Ingo Titze, Associate Editor

What Is Inverse Filtering?

Ingo R. Titze and Anil Palaparthi

Ingo R. Titze

Anil Palaparthi

Most vocologists understand the concept of filtering. In the voice technology world, it means the enhancement or attenuation of selected frequencies from a given spectrum of frequencies. The vocal tract acts like a series of band-pass filters, one for each resonance. Thus, a spectrum of frequencies from the source (the glottal airflow) is filtered by the vocal tract to produce the output at the mouth. The perception of vowels and consonants is largely determined by this filter rather than the source. Nevertheless, the source determines the strength and the tone quality (timbre) of the sound produced.

Less understood is the concept of inverse filtering. The spectrum of frequencies produced by the source is not easy to measure directly because microphone or airflow detector placement at glottal exit is invasive. Investigators generally resort to a technique known as inverse filtering, basically undoing what the vocal tract does as a filter.[1] Either the acoustic pressure radiated from the mouth or the acoustic flow at the mouth can be inverse filtered to obtain an estimate of the glottal airflow wave shape.[2]

Figure 1b shows an example of an original glottal wave shape and an inverse filtered wave shape obtained from the radiated pressure in Figure 1a. The inverse filtered flow wave shape was obtained without any *a priori* knowledge of the vocal tract (i.e., no knowledge of the vowel or any vocal tract dimension). The inverse filter algorithm assumes that the source and the filter are independent in their acoustic characteristics, one being separable from the other. In other words, there is no source filter interaction. The algorithm also assumes that the filter is linear in its response, meaning that its output is directly proportional to its input.[3] Unfortunately, neither of these assumptions is true, which means that errors should be expected. Figure 1b shows that that the open phase of the glottal flow is predicted better than the closed phase. The closing phase, closed phase, and the glottal flow amplitude estimation were considerably inaccurate.

The prediction gets better when something about the vocal tract is known *a priori*. Figure 2 shows results when a known impulse response of the vocal tract is included in the inverse filtering procedure. The impulse response was obtained by exciting the supraglottal vocal tract with a unit impulse and measuring the radiated pressure signal.

In summary, inverse filtering is useful technique for estimation of the glottal airflow pulse, but it is an indirect technique based on assumptions that simplify the relation between the source and the filter. Caution must be used in the interpretation of subtle features in the glottal airflow, some of which may be an artifact resulting from the simplifying assumptions.

Journal of Singing, November/December 2019
Volume 76, No. 2, pp. 181–182

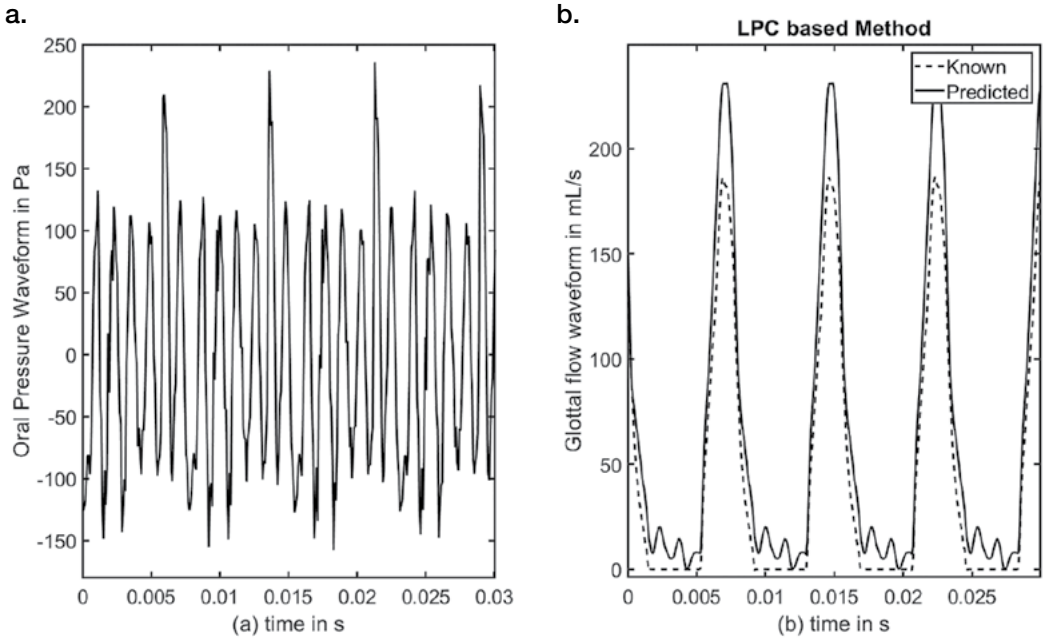

Figure 1. (a) Oral pressure waveform for an / ɑ / vowel; (b) known and predicted glottal airflow waveform from inverse filtering with no knowledge of the vocal tract

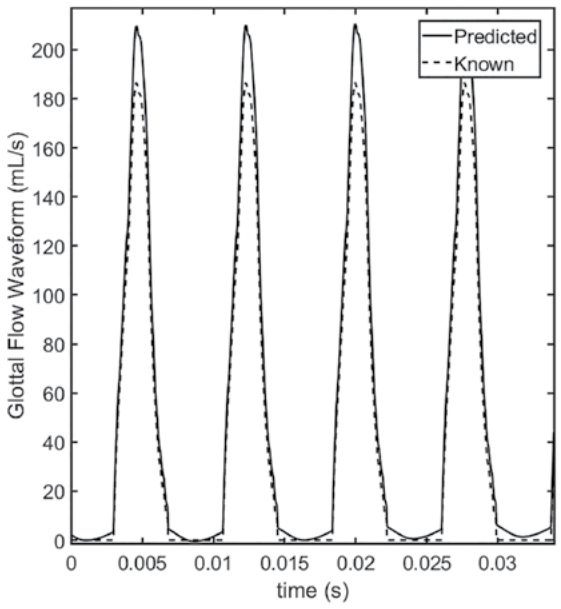

Figure 2. Known and predicted glottal flow waveform for an / ɑ / vowel from inverse filtering the oral pressure in Figure 1a with *a priori* knowledge of the vocal tract impulse response.

NOTES

1. P. Alku, "Glottal Inverse Filtering Analysis of Human Voice Production—A Review of Estimation and Parameterization Methods of the Glottal Excitation and Their Applications," *Sadhana* 36, Part 5 (October 2011): 623–650.
2. M. Rothenberg, "A New Inverse-Filtering Technique for Deriving the Glottal Flow Waveform During Voicing," *Journal of the Acoustical Society of America* 53, no. 6 (June 1973): 1632–1645.
3. M. Rothenberg and S. Zahorian, "Nonlinear Inverse Filtering Technique for Estimating the Glottal Area Waveform," *Journal of the Acoustical Society of America* 61, no. 4 (April 1977): 1063–1071.

Anil Palaparthi received his BE degree in electronics and communications engineering from the Andhra University, Visakhapatnam, India, in 2007, and the MS degree in electrical and computer engineering from the University of Utah, Salt Lake City, in 2009. He is currently working toward his PhD degree in biomedical engineering, also at the University of Utah, Salt Lake City.

From 2010 to 2015, he was a Research Scientist at National Center for Voice and Speech, The University of Utah, Salt Lake City. His research interests include motor learning, voice biomechanics, acoustic phonetics, and application of signal processing techniques toward voice and speech processing.