Linear Predictive Coding and the Internet Protocol

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Preface

The origins of this monograph lie in a talk I gave at the Special Workshop in Maui (SWIM) in January 2004 titled *California Coding: Early LPC Speech in Santa Barbara, Marina del Rey, and Silicon Valley 1967–1982* and on the historical research I conducted in preparation for the talk, including oral histories I gathered in 2003 from several of the principal players. During subsequent years the presentation evolved as I gave similar talks as part of the IEEE Signal Processing Society Distinguished Lecture Program and elsewhere. Topics and stories were added and deleted and tuned to different audiences as I received questions, comments, and anecdotes from a variety of listeners, from vintage pioneers to novice students. Some of the material converged in a short paper I wrote for the *IEEE Signal Processing Magazine*[46], and the talk material largely converged in summer 2007 with a new title of "Packet speech on the ARPAnet: A history of early LPC speech and its accidental impact on the Internet Protocol." Since giving the original talk I have hoped to find the time to collect the material into a single document combining several variations of the talk along with more of the stories and details, and this monograph is the result. As with the presentations, the historical details require an overview of linear prediction and its many cousins. While there are several excellent

treatments of linear prediction and its application to speech processing in book and survey form (see in particular the classic references by Makhoul [87] and by Markel and Gray [102]), the historical prerequisites for this monograph provide a natural motivation for providing my own overview emphasizing certain key common points and differences among the many viewpoints and approaches. The first part of this monograph is a technical survey of the fundamental ideas of linear prediction that are important for speech processing, but the development departs from traditional treatments and takes advantage of several shortcuts, simplifications, and unifications that come with years of hindsight. In particular, some of the key results are proved using short and simple techniques that are not as well known as they should be, and I use the occasion to comment on some of the common assumptions made when modeling random signals. The reader interested only in the history and already familiar with or uninterested in the technical details of linear prediction and speech may skip Part I entirely.

I have revisited all of the talks, my notes, the oral histories, my e-mail archives, and the piles of documents collected along the way. I have also chatted and exchanged e-mails recently with several principal players in the story I had not previously talked with and read through relevant documents from that time that I had not seen before. I have browsed the Internet for new views of old events, but I have preferred my original sources and have tried to not dilute the primary source material used in the talks or acquired since. I have drawn on published oral histories of some of the participants whom I did not personally interview to fill in some gaps.

As I admitted with the title of the original SWIM presentation, the original talks emphasized the Pacific Rim side of the story, reflecting my bias toward telling parts of the story which I thought were less well known at the time. This was a proper focus for the story of the serendipitous connections between the University of Southern California's Information Sciences Institute (USC/ISI) in Marina del Rey and the Speech Communications Research Laboratory (SCRL) in Santa Barbara that smoothed the cooperation of signal processors implementing the Itakura–Saito autocorrelation method of speech coding with the network scientists designing a protocol to facilitate realtime

signal processing on a rapidly growing and evolving network. During 2009 the article expanded to include more details about other partners in the project, especially MIT Lincoln Laboratory, Bolt, Beranek, and Newman (now BBN), and Stanford Research Institute (SRI, now SRI International). It is a notable part of the story that many of the key participants were with smaller organizations and universities, and not with the most famous research universities and laboratories of the time. Many other institutions play important roles which are discussed, especially Culler-Harrison Inc. (CHI) on the West Coast and Bell Telephone Laboratories (Bell Labs), and the National Security Agency (NSA) on the East Coast.

The central story is the one of the title, the combination of linear predictive coding with packet network protocols and the hardware of the time that led directly to the first successful understandable real-time digital speech on the ARPAnet, the packet network developed by the Advanced Research Projects Agency (ARPA),[1] and indirectly to the separation of the Internet Protocol (IP) and its descendents from the earlier Transmission Control Protocol (TCP). Many of the heroes of this story are listed in Figure 15.1 in Part II — the members of the Network Speech Compression (NSC) group formed by Robert (Bob) Kahn of ARPA and chaired by Danny Cohen of USC/ISI. Historical threads of the main story led to other significant technical developments, several of which are described in the history and epilogue.

In an admitted abuse of author's privilege, I have included a few technical and historical comments on topics related to the primary stories that are of lesser impact than the principal stories but are nonetheless personal favorites.

Related material of interest deemed inappropriate for print is provided as Web links, which may also be found at http://ee.stanford.edu/∼gray/lpcip.html. My favorite item is an MPEG-1 video made in 1978 to illustrate the early packet speech conference calls on the

[1] The Advanced Research Projects Agency was first created in 1958. It was renamed as the Defense Advanced Research Projects Agency (DARPA) in March 1972, renamed ARPA in February 1993, and renamed DARPA again in March 1996. We will use ARPA in this history.

ARPAnet, It effectively demonstrates the speech quality and computing equipment of the time (as well as the clothes and hair styles). I recommend the video to all students of speech processing or computer networks with an interest in history. With time I hope to scan some of the rare documents of the time and post them as well.

The two parts of the monograph can be read separately by those lacking the interest or patience for both. The first part is a largely self-contained overview of the fundamentals of linear prediction relevant to speech processing and other similar fields such as the analysis of geophysical and biomedical data. The second part is a technical history which can be read without the details of the first part; however, backwards references to the first part may be necessary for notational and vocabulary details. Writing history is a work in progress, and I recognize that some of my interpretations may be questioned and some of the historical facts may be disputed or in error. I have made every effort to resolve all of the inconsistencies or controversies that I encountered. I welcome comments and corrections that I will endeavor to include in future versions.

Those wishing a much shorter treatment of the primary story without the linear prediction material may prefer my note [46] or the Internet history coauthored by my first PhD student Barry Leiner and a list of Internet luminaries [79]. Additional Internet history resources can be found at http://www.pcbargainhunter.com/articles/history-of-the-internet.html.

Dedicated to Mike McCammon, 1943–2008

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