suffer from a rather complex relationship between the cross-sectional distance and area and, also, from a need to redefine this relationship for practically each speaker. In contrast, a 3D model is more demanding computationally, but gives a very natural way for relating articulatory movement to changes in the cross-sectional area. Furthermore, it a 3D model is easier to utilise in visual speech synthesis and makes modeling asymmetric articulation easier.

Like a 2D model a 3D model usually divided into submodels. These can be, for example tongue, lips, teeth, nasal tract, larynx and the rest of the VT as one static structure.

Table 3.2: Definitions of $A(d)$ used by Mermelstein (1973)

| Region | Formula | Note |
|---|---|---|
| Pharynx | elliptical | one axis $d$ and the other increasing from 1.5 (glottis end) to 3 cm |
| Soft-palate | $2d^{1.5}$ | |
| Hard-palate | $1.6d^{1.5}$ | |
| Alveolar ridge to incisors | $\begin{cases} 1.5d, \\ 0.75 + 3(d - 0.5), \\ 5.25 + 5(d - 2), \end{cases}$ | when $d < 0.5$<br>when $0.5 < d < 2$<br>when $d > 2$ |
| Lips | elliptical | $d$ is the other axis while the other $= 2 + 1.5(s_l - p_l)$, where $p_l$ is lip protrusion and $s_l$ vertical lip separation |

Several different idealisations can be employed to reduce the model's complexity. For example, it can be constructed symmetrically in relation to the mid-sagittal plane. Further, attached structures can be simplified. In particular, the nasal cavity can be modelled as a simplified chamber instead of its quite complex structure. Also, piriform sinuses are often either ignored completely or added to the VT's cross-sectional area at one or more points instead of modeling them as small separate branches of the tract.

Even though these models are fully three dimensional volumetric models, they are usually converted in essence to 2D models when used in sound synthesis. After the conversion represents the 3D model by an area function, which simplifies the sound generation process.

### Models

Engwall (1999b) (also (Engwall, 1999a)) defined a 3D model of the VT, which can be seen in figure 3.1. The model is sagittally symmetric. The model has a static nasal tract, which is modelled simply as one cavity. It also includes lips, tongue and teeth in the oral region.

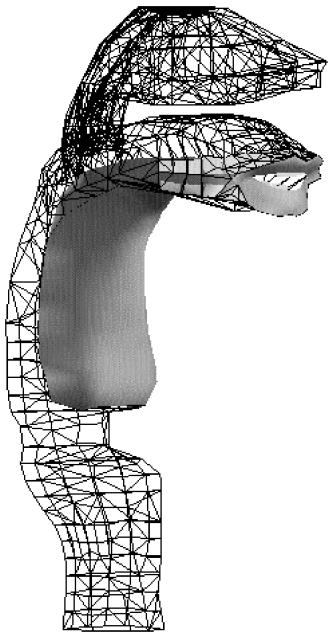


Figure 3.1: A 3D vocal tract by Engwall (1999b). The tongue, teeth and lips are shown in grayscale and the rest of the vocal and nasal tracts are shown in wireframe.

Badin et al. (1998) employed a construction method later used also by Engwall and Badin (1999). Half of a resulting VT is shown in figure 3.2. In both cases the construction of the model geometry was based on slices of the VT obtained with MRI.

After VT contour extraction Badin et al. (1998) sampled the contours in each slice at 51 points. The sample points were organised consistently so that they could be connected with fibres, which range the whole length of the VT connecting sample points with the same index in adjacent slices. Especially, fibres connecting the points 1 and 51 correspond to one side of the mid-sagittal contour while the fibre connecting point 26 in each slice corresponds to the other side.

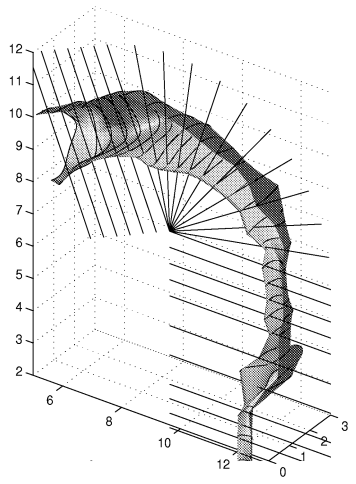



Figure 3.2: A 3D vocal tract by Badin et al. (1998). The model has been cut in half along the mid-sagittal fibre. The semipolar grid in the picture is the grid formed by the original samples.

## 3.2 Parametrisation of Vocal Tract Movement

Parametrisation of movement is performed in order to reduce complexity of measurement data and - even more importantly - to gain understanding of its structure. In other words, to understand the process of speech production.

If sufficient measurement data is available, finding a parametrisation could be forgone by simply using the raw data as a basis for a concatenation library. This, however, is usually not desirable because of the prohibitively large library size. Hence, parametrisation is performed in practically every study.

The parameters can be defined either based on heuristic decisions, statistical analysis of data or directly on the physiology of the speech production organs. All of these approaches have in common the use of one or more data sources, while at the same time using phonetic common sense to keep the model useful and realistic.

### 3.2.1 Physiological Models

In a conceptual sense, the most straightforward way to define the parametrisation of speech movements is to use muscles as the basis. However, it is also a very demanding way to define a model.

At least within the framework of the model it adds a new level with its own parameters: the muscles themselves. This is caused by the need to control and model the workings of a single muscle, which is not as simple a task as it might seem. Another aspect of the complexity is the need for physical modeling, which generates a heavy computational demand. Still, these are important models, since through them we can gain knowledge that might not be as readily available from different types of modeling.

Most of the physiological models discussed in this text have a fairly strong (or even practically inseparable) connection between movement parameters and movement generation. Therefore, examples of physiological models are given later in section 3.3.2, page 80.

### 3.2.2 Heuristically Defined Models

The term heuristic model is used here to refer to models, which are primarily based on phonetic theory and human anatomy. In particular the parameters of these models are based on theoretical knowledge rather than data driven methods such as statistical analysis.

Nevertheless, these models are usually based on measurement data. The connection comes in the form of activation levels of the parameters, which are most commonly defined by comparison and/or fitting with measurement data. This approach lessens the possibility of performing unrealistic synthesis, while keeping the model construction relatively simple and straightforward.

### Models

Based on X-ray tracings of vowels Stevens and House (1955) defined a model of the VT with three parameters: $r_0$, $d_0$ and $A/l$[1]. The parameters are illustrated in figure 3.3.

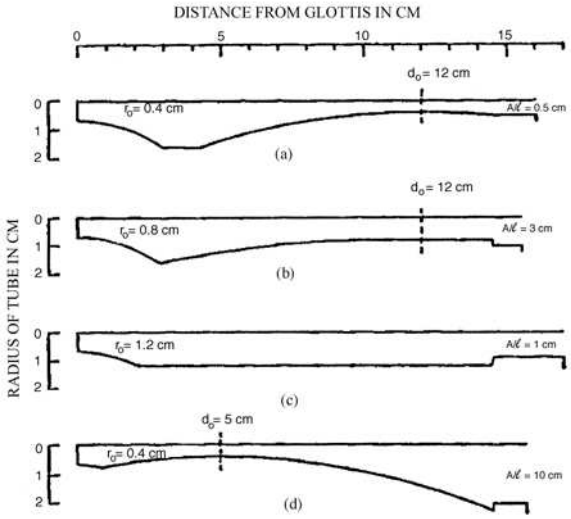



Figure 3.3: Idealised two dimensional vocal tract (Stevens & House, 1955) displaying the effects of the parameters $r_0$, $d_0$ and $A/l$.

$r_0$ is the VT's radius at the point of constriction, $d_0$ is the distance from glottis to the constriction and $A/l$ is a lip protrusion parameter defined by the ratio between mouth area and length between teeth and lips. In this model the VT's radius at other points is defined by equation 3.2.

$$r - r_0 = 0.25(1.2 - r_0)x^2, \tag{3.2}$$

where $x$ is the distance to the maximum constriction.

A model known as the APEX model was first developed by Lindblom and Sundberg (1971) and a more recent version is described by Stark, Lindblom, and Sundberg (1996) and Stark et al. (1999).

[1]Please note, that the parameter $d_0$ nor other parameters of the models discussed in this section are related to the $d$ or other parameters otherwise used in this text.

The original model was based on X-rays and photographs of a speaker of Swedish. It had several parameters for tongue, mandible and lip movement. The tongue was controlled by body position $d$ and body shape $c$. Both of the tongue parameters were defined in relation to the mandible, which moved along a predetermined trajectory according to one parameter. The effect of these parameters is illustrated in figure 3.4.

Finally the lip contour was defined by equation 3.3 and the area of the lip aperture by equation 3.4.

$$y = \pm\frac{h}{2}\sqrt{1 - \frac{2}{w}^{p} |x|^{p}}, \tag{3.3}$$

where $h$ is the inner height and $w$ the inner width of the lip opening, while $p$ is a speaker dependent curvature parameter and $x$ is the horizontal position.

$$A_{lips} = hw\frac{p}{p+1} \tag{3.4}$$

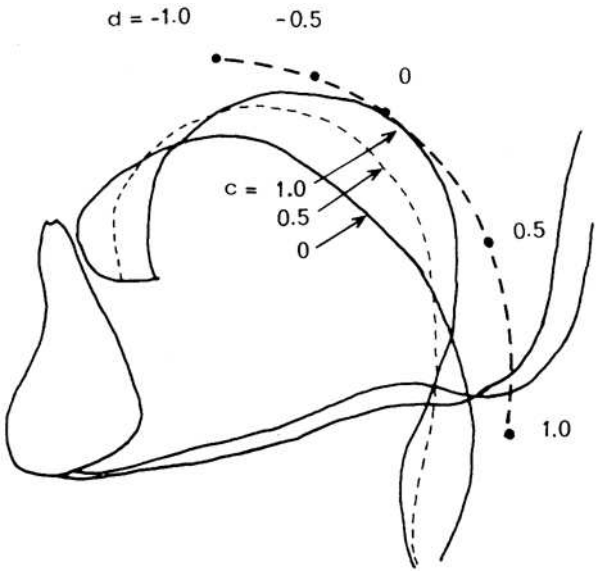



Figure 3.4: The tongue from two dimensional vocal tract model (Lindblom & Sundberg, 1971). The figure displays the effects of the parameters $d$ and $c$ - tongue body position and shape, respectively.

A more recent version of the same model is described by Stark et al. (1996) and Stark et al. (1999). The goal of the system is to study apical sounds (and hence the name). Like the original model it uses the $\alpha\beta$-model for cross-sectional distance to area conversion.

The model has eight synthesis parameters controlling the submodels. Lips are included only as an area model and epiglottis and larynx are modeled by translating and rotating static contours. Mandible (teeth and mouth floor) is modelled as a static contour, which is positioned according to the mandible parameter.

In contrast, the tongue model includes two more detailed submodels. Tongue apex is modeled with a parabolic curve between tongue body and tip and controlled with protrusion and curvature. As the apex-body fit is kept smooth by rotation the curvature parameter has the effect of lifting the tongue tip.

The tongue body is controlled with two parameters called position and deviation. Position is the back-front position and deviation is the degree of the constriction. The actual tongue hump is produced with a modified (non-symmetric) Gaussian function.

Mermelstein (1973) developed a VT model based on X-ray data from Perkell (1969). The a schematic representing the model is shown in figure 3.5. The model's parameters are also marked in the figure.

The jaw (angle $\theta_j$ and distance $s_j$) and the hyoid bone move in a fixed coordinate system. The lips and tongue-body move relative to the jaw. The tongue tip moves in relation to the tongue body and velum opening is controlled in a fixed coordinate system.

Coker (1976) define an articulatory model, which has two separate speeds for tongue movement (see figure 3.6). This definition is based on data from Houde (1967). Hence, there are two parameters for slow jaw and tongue body movements along with one parameter for fast tongue body movements. These correspond respectively to movement relating to vowels and consonants.

In addition to the above, there are two parameters for raising and curling back the tongue tip. Also, the model has a purely acoustical parameter

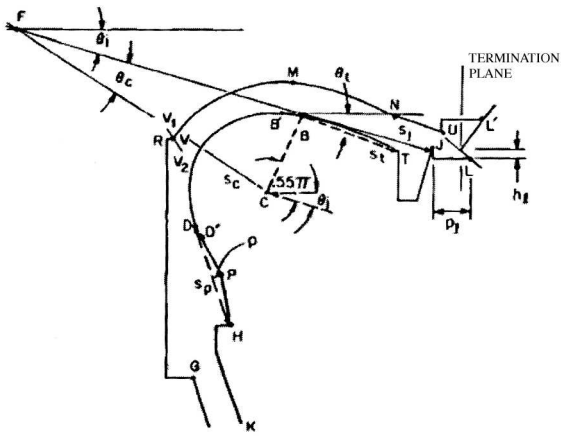



Figure 3.5: A two dimensional vocal tract (Mermelstein, 1973) displaying the effects of the parameters. (See text for explanations.)

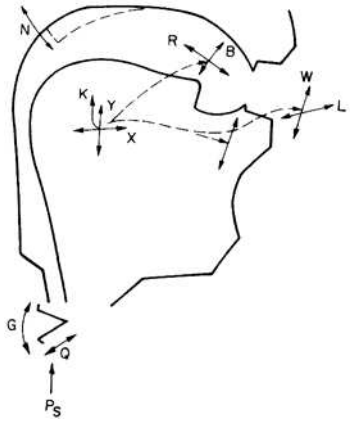


Figure 3.6: A two dimensional vocal tract (Coker, 1976) displaying the effects of the parameters. (See text for explanations.)

for controlling the velar opening and one for controlling constriction in upper pharynx. Finally, one parameter is used to limit the smallest possible according cross-sectional area of the VT according to phonetic context.

A model based on the above Mermelstein model is presented by Rubin, Baer, and Mermelstein (1981) and Rubin et al. (1996). The latter version is part of a synthesiser called CASY (Configurable Articulatory SYnthesiser). Its parameters are shown in figure 3.7 and explained in table 3.3.

An example of a practically non-geometrical articulatory synthesiser can be found in Stevens and Hanson (2003). Their synthesiser maps articula-

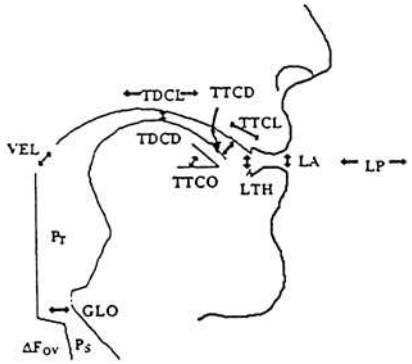



Figure 3.7: A two dimensional vocal tract (Rubin et al., 1996) displaying the effects of the parameters. (See text for explanations.)

Table 3.3: Articulatory parameters of CASY (Rubin et al., 1996).

| | |
|---|---|
| LP | lip protrusion |
| LA | lip aperture |
| TDCL | tongue dorsum constrict location |
| TDCD | tongue dorsum constrict degree |
| LTH | lower tooth height |
| TTCL | tongue tip constrict location |
| TTCD | tongue tip constrict degree |
| TTCO | tongue tip constrict orientation |
| VEL | velic aperture |
| GLO | glottal aperture |
| Ps | subglottal pressure |
| Pt | transglottal pressure |
| ΔF0v | change in virtual fundamental frequency |

tory control parameters directly to the control parameters of a format synthesiser without any geometrical representation at all. However, the parameters are physiologically motivated. They are shown in table 3.4 and their relation to VT physiology is illustrated by figure 3.8.

### 3.2.3 Statistically Defined Models

Liljencrants (1971) analysed mid-sagittal VT shapes with Fourier analysis. While Fourier representation is not very natural for VT, Liljencrants'

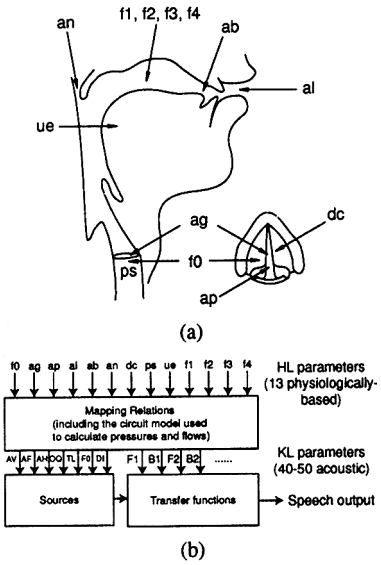



Figure 3.8: a) The relation of HLsyn parameters to VT physiology and b) the procedure of mapping the pseudo-articulatory parameters to the formant synthesis parameters (Stevens & Hanson, 2003). See table 3.4 for an explanation of the parameters.

Table 3.4: Articulatory parameters of HLsyn (Stevens & Hanson, 2003).

| | |
|---|---|
| f1-f4 | Formant frequencies F1-F4 |
| f0 | Fundamental frequency F0 |
| ag | Average area of glottal opening between the membranous portion of the vocal folds |
| ap | Area of the posterior glottal opening |
| ps | Subglottal pressure |
| al | Cross-sectional area of the constriction at the lips |
| ab | Cross-sectional area of tongue blade constriction |
| an | Cross-sectional area of velopharyngeal port |
| ue | Rate of increase of VT volume |
| dc | Change in vocal-fold or wall compliances |

work does prove an important point: Two to three parameters or components give an accurate description of the mid-sagittal movements of the VT. Hence, such results from other - more suitable - methods are very natural.

In a certain sense statistical models can be considered the other side of

the coin when compared with heuristic models. This is because statistical analysis can be considered most interesting when its results agree closely with the theory of phonetics. This means that the analysis should produce parameters, which correspond to traditional classification systems of articulation and/or speech sounds. When the chosen method does not produce suitable variables, it has to be forced to do so.

The greatest benefit of using statistical analysis to define the parameters of a VT model is its direct basis on data. Even if the parameter generation is forced, it is quite unlikely to produce unnatural results. However, at the same time the basis in data means, that often quite lengthy processes of data acquisition, preparation and analysis are needed.

#### Parallel Factorial Analysis (PARAFAC)

PARAFAC is described in Harsman (1970). It is a variant of factorial analysis, which does not require the factors or parameters to be uncorrelated. With this concession comes the benefit of having a unique solution for the problem of fitting the model to a given data set.

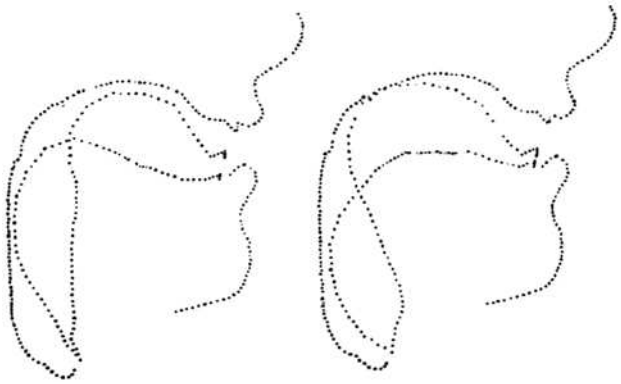


Figure 3.9: The effect of parameters $t_1$ (front-raising) and $t_2$ (back-raising) extracted from X-ray data with PARAFAC (Ladefoged et al., 1978).

Ladefoged et al. (1978) used three way factor analysis (also known as three way component analysis) to obtain components of tongue movement in the sagittal plain. The procedure is described in in (Harshman et al., 1977) (see also section 2.2.1, page 31). The effect of varying the resulting parameters can be seen in figure 3.9. They correspond - without forced analysis - to front-raising and back-raising of the tongue.

While PARAFAC is usually quite successful with mid-sagittal data, this is not always the case as reported by Zheng and Hasegawa-Johnson (2003). In addition they report, that when applied to 3D data PARAFAC is able to extract parameters, which correspond to the parameters, that it extracts from the mid-sagittal portion of the same data. In contrast, lateral effects are not represented so well by the results of PARAFAC analysis.

### Factor Analysis [2]

Factor analysis is a method for dimensional reduction of data which are considered to follow a multidimensional normal distribution. In matrix form the factor model can be written as:

$$x = \mu + Af + u, \tag{3.5}$$

where $x$ is an original observation vector, $\mu$ the expected value vector of the observed phenomenon, $A$ the loading matrix, and $f$ and $u$ two independent random variables.

Factor analysis can be applied as a kind of forced version, where certain interesting components are first extracted from the data by linear regression analysis. After this the remaining variance is analysed with normal factor analysis methods. This type of analysis is called arbitrary factor analysis (here AFA) (Maeda, 1990).

Maeda has used AFA to analyse VT shapes in two studies (Maeda, 1979) and (Maeda, 1990). The first study was based on data from one female speaker and concentrated on analysing only the tongue shape.

The second study used data from two speakers - one male and one female. The tongue parameters extracted for one of the speakers of the study are shown in figure 3.10. The study included also similar analysis of the lips, larynx and velum as independent parts of the resulting model. This

[2]Please note, that the division to factor analysis and principal component analysis reflects the terminology used by the original authors. From a statistician's point of view the terms would be almost interchangeable here. This comes from the fact, that the essential difference is whether the analysed data is considered to be multinormally distributed or not.

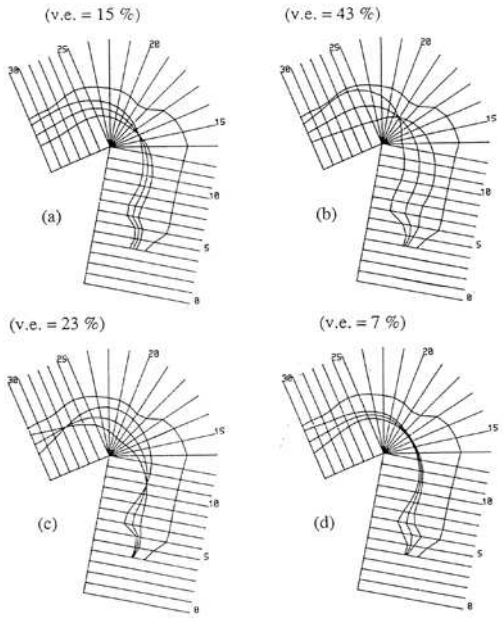



Figure 3.10: The effects and explained variances of four parameters extracted from X-ray data with AFA by Maeda (1990).

separate analysis and parametrisation was based on the assumption, that physiologically separate articulators should be separated also in the model.

### Principal Component Analysis (PCA)

PCA is a method for dimensional reduction of data. Unlike factor analysis the data need not be considered to follow a multidimensional normal distribution. However, like factor analysis, principal component analysis is commonly applied in either a guided or a forced form when used on articulatory data. In the forced form it is often called linear component analysis (LCA).

Two dimensional tongue parameters resulting from PCA were used by Kaburagi and Honda (1996). While the analysis of the tongue movement itself was unforced, the effect of jaw movement was first removed from the EMA data. Resulting parameters can be seen in figure 3.11. They explained 97% of the variance.

Badin et al. (1998) used guided PCA to analyse MRI data. The method involves iterative application of PCA to the remaining variance. At each step

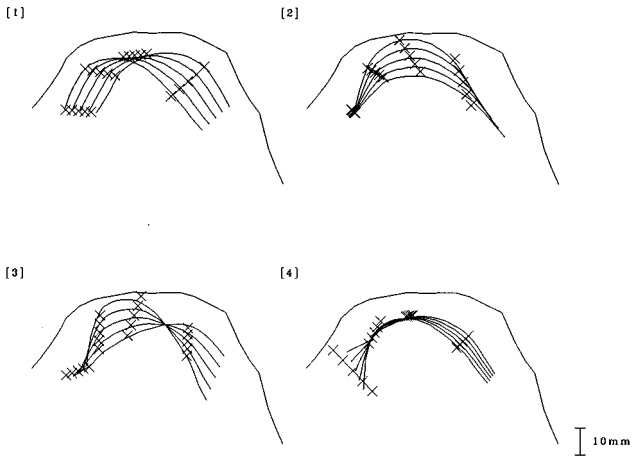



Figure 3.11: Tongue parameters derived with PCA from EMA data (Kaburagi & Honda, 1996). The crosses represent the movement of the EMA measurement coils and the solid lines the effect of the principal components.

one or two principal components are chosen based and subtracted from the data. The choice is based on clear interpretability of the components rather than on the amount of explained variance. The authors report, that this procedure does not result in the greatest possible explanation of variance, but does often yield components, which can be easily interpreted.

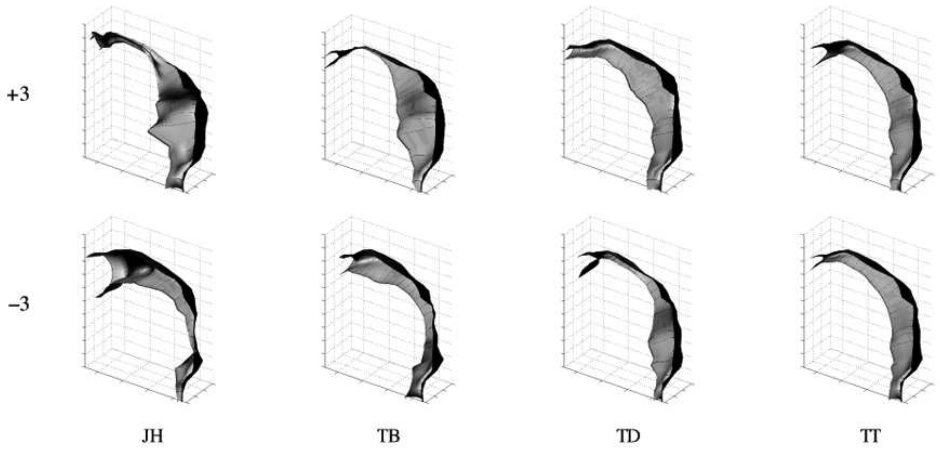



Figure 3.12: Nomograms for 3D VT parameters by Badin et al. (1998). The parameters shown here were given names as follows: JH = jaw height, TB = tongue body, TD = tongue dorsum, and TT = tongue tip.

Badin et al. (1998) extracted five components, which could be given an interpretation and four more, which could not. Four of the first five are illustrated in figure 3.12. The remaining named component, which is not shown in 3.12 was TA = tongue advance. The named components explained 75 % of the variance, and when four more components were added with classical (unguided) PCA, the whole model explained 94 % of the variance.

Engwall (2000a) constructed a 3D tongue model, which was based on an earlier mid-sagittal (2D) VT model (Engwall & Badin, 1999). While the 2D model was defined by guided PCA (see above the study by Badin and also (Beautemps et al., 2001)), the 3D model was defined with LCA. The first five parameters of the 3D model were chosen to correspond to the parameters of the 2D model. A sixth parameter was then added based on the remaining variance. The effect of these parameters can be seen in figure 3.13.

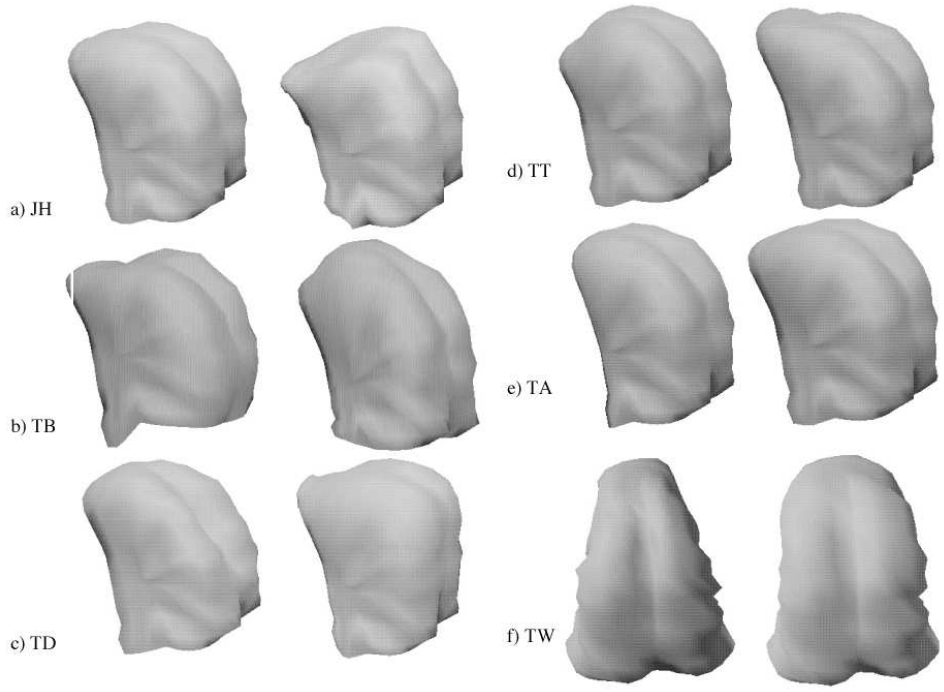



Figure 3.13: Nomograms for 3D tongue parameters by Engwall (2000a). The names of the parameters are: JH = jaw height, TB = tongue body, TD = tongue dorsum, TT = tongue tip, TA = tongue advance, and TW = tongue widening.

Four parameters of the original 2D mid-sagittal model achieved an ex-

plained variance of 90 %. In contrast, the first five components of the 3D model achieved on sagittal data an overall explained variance of 78 % and a mid-sagittal explanation ratio of 88 %.

Badin et al. (2002) developed a 3D tongue model in a similar manner as the one by Engwall above. However, its basis was a 2D tongue model, which was extracted from the mid-sagittal portion of the same data set with LCA. The resulting 3D model thus had the same five parameters as the 2D model. Three of these are illustrated in figure 3.14. The explained variances for the 3D model were 72 % of the total variance and 89 % of the mid-sagittal variance.

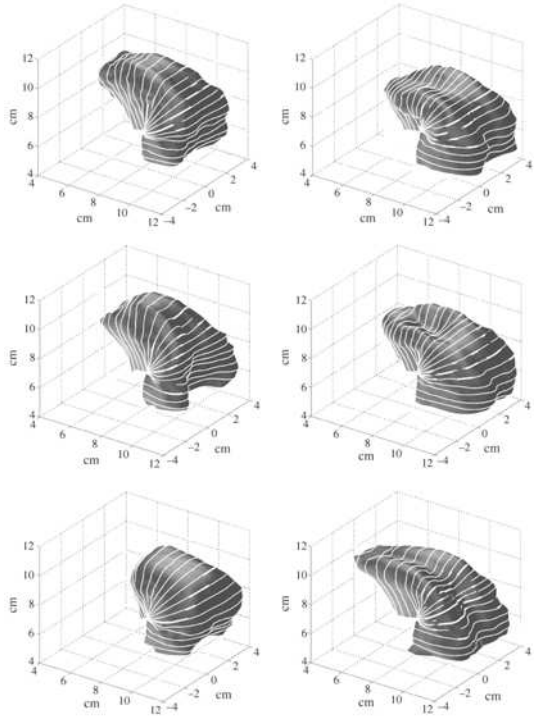


Figure 3.14: Nomograms for three of the 3D tongue parameters by Badin et al. (2002). The parameters shown here were named JH = jaw height, TB = tongue body, TD = tongue dorsum.

## 3.3 Movement Generation

When a suitable parametrisation of VT movement and a phonetic description of the speech material to be synthesised are available, movement can

be generated. This can be achieved with either heuristic rules, concatenation, muscle modeling, coarticulatory modeling, gestural modeling or in some other way.

### 3.3.1 Heuristic Movement Models

In comparison with defining movement parametrisation, defining a movement generation mechanism can be quite hard, if there is no relevant data available. This is due to the fairly complex way, that our articulatory organs move. However, if some sort of data is available, a heuristic approach is probably one of the easiest to implement as one gets to decide the types of functions used et cetera without any restrictions. Thus, heuristic models of movement generation rely on observation based rules to produce the movement trajectories.

#### Models

Mermelstein (1973) defined a dynamic articulatory model. It synthesised [VCV] -sequences by generating movement trajectories for the static articulatory model discussed in the same article.

The dynamic model was based on the following principles: Vowel movement is slow and precise while stop release involves a fast articulator movement. Stationary vowels require four parameters (out of nine defined in the mid-sagittal articulatory model). These were two tongue body position, one jaw, and one lip position parameters. In addition to these consonants need also tongue tip, lip height, and velar opening parameters. Consonant synthesis poses restraints only on certain parameters and instead of defining absolute values defines only parameter space regions which should be reached. In addition, some parameters are not restrained by consonants at all.

The varying effect of consonants on different parameters was achieved by defining a pertinence rank for the parameters in different types of consonant productions. For example tongue body parameters were given a rank of 1 for velars (condition has to be met), a rank of 2 for alveolars (parameter has to be within a given range) and a rank of 3 for labials (no restrictions).

The model also contained detailed rules for articulator activation. These were based on observations on [həCV] material. In particular, timing events were fixed at the values observed in the data. In contrast, the amplitude of activation and F0 contour were determined in an ad hoc manner.

### 3.3.2 Physiological Models

Generating movement with a physiological model can be quite complex, if the model mirrors the physiology very closely. This fact becomes evident, when one notes, that for such a model the parameters should be artificial neural signals, which cause artificial muscles to contract. The generation of the neural signals for a given utterance is made harder by the problems faced in recording such signals (see section 2.2.8, page 52 for more details).

The task of synthesis can be simplified a bit by resynthesising an utterance from EMG data and thus skipping the need to generate the control signals. Even so, the model will not be a simple one as muscle physics are very complex and not yet anywhere close to being a routine task.

As mentioned previously (see section 3.2.1, page 66), the models are described below not only in relation to movement generation, but also in relation to movement parametrisation.

#### Models

Honda (1996) has developed a 2D physiological model of the VT. The model includes the tongue, jaw, hyoid bone, laryngeal cartilages and, the vocal folds. These organs are moved in the model by 21 active muscles. In the case of rigid bodies the physics are modeled with mass-spring systems while the tongue's deformations are calculated with a finite element model (FEM).

In the model the motor organisation of the tongue is modeled as three layers: First, the peripheral mechanics or muscle physics, second, motor command generation and, third, sensorimotor integration. The model's musculature for controlling tongue movement consists of four extrinsic muscles. These are organised into two opposing muscle pairs, which are genioglossus posterior vs. hyoglossus and styloglossus vs. genioglossus anterior. These pairs and their effect on the tongue are shown in figure 3.15

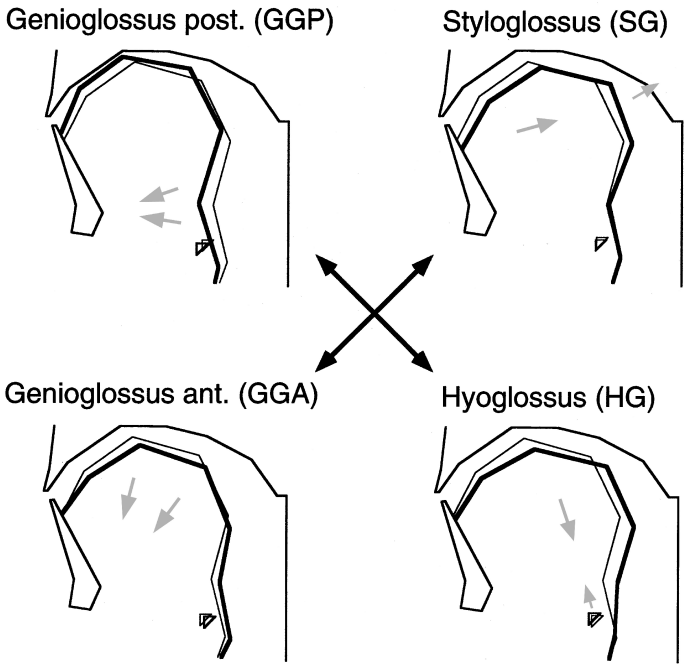



Figure 3.15: Tongue model based on extrinsic muscles (Honda, 1996). The effects of two opposing muscle pairs are illustrated.

Honda (1996) found the muscle pair model to be consistent with the statistical parameters of Maeda (1990). Additionally, the two pairs formed an orthogonal space, which enabled synthesis of tongue movements. Honda (1996) used an ensemble of average EMG waveforms as input for the model. As seen in figure 3.16, these were plotted as a vector sum, which was then used as the activating force.

Payan and Perrier (1997) have created a physiological tongue model. Like the above Honda's model it uses FEM for modeling tongue deformation, but unlike Honda's work it implements the equilibrium point hypothesis (Feldman, 1966). The hypothesis states, that the central nervous system controls a threshold muscle length. Furthermore, motorneurons integrate the proprioceptive feedback and the current threshold length to a possible activation of muscle contraction. To produce continuous movement the central nervous system sets a discrete sequence of threshold length values, which are interpolated linearly between the start and end values.

Payan and Perrier (1997) implemented the muscles seen in figure 3.17. Muscle activation was modeled as force effects on mesh nodes correspond-

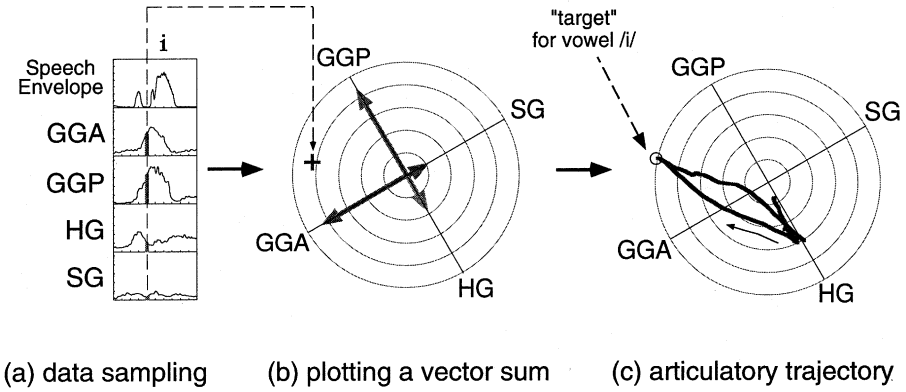



Figure 3.16: Generation of articulatory trajectories for the tongue from EMG data (Honda, 1996). (a) The averaged EMG data from [əpVp] utterances. (b) The muscle activations are summed to find the resultant force. (c) The trajectory is formed by a sequence of summed data samples.

ing to the activated muscle. A further effect was the change of elastic properties of the elements representing the activated muscle. The muscular force was given an exponential relation to muscular activation. On the other hand, the activation was given simply as the amount by which the muscle's length exceeds the threshold muscle length. Finally, the actual force exerted on the tissue was computed as a (somewhat complex) function of the muscular force and the first time derivative of the muscle's length.

Payan and Perrier (1997) tested their model by synthesising vowel to vowel sequences. They found the results encouraging as the synthesised movements matched EMA data fairly closely. Nevertheless, they also reported that better estimates were needed for tongue mass and viscosity as well as the capability of muscles to generate force and damping of muscle activation signals caused by proprioceptive feedback.

### 3.3.3 Concatenative Models

Synthesis by concatenation in general is performed by selecting the most suitable speech sample from a library of such samples. During concatenation synthesis of VT geometry suitable movement samples are selected and

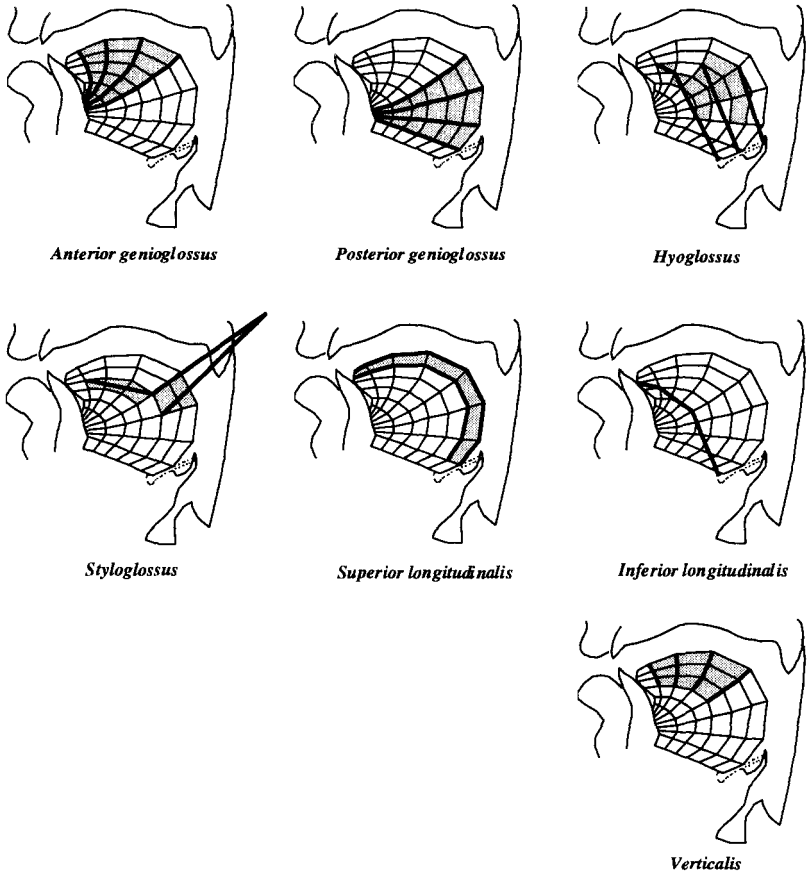



Figure 3.17: The bold lines represent orientation and points affected by muscle forces in a FEM model by (Payan & Perrier, 1997). Further, the gray areas represent elements, which change their elastic properties when the muscle is activated.

possibly filtered to form the desired articulatory movements. This is analogous with concatenative sound synthesis, where sound samples are treated in much the same way. Another analogy is the most commonly used sample unit, which for sound synthesis is the diphone and for articulatory synthesis the diart. Both correspond to a segment from the stationary part of one phoneme or articulation to the stationary part of the next phoneme or articulation.

Segmenting the data poses some problems as the stationary part of a phoneme or articulation is not always very clearly defined. However, the most difficult tasks in this type of synthesis are the construction of the sam-

ple library and the selection of the most suitable sample to be used in a given context.

The samples may be either raw unprocessed data or, for example, parametrised data. The latter approach gives the benefit of reducing the library size, which can easily be very large.

### Models

Greenwood (1997) describes the process of building a articulatory synthesiser utilising concatenation of diarts. The samples were generated from MRI data, which was simplified to provide only the area functions along with the corresponding formants as the aim was spectral copy synthesis.

Rather than use a single monolithic codebook or library Greenwood (1997) generated several separate ones. Each of these corresponded to a voiced diphone. Also, the method of finding the best sample from the library was enhanced by using a two component cost function. It took into account not only the formants, but also the changes in area functions from frame to frame.

To reduce the complexity of the data, Greenwood (1997) parametrised the MRI data. Two different parametrisations were used and compared with each other. One was a descendant of Mermelstein's model (Mermelstein, 1973) and the other was quasi articulatory model (Greenwood & Goodyear, 1994). As the synthesis results were compared little difference was found between the two systems.

Engwall (2002) presents a concatenative synthesis system capable of generating a visual representation of articulation. The system concatenates diart samples of articulatory parameter transitions. The diart library was based on data from the MOCHA database (Wrench & Hardcastle, 2000), which was parametrised by fitting it to a 3D tongue model (Engwall, 2000a). After parametrisation the data was segmented into diarts according to the segmentation used in the MOCHA database.

Timing and phoneme information was produced with Festival (Black & Taylor, 1997). Selecting diarts from the library was performed by minimising the cost function:

$$C = \sum_{j=1}^{5} \sum_{i=1}^{N} \Big( \big(P_j^s(i) - P_j^e(i-1)\big)^2 + w_d \big(P'^s_j(i) - P'^e_j(i-1)\big)^2 \Big) + w_t \sum_{i=1}^{N} \Delta t_i,$$

where $P_j$ are the articulatory parameters, $P'_j$ their derivatives, $P_j^s(i)$ the start value of diart $i$, $P_j^e(i-1)$ the end value of diart $i-1$. Further, $N$ is the number of diarts in the utterance, $\Delta t_i$ the difference between the duration of diart $i$ and a target duration, and finally $w_d$ and $w_t$ are weights used to tune the cost function.

The results were evaluated against the original data from the MOCHA database as well as X-ray films of other speakers (from the database described by Munhall et al. (1995)). The model achieved a good fit in general, but had problems with underrepresented units. In addition, the movements of tongue tip and root were found too restricted.

### 3.3.4 Coarticulatory Models

Coarticulation means the effect that the phonemic context (preceding and following phones) has on the articulation of a given phone. The possible model types reported in the literature before the early 90's range from look-ahead type of models (Öhman, 1967) and time invariant models (Bell-Berti & Harris, 1982) to hybrid models combining the two (Al-Bamerni & Bladon, 1982). The first type lets the beginning of a movement toward a certain articulatory position vary according to the phonemic context while the second type assumes a fixed onset time in relation to the achievement of the desired position.

Coarticulatory models have the advantage of providing an interface for testing theories of dynamic speech production. Indeed, the fact alone, that a separate algorithmic module for producing articulatory trajectories exists, makes it possible to try different strategies in the production. On the other hand, dynamic speech is far from a simple phenomenon. Accordingly, models of coarticulation tend to reflect this. In other words, they may be quite intricate.

### Models

Cohen and Massaro (1993) defined a model of coarticulation for their visual speech synthesis system. They used dominance (weighing) and blending functions in their model. The dominance functions were of the form:

$$D(t) = \alpha e^{-\theta_{\leftarrow}|t-t_0|^c}, \text{when } t - t_0 \geq 0 \tag{3.6}$$

$$D(t) = \alpha e^{-\theta_{\rightarrow}|t-t_0|^c}, \text{when } t - t_0 < 0 \tag{3.7}$$

Here $D(t)$ is the either the anticipatory (3.6) or following (3.7) dominance as defined for a given parameter in a certain segment. Furthermore, the activation amplitude $\alpha$ and the rate parameters $\theta_{\leftarrow}$ and $\theta_{\rightarrow}$ are also defined individually for each of the articulatory parameters and segments. Finally, $t - t_0$ is the temporal distance from the peak of dominance and $c$ a form parameter.

The different target values $T_{sp}$ for each segment $s$ and parameter $p$ were combined by using a weighted sum at each point in time $t$:

$$F_p(t) = \frac{\sum_{s=1}^{N}(D_{sp}(t) \times T_{sp})}{\sum_{s=1}^{N} D_{sp}(t)} \tag{3.8}$$

An example of how the model was applied to visual synthesis is shown in figure 3.18. The model was fitted to the individual phones manually.

Le Goff (1997) (also (Le Goff & Benoît, 1997)) extended the above definition of $D(t)$ to so as to make it a $C^n$ function (a function with $n$ continuous derivatives). This extended version is given below as equation 3.9 and examples are shown in figure 3.19.

$$D(t) = \alpha e^{-\theta_i|t-t_0|} \times \sum_{j=0}^{n-1} \frac{\theta_i^j}{j!} \times |t - t_0|^j \tag{3.9}$$

As can be seen the form parameter $c$ is not present. The sum and the last term provide the necessary smoothing for the n-continuity at the point where the rate term $\theta_i$ changes.

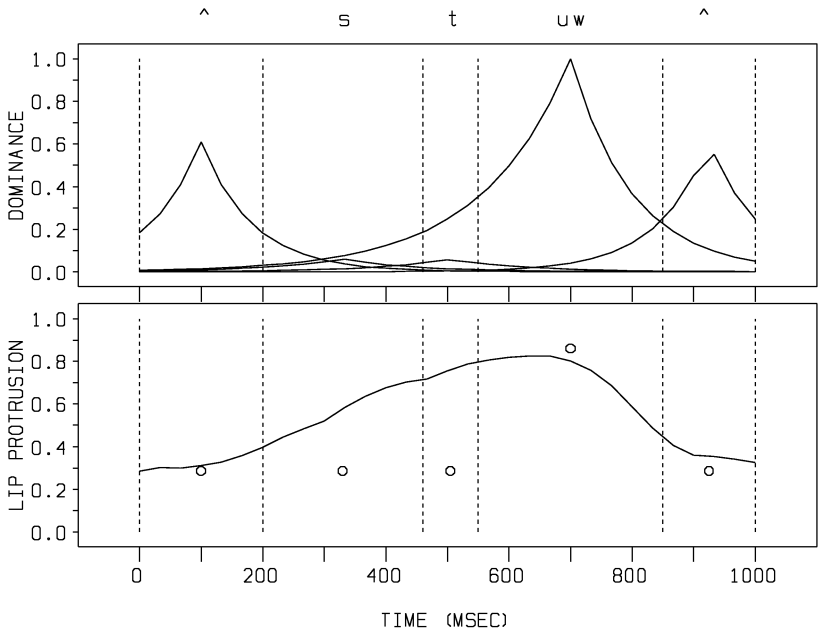



Figure 3.18: The change of an articulatory parameter value, when a trajectory is with exponential dominance functions for the word "stew". (Cohen & Massaro, 1993).

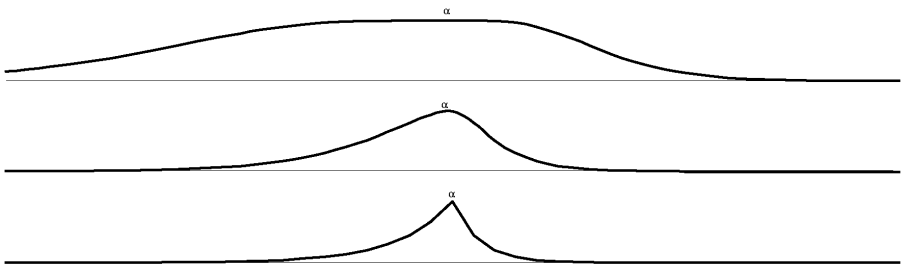



Figure 3.19: Examples of spline-like dominance functions with different continuity orders (from bottom to top $C^0$, $C^1$ and $C^3$) and asymmetrical definitions ($\theta_1 < \theta_2$) (Le Goff, 1997).

Le Goff (1997) fitted the dominance functions automatically to articulatory data. This was done by calculating the euclidean distance between observed and synthesised trajectories. The order of continuity for each articulatory parameter was chosen by trial-and-error while relaxation method was applied to the dominance function parameters.

### 3.3.5 Gestural Models

Gestural models are based on the concept of articulatory phonology (Browman & Goldstein, 1992), which states that the basic phonological unit is the articulatory gesture. This in turn is defined as a specific invariant goal such as closure of the lip aperture. However, the exact method of producing the goal is not defined by the gesture, but is rather seen as context dependent. Thus, the effect remains the same while the method of production varies.

#### Models

Kaburagi and Honda (1996) has defined an articulatory model, which is based on similar principles as the one by the developers of articulatory phonology (Rubin et al., 1996). In this model each gesture poses certain constraints on some or all of the tract variables. The basic functioning principle of the constraints is shown in figure 3.20.

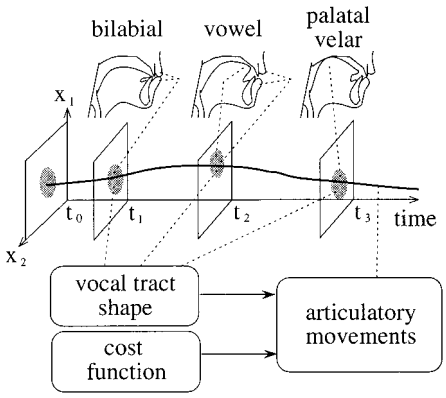



Figure 3.20: Basic functioning of movement constraints in Kaburagi and Honda (1996). Trajectories are derived by satisfying the constraints posed by articulatory gestures.

The model includes three layers of variables, which are illustrated in figure 3.21. First, the state variables $x_i$, which are defined in relation to jaw position. Second, the absolute positions $y_i$ in the mid-sagittal plane. Third, the tract variables $z_i$, which define relative tract measures such as lip opening height. The state and tract variables correspond respectively to articulator activation levels and the invariant measures that define a gesture.

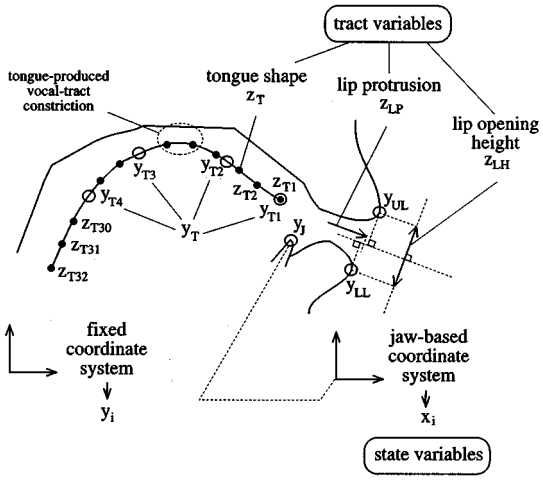



Figure 3.21: Parameters of the model by Kaburagi and Honda (1996). See text for explanation.

Not all of the tract variables are constrained by each gesture and, furthermore, the constraints can be posed as an area in the tract variable space. This makes it necessary to somehow further constrain the trajectory generation in order to make the solution unique. Kaburagi and Honda (1996) achieve this by requiring, that the solution minimises a cost function $C$:

$$C = \sum_{n=1}^{N-1} \big(C_x(n) + C_f(n)\big) + C_x(N), \text{ where} \tag{3.10}$$

$$C_x(n) = \big(\overline{x}(n) - \overline{x}(n-1)\big)^t W_x \big(\overline{x}(n) - \overline{x}(n-1)\big), \text{ and} \tag{3.11}$$

$$C_f(n) = \big(\overline{f}(n) - \overline{f}(n-1)\big)^t W_f \big(\overline{f}(n) - \overline{f}(n-1)\big). \tag{3.12}$$

In the equations above $\overline{x}(n)$ and $\overline{f}(n)$ are vector of the state variables and a vector of input forces at step $n$. The forces were used to drive second order linear systems, which produce the movement (or change) in the state variables. Additionally, $C_x$ and $C_f$ are the weighted changes and $W_x$ and $W_f$ the corresponding weight matrices for the state variables and input forces.

Kaburagi and Honda (1996) evaluated the systems performance against the same EMA data they used in defining the tongue parameters (see section 3.2.3, page 75). They report the mean error to have been 0.84 mm over 35 utterances. The material consisted of vowels and the consonants [ktp].

## 3.4 Handling Collisions

As a model's articulators move during speech it becomes necessary to ensure, that they do not reach physically impossible configurations. In other words, collisions have to be handled in some way.

Regardless of the dimensionality of the model, two different approaches present themselves readily: First, the collisions can be prevented by ensuring that impossible configurations are just that - impossible. The movement parameters can be defined relative to other parts of VT physiology. For example "move the tongue tip 95% of the way towards the upper incisors". Second, the collisions can be corrected after they happen. This means, that after each step of movement generation collisions are detected and corrected.

The first approach has the desirable property of high computational efficiency at run time. However, the second approach can be said to generate more realistic results. This is because the second approach provides a possibility to determine the force used to press the tongue against the palate. This in turn enables natural deforming of the tongue in such a situations.

### Models

Cohen et al. (1998) used the measure: $\overline{n} \cdot \overline{P}$ to detect collisions between the palate and the tongue. Here $\overline{n}$ is the palate's surface normal and $\overline{P}$ a vector from a tongue point to a palate surface point. Naturally, the collisions can be detected by examining the sign of the product.

As a way of correcting a detected collision, Cohen et al. (1998) considered using a parallel projection to move the offending tongue points back to the palate's surface. Unfortunately, this strategy can result in undesirable point distributions as the convex palate surface will produce clustering. Besides, the method is also fairly slow.

Among other enhancements of the correction process Cohen et al. (1998) divided the space around the palate and teeth into voxels as illustrated in figure 3.22. These were assigned a status according to whether a tongue

point occupying a given voxel was *ok*, *not ok* or *uncertain*. For the definite cases further checking could be forgone, while for the uncertain cases checking had to be performed along with possible corrections.

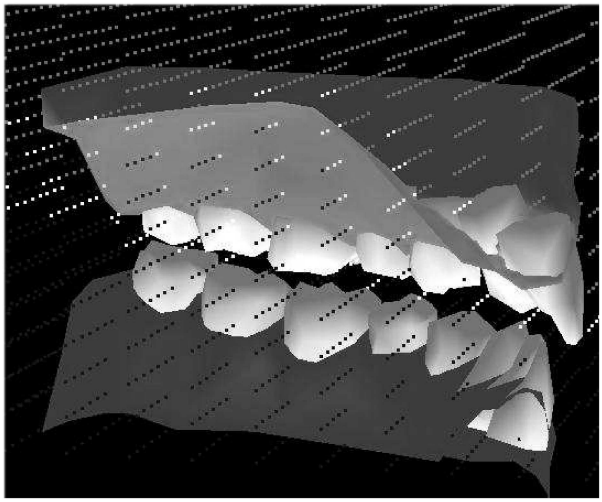


Figure 3.22: Voxel space used by Cohen et al. (1998) in collision detection. The dots' colors correspond to voxels which are *ok* (dark), *not ok* (gray) and *uncertain* (white) (see text for explanation).

(Engwall, 2001a) used the same collision detection principle as Cohen et al. (1998) above. However, he made enhancements to the detection and correction procedure as a whole.

First, the grid spanning the palate and teeth was interpolated and sampled to cover the $xy$ plane evenly as seen in figure 3.23. Thus, the search for closest palate point could be omitted altogether as a simple rounding of the tongue points coordinates gives the relevant information.

Next, if a collision is detected, the correction is equally simple. The tongue point is merely moved to the closest palate vertex. As the interpolated grid was dense enough the error produced by this procedure was of the same order as the initial reconstruction error.

## 3.5 Summary

It would seem increasingly difficult to justify the use of a 2D model of VT geometry. However, it should be noted that, if studying sound signal gen-

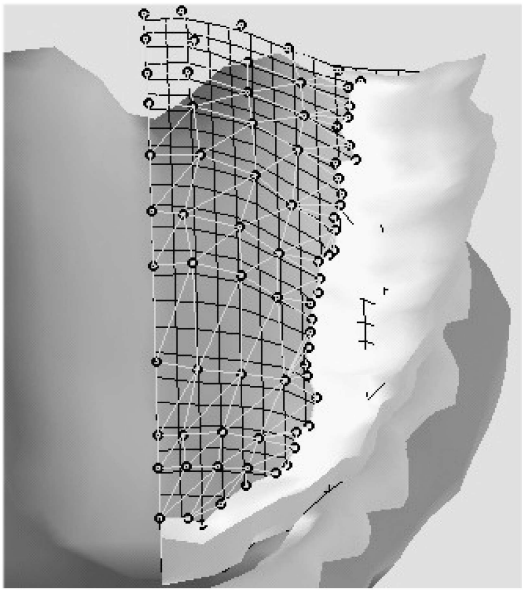


Figure 3.23: The more evenly spaced black mesh has been created by interpolating and sampling of the bright mesh. The black mesh makes collision detection simpler (Engwall, 2001a).

eration is not a central issue in the study, the dimensionality of the model may be chosen quite freely.

Accordingly, while the idea of using area functions for modeling VT geometry is by no means new, it is still of central importance. And as we shall see in the next chapter most of speech signal generation methods utilise area functions.

# Chapter 4

# Acoustic Synthesis

With the geometrical data available from one of the models described in the previous chapter, we get to the final stage in articulatory speech synthesis. Acoustic synthesis in the context of an articulatory speech synthesiser can be implemented in several different ways with very varied requirements and computational loads.

Stark et al. (1996) and (Stevens & Hanson, 2003) (see figure 3.8, page 72) do it by controlling a formant synthesiser. But as the focus of this thesis is on simulating the speech production process more interesting methods are: filter models (electrical circuit analogs as well as computational models) and models with actual turbulence modeling.

Depending on the geometrical description's level of detail, some methods of acoustic synthesis may not feasible. Additionally, not all of the methods described here are readily able to support synthesis of dynamic speech.

## 4.1 Mathematical Basis of Modeling Vocal Tract Acoustics

Before looking at examples of acoustic synthesis systems, it is necessary to understand some of the mathematics involved in their design. Therefore, we will derive some basic equations for sound pressure within the VT and also a more specialised version known as Webster's horn equation. Subsequently, we will review a traditional tube approximation used to solve the

relevant equations. Finally, the source filter model of speech production will be shortly reviewed.

### 4.1.1 Basic Equations

To describe wave propagation within the VT mathematically, we need the definitions given in table A.1 in appendix A. As a first step of our analysis we will simplify two basic equations by taking certain facts into consideration.

We will begin with a slightly modified form of the law of mass preservation (4.1) and Euler's equation (without mass sources) (4.2):

$$\frac{\partial p}{\partial t} + \rho_0 c^2 \nabla \overline{u} = 0 \tag{4.1}$$

$$\rho_0 \frac{\overline{u}}{\partial t} + \nabla p = 0. \tag{4.2}$$

Now, since the frequency range of interest (roughly from 100 to 3000 Hz) is fairly low in comparison with the frequencies of cross-mode resonances (lowest is around 10000 Hz for an average male VT), one can assume that the signal propagates as a plane wave. Therefore, the equations 4.1 and 4.2 can be reduced into only one dimension. By noting that particle velocity now becomes volume velocity divided by the area of the VT's cross-section ie. $u(x,y,z,t) = u(x,t) = \frac{U(x,t)}{A(x)}$, the equations become:

$$\frac{\partial p(x,t)}{\partial t} + \frac{\rho_0 c^2}{A(x,t)} \frac{\partial}{\partial x} U(x,t) = 0 \tag{4.3}$$

$$\frac{\rho_0}{A(x,t)} \frac{\partial}{\partial t} U(x,t) + \frac{\partial p(x,t)}{\partial x} = 0 \tag{4.4}$$

### 4.1.2 Webster's Horn Equation

Webster's horn equation is - as its name suggests - associated with wave propagation in tapered cones such as found in brass instruments. The VT geometry can be, at least in a piecewise manner, considered to consist of such cones.

By differentiating equation 4.3 with respect to time and substituting equation 4.4 into the former, the equations 4.3 and 4.4 can be combined - with certain assumptions - into a from known as Webster's equation:

$$\frac{\partial^2 p(x,t)}{\partial t^2} = c^2 \frac{1}{A(x,t)} \frac{\partial}{\partial x} \left[ A(x,t) \frac{\partial p(x,t)}{\partial x} \right] \tag{4.5}$$

A more detailed derivation of the above equation can be found in appendix A along with some discussion of the idealisations that are necessary in deriving the horn equation.

While the equation 4.5 has been the basis of speech synthesis systems and production studies, it has several drawbacks. The most obvious drawback lies in the way it was derived - linearisation removes non-linear phenomena. This means that it becomes necessary to provide a mechanism for producing turbulence outside of simulating the Webster equation. However, the greatest drawback is, that there is no analytical solution for 4.5, if $A$ is a function of both $x$ and $t$. However, it can be solved, if $A$ is a function of only $x$.

### 4.1.3 Properties of Acoustic Tubes

One way of avoiding the problems with solving the Webster equation 4.5, is to simplify the geometry of the VT. This can be done by replacing the original VT geometry with a tube which has a circular cross-section and a variable diameter. If we take a step further, and let the tubes diameter change at only specified points we get the model shown in figure 4.1 a).

Let us first consider the properties of a single tube section. For a given section the cross-sectional area $A$ is constant and hence, the equations 4.1 and 4.2 can be written in the following form:

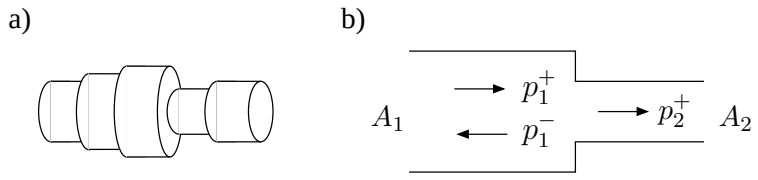



Figure 4.1: a) 1D discrete tube model of the VT. b) Reflection and transmission at a tube junction as a wave arrives from the left. $p_1^+$ is the arriving pressure wave, $p_1^-$ its reflection and $p_2^+$ the part transmitted into tube section 2. Cross-sectional areas of the tubes are $A_1$ and $A_2$.

$$\frac{\partial p}{\partial t} + \frac{\rho_0 c^2}{A}\frac{\partial}{\partial x}U = 0 \tag{4.6}$$

$$\frac{\rho_0}{A}\frac{\partial}{\partial t}U + \frac{\partial p}{\partial x} = 0 \tag{4.7}$$

These yield the one dimensional wave equation by the following procedure: First, we take the time derivative of equation 4.6, change the differentation order of $U$ in the resulting equation and solve equation 4.7 for $\partial U/\partial t$. Finally, we assign the time derivative of $U$ back to the first equation and simplify to get the wave equation for pressure 4.8. The wave equation's D'Alambert solution is given by equation 4.9. It consist of two wave of arbitrary form - one travelling to the left $p^-$ and one to the right $p^+$.

$$\frac{\partial^2 p}{\partial t^2} = c^2\frac{\partial^2 p}{\partial x^2} \tag{4.8}$$

$$p(x,t) = p^+ + p^- = p(x+ct) + p(x-ct) \tag{4.9}$$

Now, if we consider the junction in figure 4.1 b), we note the following continuity conditions: The pressure and the particle velocity have to be continuous across the junction. These conditions can be written as equations 4.10 and 4.11.

$$p_1 = p_2 \tag{4.10}$$

$$\overline{u}_1 = \overline{u}_2 \tag{4.11}$$

By further noting, that the pressures $p_i$ can be written with the help of the reflection and transmission coefficients $R$ and $T$, and that the particle velocity can be written as volume velocity divided by the area of the tube section $A_i$ we get:

$$\begin{cases} p_1^+ + p_1^- = p_2^+ \\ \overline{u}_1^+ - \overline{u}_1^- = \overline{u}_2^+ \end{cases} \Rightarrow \begin{cases} p_1^+ + Rp_1^+ = Tp_1^+ \\ (1-R)\dfrac{U_1}{A_1} = T\dfrac{U_1}{A_2} \end{cases} \Rightarrow \begin{cases} 1 + R = T \\ \dfrac{1-R}{A_1} = \dfrac{T}{A_2} \end{cases}$$

Solving the last form finally yields the the formulas for the reflection and transmission coefficients of the junction as given by equation 4.12.

$$\begin{cases} R = \dfrac{A_1 - A_2}{A_1 + A_2} \\ T = \dfrac{2A_2}{A_1 + A_2} \end{cases} \tag{4.12}$$

The next section 4.1.4 describes how the above result may be used in building a model of speech acoustics. However, before that we will take a brief look at how we might represent the acoustical properties of such a tube by electrical properties associated with transmission lines.

There are certain similarities between the equations, which the above analysis began with, and those involved in analysing transmission lines. These allow us to relate the acoustical quantities in tubes with electrical quantities in transmission lines.

Examples of the relations are shown in table 4.1. Noting these relations enables the use of circuit analysis methods developed for transmission line analysis in analysing the behavior of the VT.

Table 4.1: Correspondence of Acoustical and Electrical Parameters after Deller et al. (2000)

| **Acoustic Quantity** | | **Electrical Quantity** | |
|---|---|---|---|
| Sound pressure | $p(x,t)$ | $u(x,t)$ | Voltage |
| Total particle flow | $U(x,t) = Au(x,t)$ | $i(x,t)$ | Current |
| Characteristic impedance | $Z_0 = \rho c/A$ | $Z$ | Impedance |
| Acoustic inductance | $\rho/A$ | $L$ | Inductance |
| Acoustic capacitance | $A/\rho c^2$ | $C$ | Capacitance |

### 4.1.4 Source Filter Model of Speech Production

Building a model by concatenating several tube sections and solving the resulting system's transfer function leads us to equation 4.13. In it $P_r$ is the speech signal's frequency domain representation. It is obtained by multiplying the frequency domain representations of the sound source (glottal and/or frication), the VT and the radiation impedance present at the mouth opening. These are, in the same order, $S$, $T$ and $R$. This model is commonly known as the source filter model of speech production (Fant, 1970). Another way of looking at the model is shown in figure 4.2.

$$P_r(j\omega) = S(j\omega)T(j\omega)R(j\omega) \tag{4.13}$$

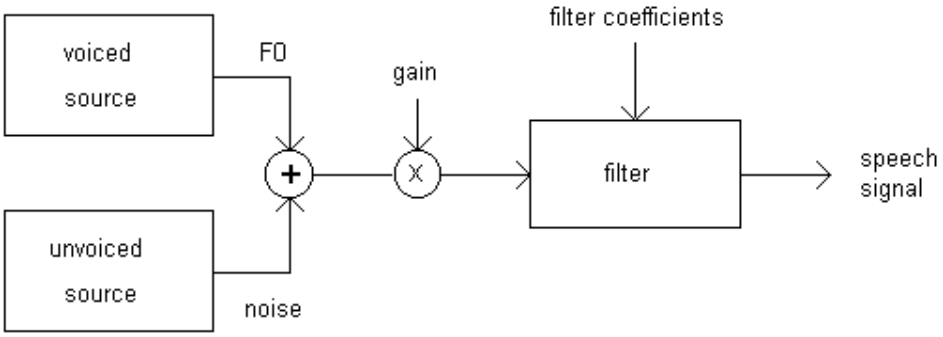



Figure 4.2: Schematic of the source-filter model of speech production Lemmetty (1999)

This model has the obvious advantage of separating the sound source from the filter and radiation load. This idea as a design philosophy, if not actually as a direct implementation is present in all of the models reviewed in the next section.

## 4.2 Linear Models

Linear models is meant here to be understood as models, which are based on the linearised equations describing wave propagation within the VT. As such they may include non-linear phenomena - particularly noise from for example frication, but they do not model noise generation process itself. Rather noise is generated with random noise generators.

Implementations of the linear models have taken two major forms: electrical analog circuits and computer simulations. The latter usually follows the former by simulating the workings of the physical circuits in a digital manner.

### 4.2.1 Electrical Analog Circuits

As the study of speech production by simulation moved from the historical attempts at constructing a mechanical artificial speaker to more systematic studies, electrical analogs of the speech organs were the first method utilised by researchers. They are based on the approximation of the acoustical variables with electrical ones as described above.

The most influential models of this type were constructed before the emergence of computer simulation. Even so, the construction of real rather than simulated circuits may yet provide an alternative avenue of research. This is possibility is demonstrated by for example Jones and Harris (1996).

#### Models

In one of the first studies using the electrical analog method with a firm relation between VT geometry and circuit construction, Dunn (1950) represents the VT by four tubes. These are in turn modeled by an analog electrical circuit of the type shown in figure 4.3. Dunn then proceeds to simplify the circuit with approximations, which result - purposefully - in a circuit without any dissipative elements. Finally, Dunn solves the circuits properties analytically for certain vowels.

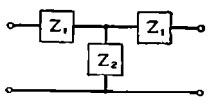



Figure 4.3: The T-section used by Dunn (1950) to represent one acoustic tube section.

After analysing some vowels analytically Dunn (1950) describes an electrical model of the VT. It consisted of 25 T-sections. Each represented a

tube with a length of 0.5 cm and a cross-sectional area of 6 $cm^2$. This system could be divided into two cavities by adding a constriction impedance shown in figure 4.4 labeled as "tongue". It could be added between any two of the T-sections. When excited with a suitable signal, this system produced good enough vowels to be found useful in research

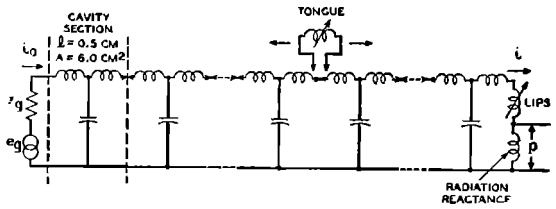



Figure 4.4: The electrical circuit used by Dunn (1950) to simulate the VT.

Stevens, Kasowski, and Fant (1953) built an electrical model of the VT with 35 tube sections represented by $\pi$-sections of the type shown in figure 4.5. Each of these had a virtual length of 0.5 cm and an individually variable area and damping. The model's length could adjusted in 0.5 cm increments by setting the impedance, which represented the lips, after any of the $\pi$-sections. Finally, the model could be excited either from the glottal end or by adding a noise source between any two of the $\pi$-sections.

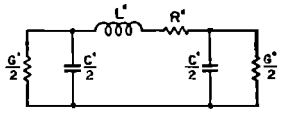



Figure 4.5: The $\pi$-section used by Stevens et al. (1953) to represent one acoustic tube section. The impedances are explained in the text.

The sounds produced by the model were reported to differ from those of real speech especially at high frequencies. While the model was quite flexible in being able to produce also other sounds besides vowels, it was limited to producing only sounds where the VT does not divide into more then one tube.

### 4.2.2 Computer Simulation

Computer simulation of linear speech signal generation has followed the path started by electrical analog circuit models. The most notable model

by far is the Kelly-Lochbaum model, which has been expanded to two and three dimensional versions in recent years. However, some studies have employed other methods for moving the electrical analog circuits into digital simulations.

### Models

The basic building block of the model developed by Kelly and Lochbaum (1962) is shown in figure 4.6. The model's principle is to model each tube junction by the reflection and transmission coefficients derived in section 4.1.3. These are used both for a wave propagating towards the mouth and a wave propagating towards the glottis. Time discretisation is accomplished by adding a unit time delay in both directions or alternatively a double delay in one direction only. These have the same overall effect, but the latter version is computationally simpler.

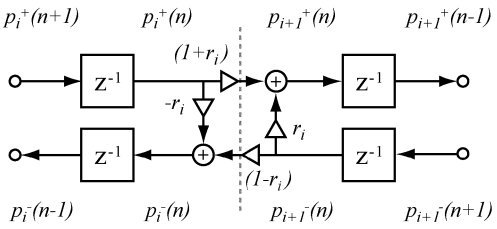



Figure 4.6: The model of an acoustic tube junction by Kelly and Lochbaum (1962) (picture from Mullen et al. (2006)). The blocks labeled $z^{-1}$ represent unit time delays. $r$ stands for the reflection coefficient $R$ and $p^{\pm}$ for sound pressure waves. The last term is a function of both node and iteration step.

The original model was found to have a fairly bad voice quality, but surprisingly good consonants. In conclusion, Kelly and Lochbaum (1962) state that contemporary formant synthesis was more satisfactory. Nevertheless, as is evident from later studies of articulatory speech synthesis, this is a very successful model.

Badin and Fant (1984) propose a simulated electrical circuit analog shown in figure 4.7. They also report some preliminary results from using the model without the subglottal system. The model itself consists of tube and horn section modeled with T-sections. Also, it has sources at the glottis and possible constrictions.

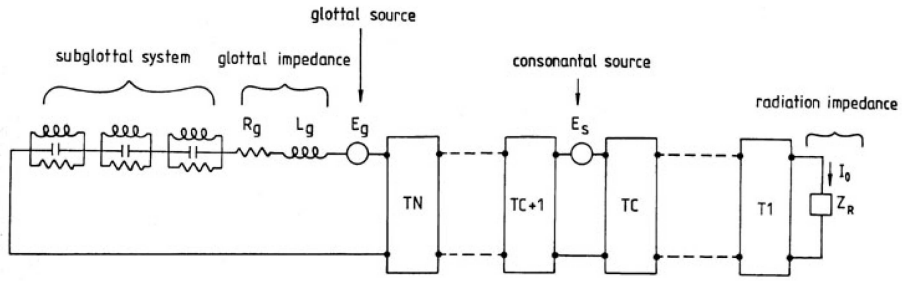



Figure 4.7: The circuit proposed by Badin and Fant (1984) to be used in a computer simulation of the VT. The blocks labeled T[N,C+1,etc] are T-sections, which represent acoustical tube sections.

Badin and Fant (1984) had all three types of energy losses in their model. Viscous and heat conduction losses were represented by resistive impedances in series with the $Z_1$ impedance of Dunn's T-section. In addition, wall losses were represented by adding a resistive element in parallel with the $Z_2$ element.

Mullen et al. (2006) expand the Kelly-Lochbaum model into a 2D model by constructing a rectilinear mesh of junctions. In this system a signal is able to propagate in four directions from each junction instead of just two. Thus, the cross-sectional area of each tube section is represented by the number of junctions, that the model has across the VT.

The sound pressure $p_J$ at each junction $J$ can be calculated by solving the finite difference equation 4.14.

$$p_J(n) = \frac{2}{N} \sum_{i=1}^{N} p_i(n-1) - p_J(n-2), \tag{4.14}$$

where $n$ is the iteration step and $N$ the number of junctions. Furthermore, the walls, the lips and the glottis are simulated by adding appropriate reflections at the boundary nodes $p_B$ of the form defined by equation 4.15.

$$p_B = (1+R)p^+_{1,J} \tag{4.15}$$

Mullen et al. (2006) compared their 2D model with a traditional 1D Kelly-Lochbaum model. A sample of the simulated vowel spectrums is shown in

figure 4.8. They found that the 2D version was not conclusively better than the 1D version, but the comparison indicated that the 2D version had the capacity to produce more natural speech sounds.

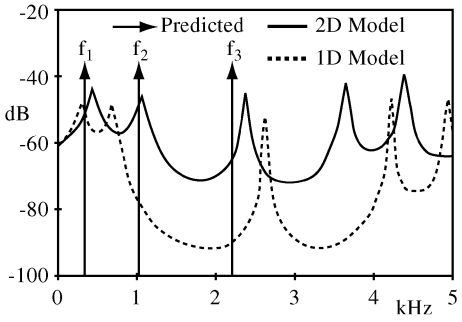



Figure 4.8: Predicted formants of [u] along with simulated frequency responses of 1D and 2D waveguide models (Mullen et al., 2006).

A 3D version of this technique was employed by El Masri, Pelorson, Saguet, and Badin (1996) to investigate higher resonance modes of the VT. As expected they found transverse modes of resonance to appear around 5 kHz.

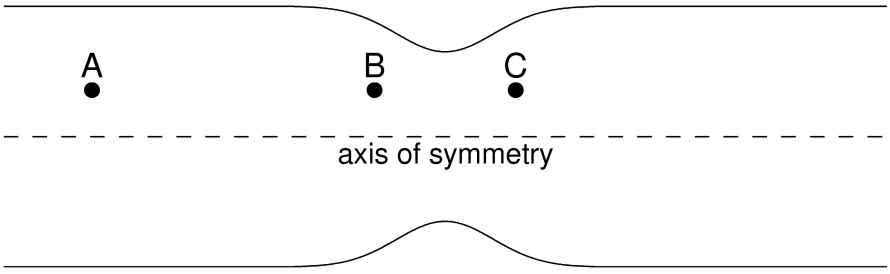
## 4.3 Non-linear Models

Even though speech sound production does have non-linear qualities, the linear model is an excellent first approximation. Therefore, rather then define a completely non-linear model, the model discussed below adds a non-linear component to linear models. This has the advantage of keeping the model modular and thus easier to manage.

### 4.3.1 Noise Generation Modeled with Vorticity

Sinder (1999) developed speech production model by applying discretised fluid dynamics to the problem of generating noise within constrictions in the VT. The model is valid not only for noise associated with fricatives, but for noise associated with other speech sounds as well.

The model is based on the theory of vorpal sound generation. The principles of this theory are summarised by figure 4.9.



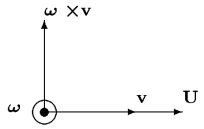
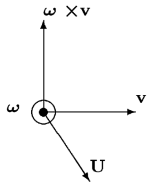
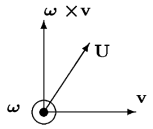

| | Position A | Position B | Position C |
|---|---|---|---|
| Vectors |   |   |   |
| $\boldsymbol{\omega} \times \mathbf{v} \cdot \hat{\mathbf{U}}$ (source strength) | negligible | negative | positive |

Figure 4.9: Illustration of vorpal sound generation theory from Sinder (1999). Illustrated are vortex positions and the relevant vector variables. Source strength is calculated as the vector product of the vorticity vector $\omega$ (measures strength and direction of rotation), the propagation velocity of the vortex $v$ and the mean irrotational flow velocity $U$.

The computational model is made of three parts, which are shown in figure 4.10. First, the jet model, which controls vortex formation and convection along the VT. Second, the mean flow model, which controls the irrotational component of flow. Third, the sound propagation model, which controls the propagation of sound from given acoustic sources. In other words, the noise generation model works by modeling fluid dynamics and after a noise source is found, the sound generated by it is fed into the traditional wave propagation model.

Sinder used the model in speech synthesis experiments. The model's accuracy was mostly limited by the accuracy of the geometrical descriptions used. Nevertheless, the model was found capable of producing not only both voiced and unvoiced fricatives, but also voiced and unvoiced stop consonant.

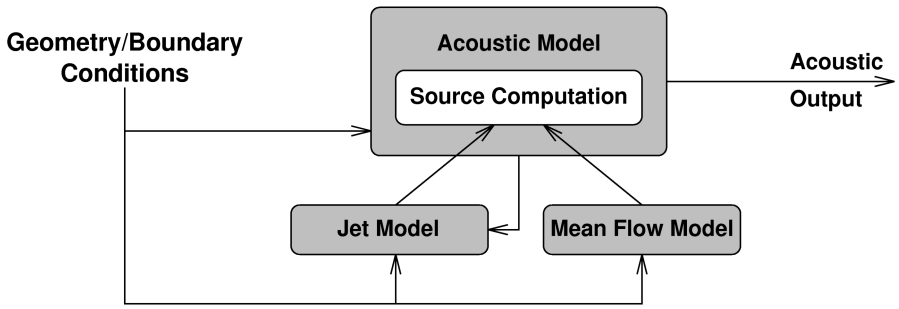



Figure 4.10: Relation ships of the submodels of the aerodynamic sound generation model by Sinder (1999).

## 4.4 Summary

This chapter concludes the story of constructing an articulatory speech synthesiser as far as it can be told within the limits of this thesis. All that remains is to discuss the overall context and future of speech production modeling. These are the topics of the final chapter.

# Chapter 5

# Discussion

As this thesis draws towards its end, it is time to look back and share some of my own thoughts on the discussions of the previous chapters.

## 5.1 Data Acquisition

While the current methods are very impressive, they are still - and probably always will be - developing towards perfection. Based on the speed of articulatory movements Perkell et al. (1992) sets the goal for the sampling frequency band of articulatory measurements as 0 to 500 Hz.

To complement this I would set goals for spatial resolution and dimensionality of the data: The former should be better than 1 mm, if possible by at least one or two orders of magnitude just to be on the safe side. At the same time the latter requirement should be set at full 3D acquisition with a continuous sampling of space.

These requirements are quite hard to reach. However, different measurement methods do already fulfil some of them: First, modern EMA systems are capable of the kind of time resolution required by Perkell. Second, MRI is already on the better side of a spatial resolution of 1 mm. This is encouraging even if MRI does not reach very good time resolution in full 3D acquisitions. Third, ultrasound is also becoming what can be considered a full 4D method.

## 5.2 Models of Vocal Tract Geometry

In my opinion, it is clear, that 3D models of the VT are the only viable choice in the long run. However, it is much harder to choose the methods for parametrisation and generation of movement. For the parametrisation I would consider two possibilities to be the best candidates: Statistical analysis in the near future and physiological modeling as time progresses.

As for the generation of movement, I do not think that I am really qualified to answer. It is very hard to form an opinion based solely on the literature. For example, classical coarticulation studies disagree on the relation of vowel and consonant timing (see for example (Öhman, 1966), (Gay, 1977)). Even, if one would accept that the model should be a coarticulatory one remains unclear as one has to consider the arguments put forward by Xu and Liu (to be published).

## 5.3 Acoustic Synthesis

The basic model for modeling VT acoustics is well established as such. What still remains unclear is the accuracy requirements, which should be posed on different parts of the model. These requirements are not the same for all sounds. Vowels require the general shape of the VT geometry to be accurate, while in fricative production the main focus is on sudden changes in the VT shape as air flows from the glottis to the mouth (Sinder, 1999).

To study the relative importance of an articulatory model's components they have to first be made as accurate as possible. Care has to be taken not to emphasise any of the components over the others. Rather they have to be made equally exact.

To this end, especially the glottal model and the radiation load connected to the mouth have to be modeled carefully. Within the VT acoustic modeling needs to be expanded into the third dimension.

Also, new methods for acoustic modeling need to be considered. The finite element method has been used recently to solve tube system resonances (Peplow & Finnveden, 2004), (Lau & Tang, 2005). As our understanding of the mathematics involved in such models progresses, they will most likely be found useful also in solving tube - or VT - dynamics.

Another important consideration in acoustic synthesis has to be the development of non-linear sound generation models. This may be even of importance for vowel modeling as phonation is reported to give raise to rotational flow within the VT, which in turn contributes to the resulting sound (Barney et al., 1999), (Shadle et al., 1999).

## 5.4 Further Topics of Interest

Two topics have been only superficially present in this text: Glottal models and the performance evaluation of an articulatory speech synthesiser. The former topic's complexity is easily comparable to that of VT modeling. Thus I can not present it here in any meaningful way. Instead, the latter topic does allow a brief discussion.

### 5.4.1 Evaluating an Articulatory Speech Synthesiser

There are several ways an articulatory synthesiser may be evaluated. Some of these have been given cursory attention in this text. Nevertheless, this is a very important aspect of any synthesiser project. Therefore, to give the reader some notion of the possibilities some common methods are presented below.

All speech synthesisers may be subjected to perceptual tests under different conditions. The tests most commonly try to establish a measure of how natural and/or how intelligible the synthesiser is. In these tests the results of synthetic speech are usually compared with the results of natural speech under the same test conditions. These conditions may include noise and/or other distracting factors. Examples of the methodology can be found in Möttönen (1999) and Ojala (2006).

An other evaluation method, common to all speech synthesisers, is direct mathematical or statistical comparison of the produced speech with a target sample of natural speech. With synthesisers, which are capable of geometric or visual output, an added dimension is comparison of articulatory data with synthetic articulatory data, such as EPG (Engwall, 2001b). Most statistical analysis methods employed in defining the model's parameters

already give one such measure - the amount of explained variance (see section 3.2.3, page 71).

## 5.5 Possible Future Directions

Listing all possible directions for future research is not possible and an attempt would not even be very interesting. For this reason I will give here only two suggestions removing idealisations and taking the physical simulations even further from their current position.

### 5.5.1 Removing Idealisations

As the computational power available for research purposes continues to grow, some of the traditional simplifications could be removed. As an example, the VT is not a straight tube. The curvature does not affect the resonances under 4kHz by more than 2% - 8% (Sondhi, 1986). Nevertheless this effect should, in my opinion, be taken into account when aiming for truly accurate articulatory synthesis.

### 5.5.2 Physical Simulations

One vision for the future might be a model where perhaps the tissues and air are separate, but the air within and without the VT is handled by one fluid dynamic model. Sound in such a model would be observed by listening to the model in its own numerical far field. Specifically, there would be no separation between the turbulence model and the acoustic model. Both would handled by simulating computationally the dynamical behaviour of the air in the system. Detailed tools for this kind of aerodynamical modeling do exist and could be applied to aeroacoustic research.

## 5.6 Conclusion

Recent research has produced some exciting results. Some could be covered by this thesis, while others could not. As an example of the latter group one should mention a finite element model of vocal folds, which is caused to vibrate by the subglottal pressure (Thomson, Mongeau, & Frankel, 2005). As such realistic models are becoming a standard in the area, it can be said with good confidence, that we live in truly interesting times.

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# Appendix A

# Webster's Horn Equation

To describe wave propagation within the VT mathematically, we will use the definitions given in table A.1.

Table A.1: Definitions for analysis of wave propagation in the VT

| | |
|---|---|
| $\rho$ | density of the medium |
| $\rho_0$ | constant component of the density of the medium |
| $p$ | perturbation pressure (sound pressure) |
| $q'$ | mass source |
| $\overline{u}$ | particle velocity |
| $U$ | volume velocity |

We will begin by supposing that the air within the VT is ideal gas. Therefore we have the law of mass preservation and Euler's equation:

$$\frac{\partial \rho}{\partial t} + \nabla \cdot \rho \overline{u} = \rho q' \tag{A.1}$$

$$\rho_0 \frac{\partial \overline{u}}{\partial t} + \nabla p = 0 \tag{A.2}$$

As we take into consideration the absence of mass sources in the VT ($q' = 0$) and the fact that the frequency range of interest (roughly from 100 to 3000 Hz) is fairly low in comparison with the frequencies of cross-mode resonances (lowest effect is around 5 kHz for an average male VT), we can consider the VT as an effectively one dimensional tube and thus get:

$$\frac{\partial \rho}{\partial t} + \frac{\partial \rho u}{\partial x} = 0, \tag{A.3}$$

$$\rho_0 \frac{\partial u}{\partial t} + \frac{\partial p}{\partial x} = 0. \tag{A.4}$$

Then we write $\rho$ as a function of $p$ with the ideal gas law:

$$\frac{\partial}{\partial t}\frac{p}{nRT} + \frac{\partial \rho u}{\partial x} = 0, \tag{A.5}$$

$$\rho_0 \frac{\partial u}{\partial t} + \frac{\partial p}{\partial x} = 0, \tag{A.6}$$

where $n$ and $R$ are the number of moles and the universal gas constant (mass of the gas element is taken as 1). If we hold the temperature $T$ to be constant in relation to time and, similarly, $\rho$ in relation to $x$, we will find, that by applying the definition of the speed of sound, the following holds for ideal gas:

$$c^2 = \left[\frac{\partial p}{\partial \rho}\right]_{S_0} = nRT \tag{A.7}$$

Thus we can write the equations as:

$$\frac{\partial p}{\partial t} + \rho_0 c^2 \frac{\partial u}{\partial x} = 0 \tag{A.8}$$

$$\rho_0 \frac{\partial u}{\partial t} + \frac{\partial p}{\partial x} = 0 \tag{A.9}$$

By expressing particle velocity as volume velocity divided by the area of the VT's cross-section ie. $u = U(x,t)/A(x,t)$, the equations become:

$$\frac{\partial p(x,t)}{\partial t} + \rho_0 c^2 \frac{\partial}{\partial x}\frac{U(x,t)}{A(x,t)} = 0 \tag{A.10}$$

$$\rho_0 \frac{\partial}{\partial t}\frac{U(x,t)}{A(x,t)} + \frac{\partial p(x,t)}{\partial x} = 0, \tag{A.11}$$

which become

$$\frac{\partial p(x,t)}{\partial t} + \frac{\rho_0 c^2}{A(x,t)}\frac{\partial}{\partial x}U(x,t) + \rho_0 c^2 U(x,t)\frac{\partial}{\partial x}\frac{1}{A(x,t)} = 0 \quad \text{(A.12)}$$

$$\frac{\rho_0}{A(x,t)}\frac{\partial}{\partial t}U(x,t) + \rho_0 U(x,t)\frac{\partial}{\partial t}\frac{1}{A(x,t)} + \frac{\partial p(x,t)}{\partial x} = 0. \quad \text{(A.13)}$$

The terms containing a partial derivative of $1/A(x,t)$ are small. This can be seen by noting that they develop into $A'/A^2$, which is small as long as the change in VT area is small in comparison to square of the actual area. More specifically, we are supposing, that the derivatives are small and of the same magnitude. If the derivative with respect to $x$ was the greater of of the two (and hence could not be removed), the following operations would lead to the normal wave equation. This holds at least for vowels in relation to both $t$ and $x$. Thus we get the following:

$$\frac{\partial p(x,t)}{\partial t} = -\frac{\rho_0 c^2}{A(x,t)}\frac{\partial}{\partial x}U(x,t) \quad \text{(A.14)}$$

$$\frac{\partial}{\partial t}U(x,t) = -\frac{A(x,t)}{\rho_0}\frac{\partial p(x,t)}{\partial x} \quad \text{(A.15)}$$

By differentiating equation A.14 in relation to time, we get:

$$\frac{\partial^2 p(x,t)}{\partial t^2} = -\frac{\partial}{\partial t}\left[\frac{\rho_0 c^2}{A(x,t)}\frac{\partial}{\partial x}U(x,t)\right] \quad \text{(A.16)}$$

Then we further note that again after differentation the terms containing $A'/A^2$ are small and that the differentation order of $p$ can be changed. Thus, we get:

$$\frac{\partial^2 p(x,t)}{\partial t^2} = -\frac{\rho_0 c^2}{A(x,t)}\frac{\partial}{\partial x}\frac{\partial}{\partial t}U(x,t) \quad \text{(A.17)}$$

Finally, by substituting equation A.15 into equation A.17 we get the equations combined into a form known as Webster's equation:

$$\frac{\partial^2 p(x,t)}{\partial t^2} = c^2\frac{1}{A(x,t)}\frac{\partial}{\partial x}\left[A(x,t)\frac{\partial p(x,t)}{\partial x}\right] \quad \text{(A.18)}$$

## A.1 Notes on Deriving the Equation

The idealisations involved in deriving equation A.18 have certain consequences some of which I will discuss here.

First and foremost is the step where we remove two of the original three spacial dimensions. As a result, there can be no cross-modes or vortices present in the resulting model. The absence of cross-modes should not be a problem as long as we are not interested in formants beyond the first two, but after that the effects are probably considerable. Likewise, the absence of vortices or turbulent flow does not cause problems, if we are interested in modeling only vowel production. However, consonant - and escpecially fricative, modeling will need some other mechanism for producing turbulent noise, if the above 1D equations A.8 and A.9 are used.

A further thing to consider are the assumptions, which removed losses from the model. These are, firstly, the fact, that we implicitly assume the walls of the vocal tract to be infinitely hard. Thus no energy is dissipated as sound propagation into the tissues surrounding the VT. Secondly, we assumed the air within the VT to be ideal gas. This in turn means, that there are no viscous or heat conduction losses within the gas itself. Fortunately, these types of losses can be, and successfully have been, added to 1D models.

Finally, we should remember, that the equations A.1 and A.2 assume that there is no flow within the VT, and the assumption, that $A'/A^2$ is small. The first of these clearly does not hold. The flow can of course be assumed to behave in a suitably regular way so, that its effects can be left out as a minor approximation. Nevertheless, as models of speech production grow more sophisticated this idealisation has to be eventually removed. As for the second assumption, it holds - as stated above - only for vowels. Especially in the case of plosive or tremulant sounds, such an approximation can not be said to be true.