YOU CAN'T MEASURE IT BUT YOU CAN KNOW IT: PRECISELY SYNTHESISING ACOUSTIC FLOW

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Abstract

Typical acoustic currents are usually far too small to be measurable with any precision. This paper presents a technique that uses an impedance head to generate with precision a desired, acoustic current waveform, usually periodic. An impedance head with multiple microphones, after a suitable calibration procedure, allows calculation of the acoustic pressure and current waveforms at the appropriate reference plane. The current waveform injected by such a device is a complicated function of the waveform supplied to the transducer driving the impedance head, the acoustic properties of the impedance head itself, the frequency responses of the transducer and microphones, and the frequency responses of the electronic circuitry involved. It is possible however, using an iterative technique, to adjust the waveform driving the input transducer so that the waveform of the current injected into the device under study has the desired waveform. In this fashion an impedance head can be used to synthesise and to inject a precisely known acoustic flow waveform. As an application we report that the synthesis of a model glottal flow waveform which is then input to a rigid, unvarying physical model of the vocal tract. This allows, for the first time, the rigorous evaluation of the various techniques that are used to determine the acoustical properties of the vocal tract and the glottal waveform under a wide range of conditions.

1. Introduction

It is relatively easy to produce an acoustic flow or current; loudspeakers and piezoelectric drivers are readily available. Knowing what flow they produce, however, is only easy in cases with simple geometries, such as one dimensional flow with no reflections, in which case it is known from the measured pressure and the characteristic impedance. Measuring acoustic currents has thus always posed a problem for experimenters. First, typical acoustic velocities are small: at 60 dB in air, the rms speed $v = p/\rho c = 50\ \mu\text{m s}^{-1}$; at 100 dB in air, the speed is only 5 mm s^{-1}. If a suitable optical system is available, and if the flow can be labelled with suitable and identifiable particles, such speeds can be measured with particle velocimetry.

One method of measuring acoustic currents involves hot wire anemometry; a technique that measures the cooling effect of airflow on the resistance of a fine wire, e.g. [1]. The wires used can be very thin (*e.g.* 5 µm diameter), but are consequently fragile, and their thermal inertia limits the high frequency response. A superior modern version is the 'microflown' that is manufactured using a pair of very fine (200 nm diameter) heated platinum wires on a silicon substrate [2]. Although the microflown can measure acoustic currents over the full audio range, it is still necessary to insert the device into the current flow and consequently disturb it. This can be significant in many applications. Furthermore, the sensitivity of all such measurements is limited because of the small velocities of interest.

Another method involves determining the flow by measuring the pressure on both sides of a compact acoustic resistance: the flow is then proportional to the pressure difference. Introducing this resistance of course can change the geometry being measured very significantly. Nevertheless, this approach is used to measure the oral volume velocity during speech and singing; a specially constructed, circumferentially vented, pneumotachograph mask covers the subject's face and the flow determined from the pressure difference developed across the resistance of a very fine mesh [3]. A problem with such compact resistances is that they cease to be compact at short wavelengths, and that they are usually not purely resistive at all frequencies: they tend to have non-negligible inertance and compliance at high and low frequencies respectively and this makes calibration difficult.

Since setting up our lab late last century, we have used a different approach, which we describe here. Rather than using an acoustic current source with unknown characteristics and trying to measure the current flow so produced, we inject an acoustic current which is known, either from a previous calibration or from simultaneous pressure measurements in a calibrated waveguide. Injecting a current with a known waveform removes the need to measure the current in the system under investigation. We list and refer to a number of examples of the technique.

2. Producing a known acoustic flow using a high impedance current source

One of the simplest methods of producing a known acoustic flow is to use a calibrated source with very high output impedance [4]. Providing that the impedance of the load is much smaller than that of the source, the current will be independent of the load. Consequently the source can be calibrated by measuring the pressure produced when connected to a known reference load. The requirement that the source impedance be very much larger than the load can be a disadvantage if the acoustic impedance of the load is large at some frequencies, which is often the case for resonant systems. Further, to generate useful flows from a high impedance source requires a driver or loudspeaker operating at high pressure. This can raises problem with distortion and the possible transmission of sound from the loudspeaker to the device under test either through the air or via mechanical vibration.

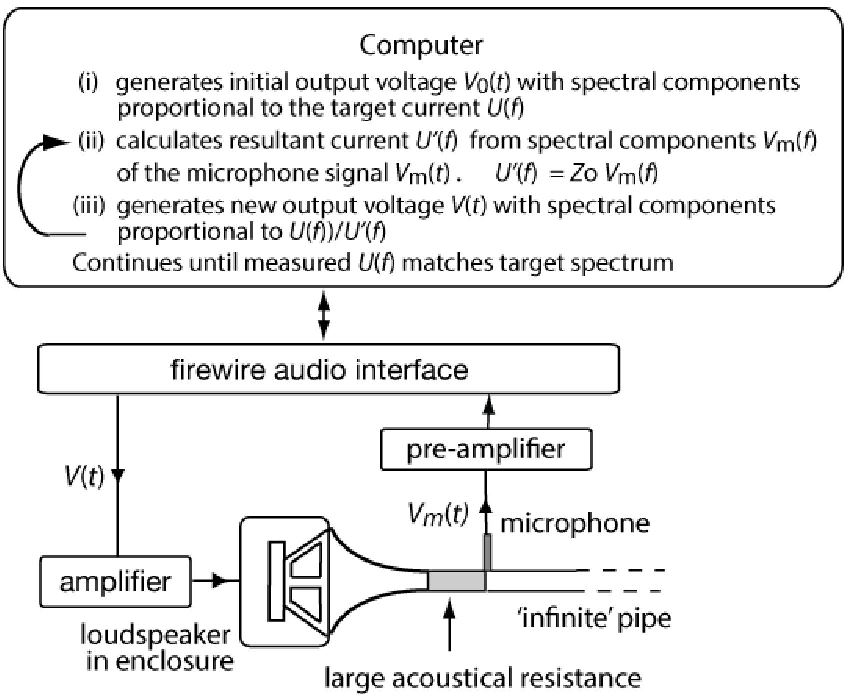

Figure 1. Schematic diagram (not to scale) showing a high impedance current source during calibration using an acoustically infinite pipe

A further problem is the difficulty of producing a sufficiently high and reliable source impedance. Backus (1974) used a very small annular gap (~100 µm) between a cylinder and an internal rod [4]. For our experiments we used a truncated slightly conical plug and socket that were lapped until they fitted precisely together; they were then separated and replaced with three very fine wires (120 μm diameter) now acting as spacers [5]. This can produce values in the range 100 to 200 MPa s m^{-3}. Although it is possible to correct for the finite value of the source impedance [5], variations in its value due to condensation in the very small annular gap can cause difficulties.

Calibrating such a source requires loads with known impedance. Ducts with known geometry usually have singularities at their resonant frequencies. These can be avoided, however, using the pure resistance of an acoustically infinite pipe [5,6]. For this purpose, we have installed long cylindrical ducts in the ceiling spaces of the physics building at UNSW.

Usually we generate an acoustic current $U(f)$ with a flat spectrum, but in several situations it is to advantageous to alter the spectrum of $U(f)$ to improve the signal to noise ratio. Figure 1 illustrates the iterative procedure used to produce a known current with the desired spectrum. The useful range of these measurements can extend up to 12 kHz [7].

3. Producing a known acoustic flow using an impedance head

Many of the above problems can be overcome by using a 2- or 3-microphone impedance head as the current source. The flow $U(t)$ at a reference plane within the duct of the head can be determined from the measured signals of the microphones in the impedance head. First, consider the case with just two microphones. If the duct is uniform and the microphones identical, then the complex amplitudes of the travelling pressure waves in the two opposite directions within the head, $p_{\mathrm{left}}(x,t)$ and $p_{\mathrm{right}}(x,t)$, can be solved at any frequency from the two measured microphone signals. The acoustic flow to the right at the reference plane at the output of the head ($x = 0$) is then simply given by $(p_{\mathrm{right}}(0) - p_{\mathrm{left}}(0))/Z_0$, where Z_0 is the characteristic impedance of the cylindrical duct.

In practice, a singularity arises when the distance between two microphones is half a wavelength [8]. Consequently, a close spacing is needed for high frequency response. At close spacing, however, the phase difference between the two microphones is small at lower frequencies, so sensitivity is then low. For this reason, we use three microphones and solving the simultaneous equations (see previous paragraph) is replaced by minimising the squares of residues. Of course the microphone gains and local geometries are never identical. For this reason, we calibrate using three non-resonant loads: a very large impedance, a large flange open to the radiation field and the acoustically infinite cylindrical ducts mentioned in the preceding section [9].

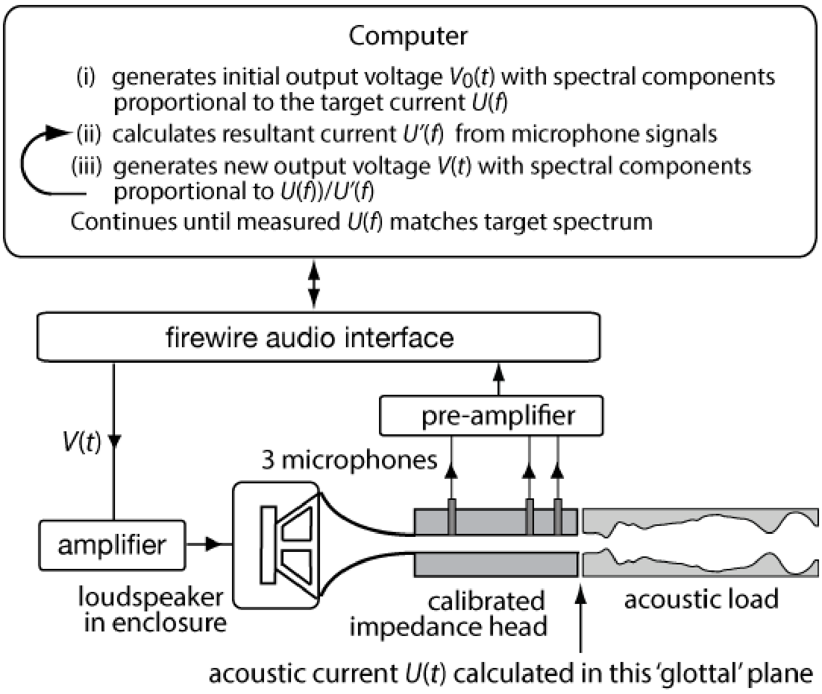

Figure 2. Schematic diagram (not to scale) showing an impedance head used to inject a known acoustic current

Using such an impedance head, a desired acoustic flow may be produced at the reference plane by employing an iterative, inversion technique [6] – see figure 2. The complex Fourier components *U*(*f*) of the desired periodic glottal flow *U*(*t*) are first determined. The computer then synthesises a voltage signal $V_O(t)$ with spectral components of magnitudes *Vo*(*f*) that are proportional to the desired *U*(*f*). This signal is connected to the amplifier and loudspeaker, and the spectral components *U'*(*f*) of the flow *U'*(*t*) at the output of the impedance head are calculated from the microphone signals. The computer then synthesizes a new output voltage *V*(*t*) with complex components proportional to *U*(*f*)/*U'*(*f*) that should, in a completely linear system, produce the desired *U*(*t*) at output of the impedance head. Sometimes additional iterations are required, due mainly to small nonlinearities in the loudspeaker.

4. Measuring impedance using a known acoustic flow

Our first use of a known acoustic flow was to measure acoustic impedance by measuring the pressure developed when the known current is applied to the system under investigation. Initially, we used a broadband acoustic current whose spectrum was independent of frequency [6]. In later studies, we refined this system, adapting the spectrum of the acoustic current to increase the signals at frequencies where the response of the measured system was weak, or where noise was greatest [9]. The input impedance of several families of wind instruments including flutes [5], clarinets [10], didjeridus [11] saxophones [12], and trumpet, horn and trombone [13,14] have been investigated in this fashion – see figures 3 and 4 for some examples. Furthermore, it has proved possible to determine the influence of the vocal tract by measuring its impedance whilst playing the didjeridu [15], clarinet [16], saxophone [17,18], trumpet [19] and trombone [20].

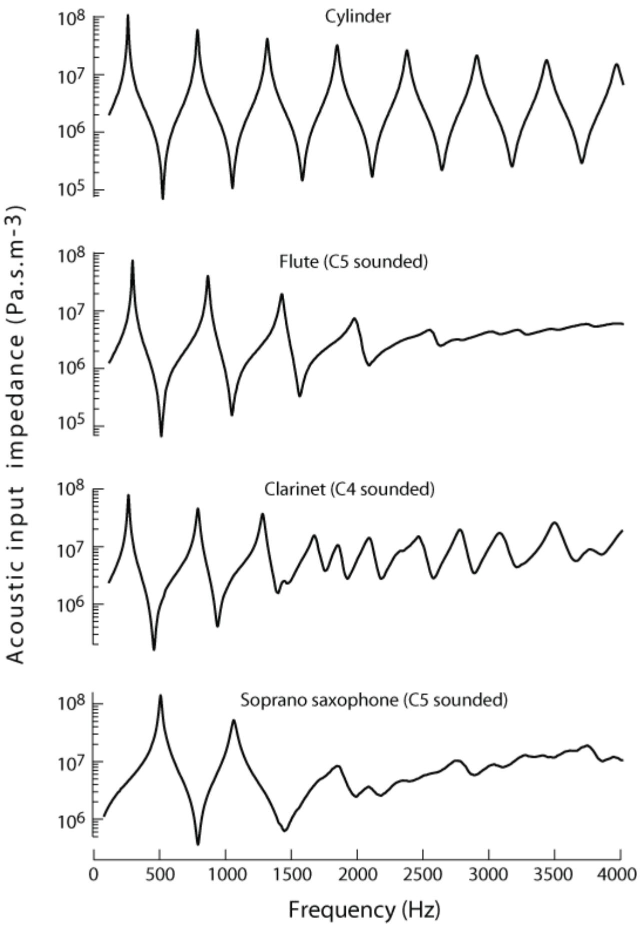

Figure 3. Examples of the measured input impedance of three woodwind instruments. The impedance of a cylinder of effective length 325 mm, a half wavelength of the note C5, is also included for comparison. See [14] for further discussion

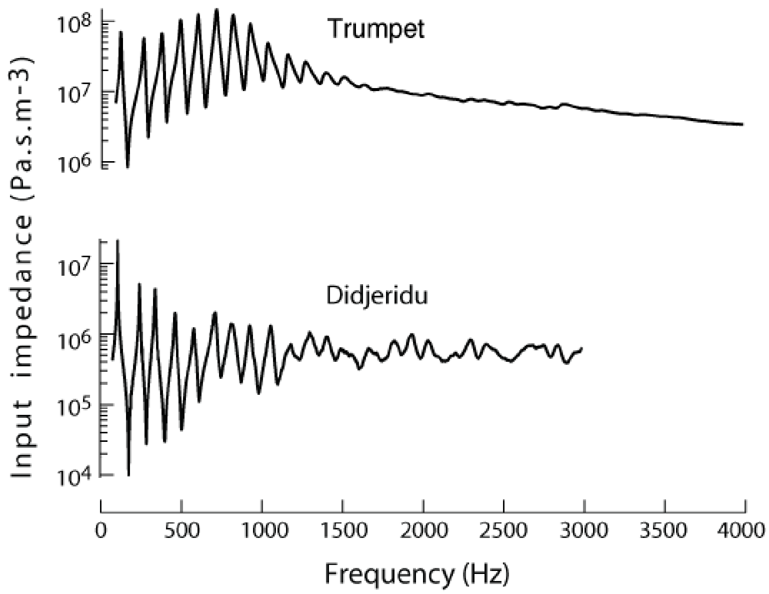

Figure 4. Examples of the measured input impedance of two lip valve instruments; a Bb trumpet with no valves depressed, and a didjeridu. See [14] for further discussion

5. Evaluating the method of inverse filtering using a known acoustic flow

Interesting applications arise when this scheme is used to provide an example of known flow in the time domain $U(t)$. As an example we report some testing of various schemes for inverse filtering, a technique used to determine the flow $U(t)$ into the vocal tract [21].

Although the waveform of the acoustic flow through the glottis — the aperture between the vocal folds — plays a fundamental role in the production of speech, it cannot usually be measured directly in normal speech. Instead, it is often estimated from the pressure signal measured outside the mouth using inverse filtering; a procedure whereby the filtering effects of the vocal tract and the radiation from the lips are removed by adjusting inverse filters until the derived glottal flow has the expected characteristics. This approach, which involves the separation of two unknowns with complex behaviour (the glottal flow and the tract transfer function) from a single signal, cannot be unequivocal and will inevitably introduce some uncertainty [22]. A major difficulty is that there has hitherto been essentially no method of testing the performance of the widely used and various inverse filtering algorithms. Also it is often assumed that the glottal flow is zero during some portion of the glottal waveform, yet there is some experimental evidence that a non-negligible 'sweeping flow' of current occurs during such a phase [23].

Our approach of generating a known glottal flow into a model tract with known characteristics and measuring the resultant speech sound means that rigorous testing of the various algorithms associated with inverse filtering is now possible.

5.1 Synthesising the glottal flow waveform.

We chose a representative glottal waveform from Alku (1992) [24] with a glottal contact quotient of 0.5; i.e. there is zero flow for half of each cycle. The complex Fourier components $U(f)$ of the desired periodic glottal flow $U(t)$ were then determined (see figure 5). The desired glottal flow was then generated as described in section 3. The glottal waveform generated had a fundamental frequency of 172 Hz (= 44.1 kHz/2^8) with harmonic components extending to 3.8 kHz (i.e. the 22nd harmonic).

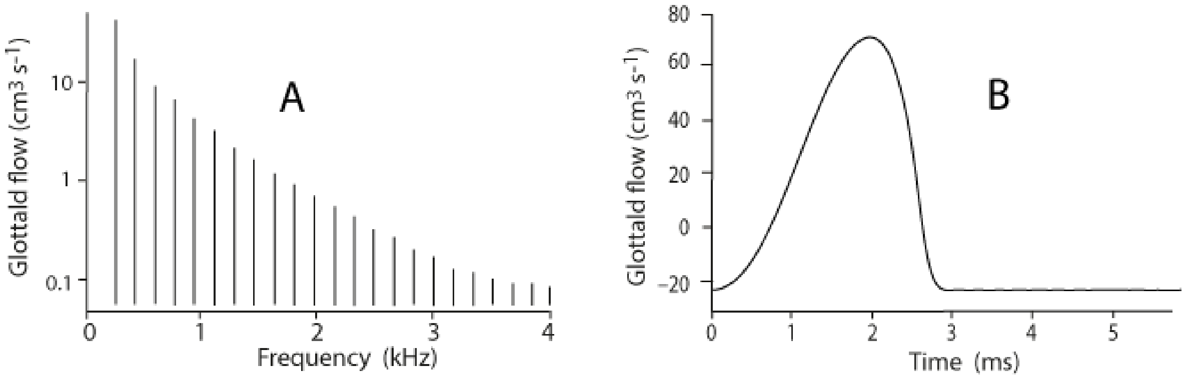

Figure 5. Synthesis of a periodic glottal current. The Fourier components with magnitudes shown in (A) when combined with appropriate phases produce the glottal waveform in (B)

5.2 A hardware model of the vocal tract

Physical models of the tract were constructed using a 3D printer with profiles based upon the area function $A(x)$ for the vowel /æ/ taken from a study based on MRI images [25]. The resonance frequencies of this tract could then be measured at the glottis using the 3-microphone impedance head. It is worth noting that the resonances of this physical model are expected to have higher Q-factors than would be present in a human vocal tract, where the magnitude of the (complex) visco-thermal loss factor α is typically increased by a factor of five [26].

5.3 Testing inverse filtering.

The impedance head was calibrated, programmed to produce the desired glottal flow and connected to a hardware model of the vocal tract at its 'glottis' – see figure 6. The radiated sound was then measured, along with the gain (the transpedance) of the tract.

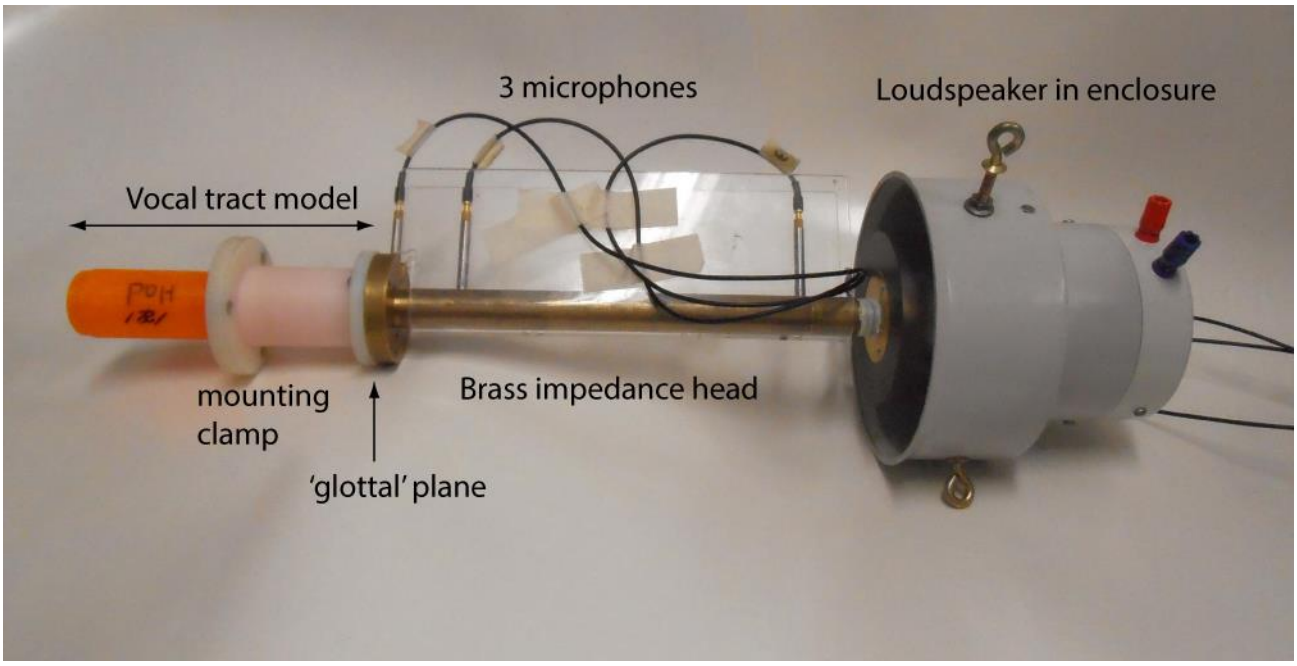

Figure 6. Photograph showing the 3 microphone brass impedance head connected to a physical model of the vocal tract (orange). Part of the vocal tract model is obscured by the white mounting clamp that holds the model securely against the output of the impedance head. The impedance head is used first to measure the impedance of the tract, and then to inject the synthesised glottal current

Figure 7 compares the glottal flow as determined by examples of two different inverse filtering techniques. As might be expected, there is insufficient information in the output sound to be able to estimate the glottal waveform exactly. It is also possible to examine the differences between the known resonances of the vocal tract model, and those estimated via inverse filtering – in this trial we found that some resonances could be missing, that some frequencies could be underestimated, and that bandwidths were overestimated [21]. It would also be possible to use the impedance head to produce an undistorted sinusoidal acoustic current by adapting our earlier method of generating undistorted acoustic sine waves [27].

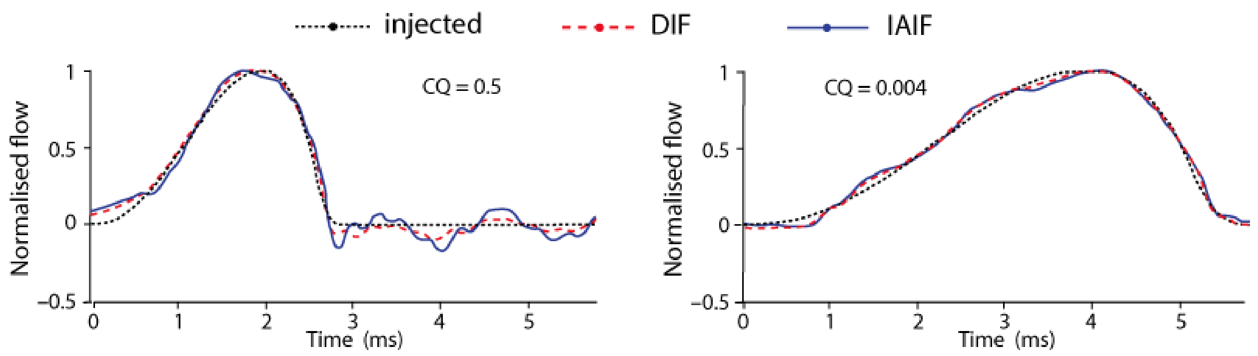

Figure 7. A comparison of the glottal flow estimated using direct inverse filtering (DIF) and iterative adaptive inverse filtering (IAIF) with the actual injected synthesized glottal flow. Results are shown for two glottal flow waveforms with closed quotients CQ = 0.5 (left) and CQ = 0.004 (right). Both used estimates used models of order 8. From [21]

5. Conclusions

It is difficult to measure acoustic currents directly. Our approach, rather than using a source with unknown characteristics and then trying to measure the current, is to inject an acoustic current which is precisely known from pressure measurements made within the impedance head.

Acknowledgements

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