UNITED STATES PATENT AND TRADEMARK OFFICE

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BEFORE THE PATENT TRIAL AND APPEAL BOARD

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ZTE USA, INC.
Petitioner

v.

SAINT LAWRENCE COMMUNICATIONS LLC
Patent Owner

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Case: IPR2016-00705
Patent No. 7,260,521

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# DECLARATION OF ODED GOTTESMAN, PH.D.

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Patent Trial and Appeal Board
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P.O. Box 1450
Alexandria, VA 22313-1450

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I, Oded Gottesman, hereby declare as follows:

# I. INTRODUCTION

## A. Background

1. My name is Oded Gottesman. I am a researcher and consultant working in areas related to speech and audio coding and enhancement, digital signal processing, telecommunications, networks, and location and positioning systems.

2. I have been retained to act as an expert witness on behalf of SAINT LAWRENCE COMMUNICATIONS Inc. ("Patent Owner") in connection with the above captioned Petition for Method and System Review of U.S. Patent No. 7,260,521 ("Petition") submitted by ZTE USA, INC. ("Petitioner"). I understand that this proceeding involves U.S. Patent No. 7,260,521 ("the '521 Patent"), titled "High frequency Content Recovering Method and Device for Over-Samples Synthesized Wideband Signal." The '521 Patent is provided as Exhibit 1001.

3. I understand that Petitioner challenges the validity of Claims 1, 2, 5, 7, 8, 10, 11, 14, 15, 17, 28, 29, 32, 33, 35, 37, 38, 41, 42, 44, 55, 56, 59, 60 and 62 of the '521 Patent (the "challenged claims").

4. I have reviewed and am familiar with the '521 Patent as well as its prosecution history. The '521 prosecution history is provided as Exhibit 1003. Additionally, I have reviewed materials identified in Section III.

5. As set forth below, I am familiar with the technology at issue as of the effective filing date of the ‘521 patent. I have been asked to provide my technical review, analysis, insights, and opinions regarding the prior art references that form the basis for the Petition. In forming my opinions, I have relied on my own experience and knowledge, my review of the ‘521 Patent and its file history, and of the prior art references cited in the Petition.

6. My opinions expressed in this Declaration rely to a great extent on my own personal knowledge and recollection. However, to the extent I considered specific documents or data in formulating the opinions expressed in this Declaration, such items are expressly referred to in this Declaration.

7. I am being compensated for my time in connection with this covered patent review at my standard consulting rate, which is $525 per hour. My compensation is not contingent upon and in no way affects the substance of my testimony.

**B. Qualifications**

8. I am a citizen of the United States, and I am currently employed as the Chief Technology Officer (“CTO”) of Compandent, Inc.

9. My curriculum vitae, including my qualifications, a list of the publications that I have authored during my career, and a list of the cases in which, during the previous four years, I have testified as an expert at trial or by deposition,

is attached to this report as Attachment A. I expect to testify regarding my background, qualifications, and experience relevant to the issues in this investigation.

10. I earned my Bachelor of Science degree in Electrical Engineering from Ben-Gurion University in 1988.

11. In 1992, I earned my Master of Science degree in Electrical and Computer Engineering from Drexel University, which included performing research at AT&T Bell Labs, Murray Hill, at the time considered the world "holy grail" of speech processing research. My research was in the area of wideband speech coding, and titled "Algorithm Development and Real-Time Implementation of High-Quality 32kbps Wideband Speech Low-Delay Code-Excited Linear Predictive (LD-CELP) Coder". The work continued a prior research by E. Ordentlich, and Y. Shoham who was also my M.Sc. research advisor. As a part of my work, I have also implemented that algorithm in DSP Assembly Language on two DSPs running in parallel. I subsequently co-authored and published two articled about this work.

12. I earned my Doctorate of Philosophy in Electrical and Computer Engineering from the University of California at Santa Barbara in 2000.

13. I have worked in the field of digital signal processing ("DSP") for over 25 years, and have extensive experience in DSP research, design, and

development, as well as the design and development of DSP-related software and hardware. Presently, I am the CTO of Compandent, a technology company that develops and provides telecommunication and DSP-related algorithms, software and hardware products, real-time DSP systems, speech coding and speech enhancement-related projects, and DSP, software, and hardware-related services. While at Compandent, I have contributed to a speech coding algorithm and noise canceller algorithm that has been adopted for secure voice communication by the U.S. Department of Defense & NATO. Currently, I am supporting the DoD's and NATO's use of these algorithms, and am performing real-time implementation projects for DoD and NATO vendors, as well as the Defense Advanced Research Projects Agency (DARPA).

14. I have worked for numerous different companies in the field of digital signal processing during my career. I am very familiar with most, if not all, speech coding, speech enhancement, audio coding, and video coding techniques for various applications. As part of my work, I have developed real-time DSP systems, DSP software for telephony applications, various serial communication software and hardware, and Internet communication software. I have led real-time DSP and speech coding engineering groups in two high-tech companies before my present company (Comverse Technology, Inc. and Optibase Ltd.), and, at DSP

Communications, Inc., I was involved with echo cancellation, noise cancellation, and the creation of state-of-the-art chipsets for cellular telephones.

15. I have been working with, and have written programs for, personal computers since around 1986. Initially on DOS, and later on Windows 3.1, Windows 96, Windows 98, and Windows 2000, as well as Apple computers. Much of my programming concerned digital signal processing, particularly speech, audio and image coding, and communications.

16. From 2001, I also have been providing expert technology services in patent disputes. My biography and experience relevant to my work in these matters is more fully detailed in Attachment A.

17. I have been the co-recipient, with Dr. Allen Gersho, of the Ericsson-Nokia Best Paper Award for the paper: "Enhanced Waveform Interpolative Coding at 4 kbps," *IEEE Workshop on Speech Coding*, Finland, 1999. The IEEE Workshop on Speech Coding is an exclusive workshop for speech-coding researchers from around the world.

18. I have authored and co-authored approximately eight journal publications, in addition to conference proceedings, technical articles, technical papers, book chapters, and technical presentations concerning a broad array of signal processing technology. I have also developed and taught many courses

related to digital signal processing and signal processing systems. These courses have included introductory level and advanced courses.

19. I have several international patents related to the field of audio signal enhancement, including U.S. Patent Nos. 6,614,370; 7,643,996; 7,010,482; and 7,584,095.

20. I am being compensated at the rate of $525 per hour for my work in connection with this matter. The compensation is not dependent in any way on the contents of this report, the substance of any further opinions or testimony that I may provide, or the ultimate outcome of this matter.

## II. LIST OF DOCUMENTS CONSIDERED IN FORMULATING MY OPINIONS

21. In formulating my opinions, I have reviewed and considered all of the following documents:

| EXHIBIT NO. | DESCRIPTION |
|---|---|
| 1001 | U.S. Patent No. 7,260,521 |
| 1002 | File history of U.S. Patent No. 7,260,521 |
| 1003 | Petition For Inter Partes Review Of U.S. Patent No. 7,260,521 |
| 1004 | ITU-T Recommendation G.729, "Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CSACELP)," March 1996 |
| 1005 | ITU-T Recommendation G.728, "Coding of Speech at 16 kbit/s Using Low-Delay Code Excited Linear Prediction," September 1992 |
| 1006 | U.S. Patent No. 5,797,119 ("Ozawa") |
| 1007 | U.S. Patent No. 5,235,669 ("Ordentlich") |
| 1008 | U.S. Patent No. 5,444,816 ("Adoul") |
| 1009 | Atal, B. and Remde, J., "A New Model of LPC Extension for Producing Natural-Sounding Speech at Low Bit Rates," IEEE, |

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| | pp. 614-617 (1982). |
| 1010 | Spanias, A., "Speech Coding: A Tutorial Review," Proceedings of the IEEE, 82(10):1539-82 (1994) |
| 1011 | Federal Coding Standard 1016 (February 14, 1991) |
| 1012 | Schroeder, M. and Atal, B, "Code-Excited Linear Prediction (CELP): High Quality Speech at Very Low Bit Rates," IEEE, pp. 937-940 (1985) |
| 1013 | Honkanen, T., et al., "Enhanced Full Rate Speech Codec for IS-136 Digital Cellular System," IEEE, pp. 731-734 (1997) |
| 1014 | ETSI - GSM Digital Cellular Telecommunications System; Enhanced Full Rate (EFR) Speech Transcoding (GSM 06.60) (1996) |
| 1015 | ITU G.722.2 Series G: Transmission Systems and Media, Digital Systems and Networks (2002) |
| 1016 | Gersho, A. and Cuperman, V., "Vector Quantization: A Pattern-Matching Technique for Speech Coding," IEEE, pp. 15-21 (1983) |
| 1017 | Chan, W., et al., "Enhanced Multistage Vector Quantization by Joint Codebook Design," IEEE Transactions on Acoustics, Speech, and Signal Processing, 40(11): 1693-1697 (1992) |
| 1018 | Singhai, S. and A. Bishnu, "Improving Performance of Multi-Pulse LPC Coders at Low Bit Rates," IEEE, pp. 1.3.1-1.3.4 (1984) |
| 1019 | Ramachandran, R. and Kabal, R., "Pitch Prediction Filters in Speech Coding," IEEE Transactions on Acoustics, Speech, and Signal Processing, 37(4):467-478 (1989) |
| 1020 | Kleijn, W. and Ketchum, R., "Improved Speech Quality and Efficient Vector Quantization in SELP," IEEE, pp. 155-158 (1988) |
| 1021 | Marques, J, et al., "Improved Pitch Predication With Factional Delays in CELP Coding," Filters in Speech Coding," IEEE, 665-668 (1990) |
| 1022 | Chen, J., and Gersho, A., "Adaptive Postfiltering for Quality Enhancement of Coded Speech," IEEE Transactions on Acoustics, Speech, and Signal Processing, 3(1), pp. 59-71 (1995) |
| 1023 | ITU G.722 – General Aspects of Digital Transmission Systems Terminal Equipments (1988) |
| 1024 | Cheng, Y., et al., "Statistical Recovery of Wideband Speech From Narrowband Speech," IEEE Transactions on Acoustics, |

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|---|---|
| | Speech, and Signal Processing, 2(4):544-548 (1994) ("Cheng") |
| 1025 | Ordentlich, E. and Shoham, Y., "Low-Delay Code-Excited Linear-Predictive Coding of Wideband Speech at 32 KBPS," IEEE, pp. 9-12 (1991) |
| 1026 | Avendano, C., et al., "Beyond Nyquist: Towards the Recovery of Broad-Bandwidth Speech from Narrow-Bandwidth Speech," Eurospeech, 95 (1995) |
| 1027 | J. Schnitzler, A 13.0 KBIT/S Wideband Speech Codec Based on SBACELP, IEEE, pp. 157-161 (1998) |
| 1028 | U.S. Patent 5,293,449 ("Tzeng") |
| 1029 | U.S. Patent 4,868,867 ("Davidson") |
| 1030 | Joint Claim Construction and Prehearing Statement, Saint Lawrence v.<br>LG Electronics, Inc. - 14cv1055 (EDTX) dated July 17, 2015 |
| 1031 | Kroon, P., "Regular-Pulse Excitation – A Novel Approach to Effective and Efficient Multipulse Coding of Speech," IEEE Transactions on Acoustics, Speech, and Signal Processing, 34(5):1054-1063 (1986) |
| 1032 | Atal, B. and Schroeder, M., "Predictive Coding of Speech Signals and Subjective Error Coding," IEEE Transactions on Acoustics, Speech, and Signal Processing, 27(3):247-254 (1979) |
| 1033 | Rabiner, L., "On the Use of Autocorrelation Analysis for Pitch Detection," IEEE Transactions on Acoustics, Speech, and Signal Processing, 25(1):24-33 (1977) |
| 1034 | Infringement Contentions |
| 1035 | Oppenheim and Schafer, Discrete-Time Signal Processing, pp. 17-33 (1989) |
| 1036 | Declaration of Michael T. Johnson, PhD, Case: IPR2016-00705 Patent 7,260,521 |
| 2005 | "Discrete-Time Signal Processing," by Alan V. Oppenheim, Ronald W. Schafer |
| 2001 | P. Mermelstein, "G.722, A new CCITT Coding Standard for Digital Transmission of Wideband Audio Signals," IEEE Comm. Mag., Vol. 26, No. 1, pp. 8-15, Jan. 1988. |
| 2002 | Fuemmeler et. al, "Techniques for the Regeneration of Wideband Speech from Narrowband Speech," EURASIP Journal on Applied Signal Processing 2001:0, 1-9 (Sep. 2001). |

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| 2003 | C.H. Ritz et. al., "Lossless Wideband Speech Coding," 10th Australian Int'l. Conference on Speech Science & Technology, p. 249 (Dec. 2004). |
| 2005 | "Discrete-Time Signal Processing," by Alan V. Oppenheim, Ronald W. Schafer |

22. I have reviewed and am familiar with the response to Petition submitted on behalf pf Patent Owner for covered patent review submitted with this Declaration and I agree with the technical analysis that underlies the positions set forth in the response to Petition.

23. I have reviewed and am familiar with the Petition for covered patent submitted by Petitioner, and I disagree with some of it, and with its conclusions. I have reviewed and considered Dr. Johnson Report submitted on behalf of Petitioner, and I disagree with some of it, and with its conclusions.

24. I may consider additional documents as they become available or other that are necessary to form my opinions. I reserve the right to revise, supplement, or amend my opinions based on new information and on my continuing analysis.

## III. TECHNICAL BACKGROUND AND STATE OF THE ART AT THE TIME OF THE ALLEGED INVENTION

25. Speech signal is generated by air flow emanating from the lungs, passed through the vocal chords which may vibrate to create quasi-periodic air

pulses, which are then passed through the vocal tracts (throat cavity, mouth cavity and nasal cavity) and radiates outside though the lips and/or the nose. When the vocal chords vibrate the generated quasi-periodic speech is called "voiced", and when they are at rest the generated speech is more noise like and called "unvoiced". A classical model for speech production ("Digital Processing of Speech Signals, L. R. Rabiner and R. W. Schafer, Prentice Hall, 1978) is illustrated below. This early model includes a switch for selecting between the "voiced" component generated by the periodic source and "unvoiced" component generated by the noise source, each multiplied by an appropriate gain, to form the excitation signal, which is then passed through a system modeling the vocal tract and uses time varying parameters, and the resulted signal is finally passed though the lips radiation model to form the generated speech. The fundamental period of the vocal chords vibration is known as the pitch period. The vocal tract is typically modeled by time varying filter implemented by set of parameters (or coefficients).

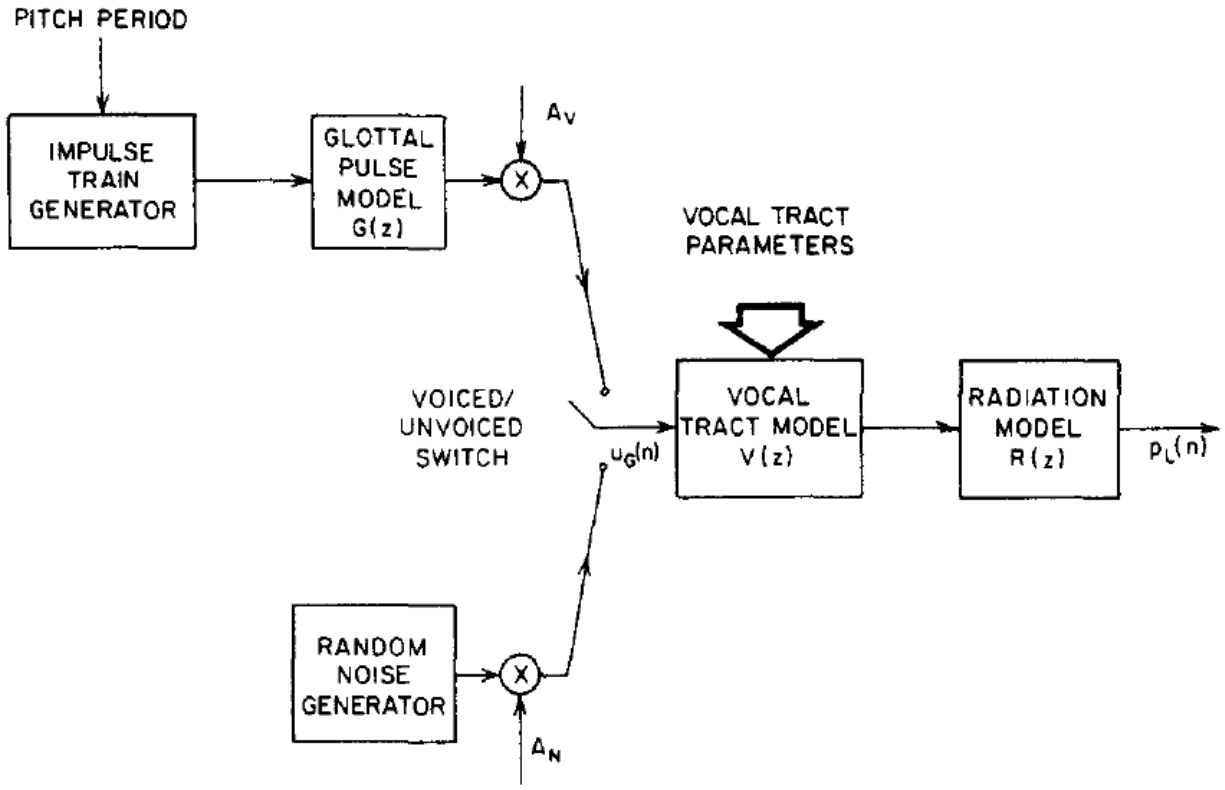


**Fig. 3.50 General discrete-time model for speech production.**

26. Speech signal is much richer than simply voiced and unvoiced, for example buzzing sounds like "z" and "v" include both periodic and noise components, and therefore in later models, such as the one below, the switch was replaced by a mixer that combined timed varying mixture. For simplicity let's consider the following simplified diagram. For producing good quality speech, the vocal tract parameters, representing the vocal tract resonance, are sufficiently updated every 20-30 ms, while the excitation components are sufficiently updated every 5-7 ms. It can be shown that the vocal tract effect adds correlation among

neighboring samples, which is often treated as short-term prediction or redundancy. The vocal chords quasi periodicity is often treated as long-term prediction or redundancy.

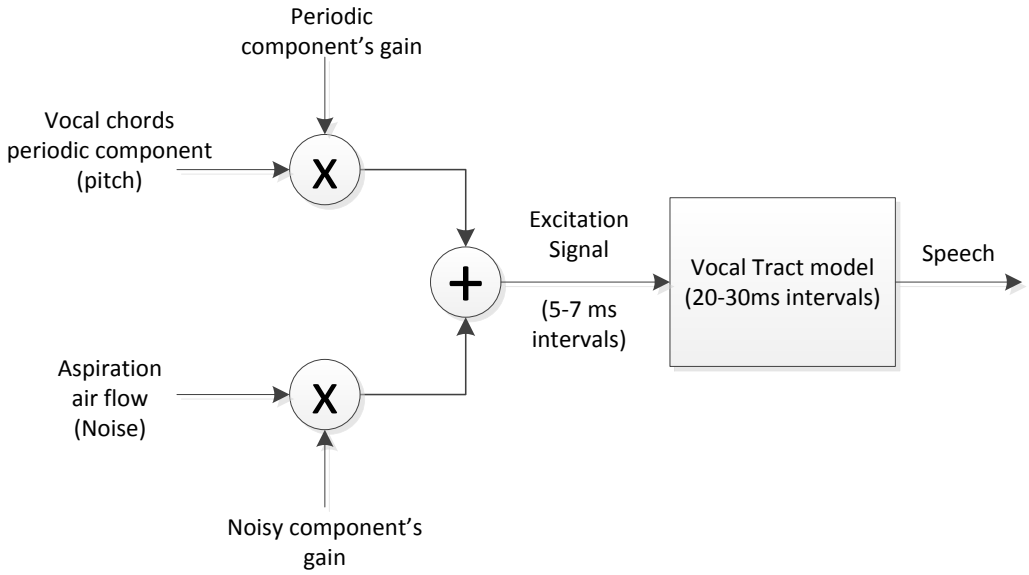


**Figure 1.** Block diagram of simplified speech production model

27. Speech coding is used for representing speech signals in a compact form for transmission or storage. The compression is typically achieved by first capturing and removing the redundancies in forms of short-term prediction and long-term prediction that exist in the speech signal, and forming a residual signal having a much smaller dynamic range and energy. Then the residual signal is quantized, a process of representing it by a finite number of bits and limited resolution. The difference between the reduced resolution signal and the original signal is considered quantization noise (or quantization error). The number of bits

used for the quantization governs the amount of quantization noise, or alternatively the overall quality of the coded speech.

### A. Speech coding and Linear Prediction Coding (LPC) analysis

28. The linear Predictive Coding (LPC) is a well known analysis for calculating the short term prediction coefficients that are used to model the vocal tract. It is typically applied directly to the input speech signal and LPCs are encoded (or quantized) once per frame at of 20-30 ms, which is an adequate interval based on the vocal tract change speed.

29. The LPC are used to capture and remove the short-term prediction from the input speech, where the LPC coefficients are used as weighted sum of the most recent 10 samples to predict the present sample, which is essentially a filtering operation. The short-term predicted sample is subtracted from the input sample to generate the prediction error also known as the residual signal sample having a much smaller dynamic range and energy than the input speech. For this reason the residual signal is more desirable for quantization than the speech, since thanks to its smaller energy and dynamic range, its quantization error (or noise) is also smaller.

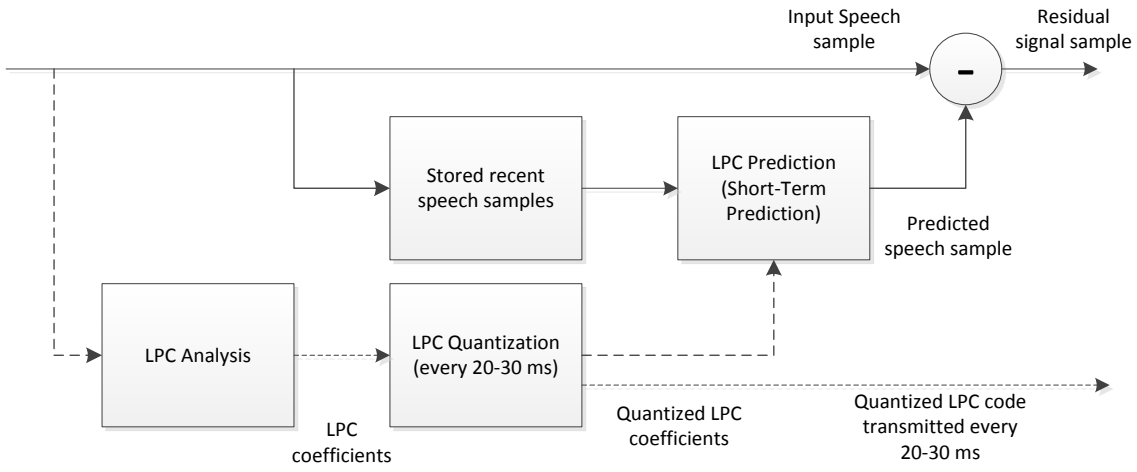


**Figure 2.** Block diagram of open-loop short-term prediction removal example using LPC analysis

30. The removal of the short-term prediction from the input speech to generate the residual signal is essentially a decomposition of the speech signal into its slowly varying vocal tract characteristics (i.e. the LPC coefficients) and its rapidly changing excitation characteristic (i.e. the residual signal). Such a decomposition allows for applying slower LPC quantization at a frame intervals of typically 20-30 ms, and a faster residual encoding at a subframe rate of typically 5-7 ms. Examples of speech signals and their corresponding residual signals (scaled up in the figure for better view) is illustrated at Id Fig 8.5. As can be seen, the residual signals exhibits less short-term correlation among neighboring samples, looks more like noise and comb pulses, but still exhibits long-term correlation between neighboring pitch periods.

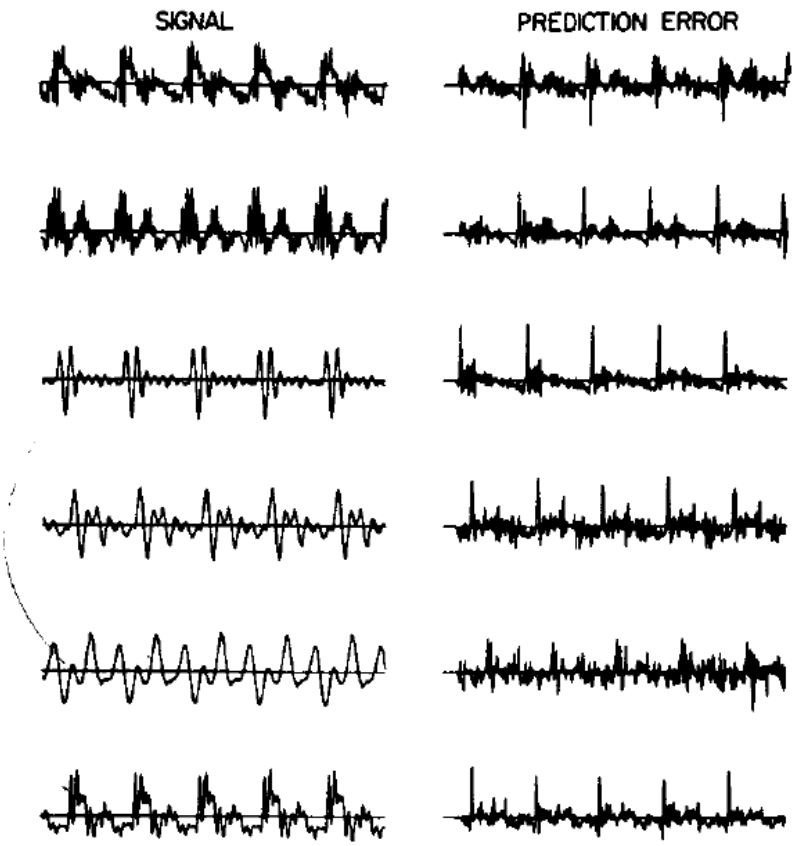


**Fig. 8.5** Examples of signal (differentiated) and prediction error for vowels (*i*, *e*, *a*, *o*, *u*, *y*). (After Strube [14].)

31. In the frequency domain, the vocal tract shapes the speech spectral envelope, while the excitation component shapes the fine spectral structure about that envelope. The figure below illustrates from top to bottom time domain sampled speech segment, the corresponding residual signal that contains much less short-term prediction and energy, the speech signal's (log) spectrum and the corresponding spectral envelope represented by the LPC, and the residual signal's (log) spectrum which is flat since the spectral envelope captured by the LPC was removed from the speech. As illustrated, the speech was decomposed to the

spectral envelope - represented by the LPC, and the fine spectral variations around the spectral envelope - represented by the residual signal. As illustrated, during voiced speech segments, the residual signal exhibits comb-like harmonic structure at the fundamental frequency's (pitch) multiple frequencies (harmonics). During buzzy sounds like "v" and "z", and during transitions, the harmonic structure appears more prominent in the lower frequency range, and the higher frequency range (e.g. 4-5kHz in the figure below) becomes less structured and more noise like. The LPC only captures the spectral envelope, and does not capture the harmonic structure. As illustrated, by removing the LPC spectral envelope from the speech in (c), the residual signal (d) has a flat spectrum that exhibits harmonic structure (during voiced speech).

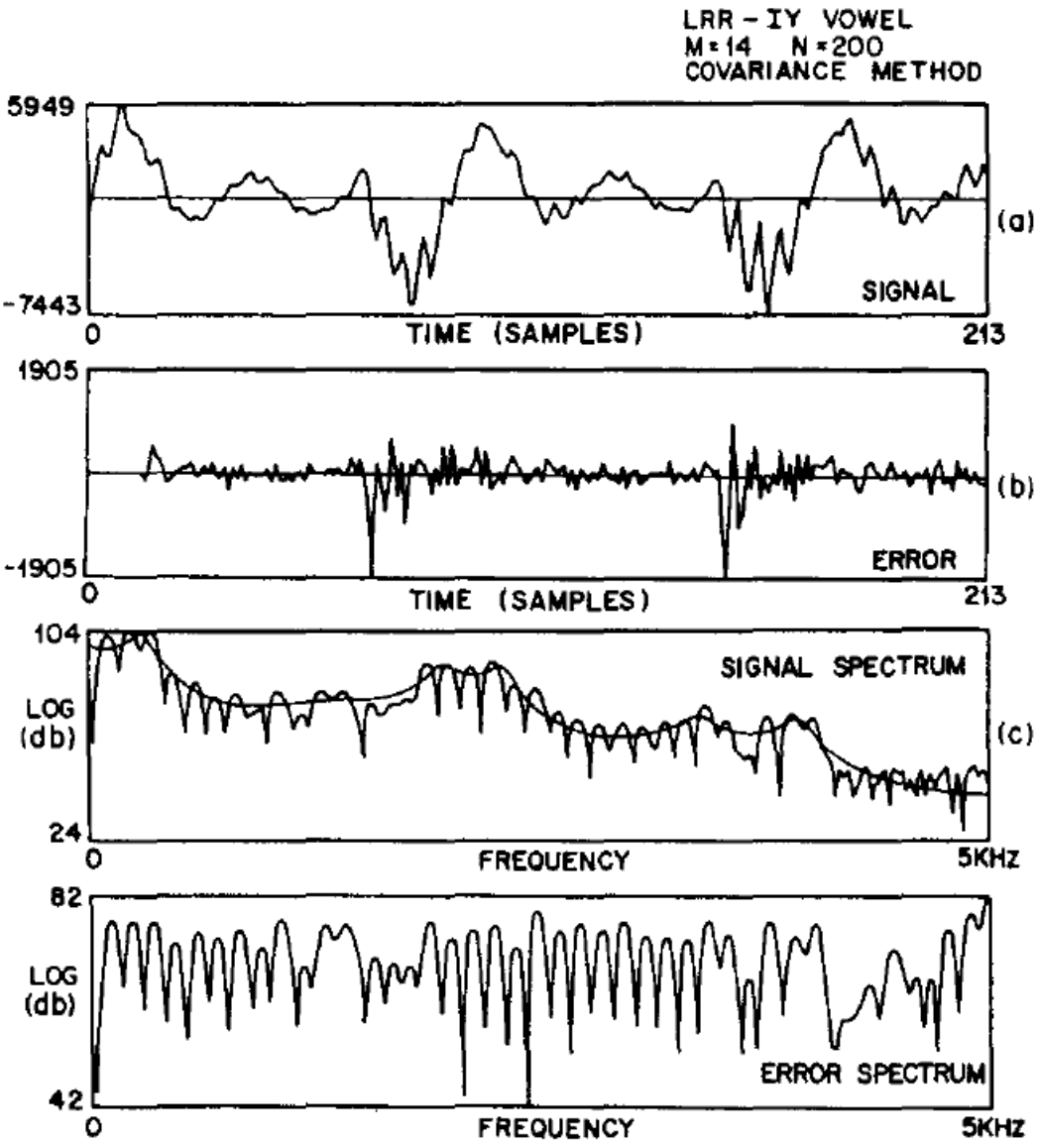


**Fig. 8.6** Typical signals and spectra for LPC covariance method for a male speaker. (After Rabiner et al. [16].)

32. This also illustrates why some statements in the '521 petition are completely wrong, for example in p.7 "using harmonic structure represented the encoded LPC analysis…" Dr. Johnson is also flatly wrong about that, and the following statements in his reports reflect much confusion (Ex. 1035 regarding '802 patent and Ex. 1036 regarding '521 patent, p.23 section.55) "such frequency

extension can be accomplished in a simple manner by simply using the harmonic structure represented by the encoded LPC analysis…" Person of ordinary skills in the art knows that the LPC analysis does not represent the harmonic structure and cannot be used for that, rather than the residual signal does so. The LPC represents merely the spectral envelope, its resonances (formants) and valleys.

33. Interestingly, the Ozawa is directed toward a heuristic preselection election of the long term prediction path for modeling the harmonic structure for the frame, based on the calculated LPC, and given that preselected single path the long-term prediction is determined for each subframe within the frame.

**B. Long Term Prediction (Pitch Prediction)**

34. As explained above, the vocal chords "pitch" periodicity generates long term correlation between neighboring speech periods. For encoding purposes, such long-term correlation can be modeled and captured by means of long-term prediction. The parameters that are typically used to encode the long-term prediction are the pitch period which may be integer or fractional, and the gain used to predict the present period from the past period. Such parameters may be viewed as describing a comb filter, where the comb pulses are spaced by the pitch period, and their exponential progression is given by the gain. A simple scheme is illustrated below. In this scheme the past residual signal is stored and used to generate a predicted residual sample which is then subtracted from the input

residual signal to form a long-term prediction error often referred to as the remnant signal.

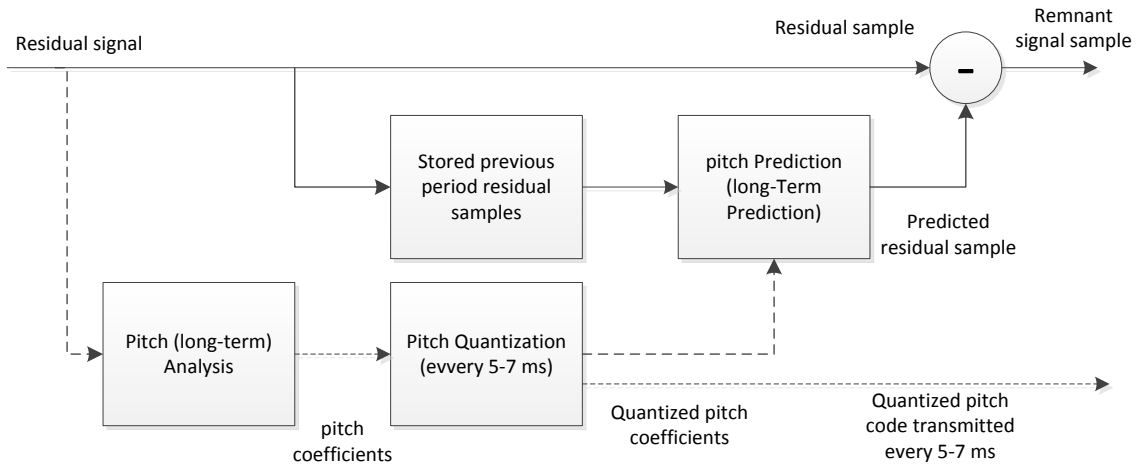


**Figure 3.** Block diagram of simplified open-loop long-term prediction removal example

35. The remnant signal is noise like signal that exhibits almost no short term and no long-term correlation, and its energy and dynamic range are much smaller than the input speech. It is very hard to encode since it has no particular structure, and it is hard to tell how much its quantization error would affect the overall generated speech quality at the decoder. In other words, the simple encoding scheme described so far was performed in open-loop and did not really consider the speech signal generated by the decoder. We will discuss below better coding scheme that operate in closed-loop and consider the synthesized speech.

### C. Quantization

36. Quantization is the process of reducing signal resolution, and representing the reduced resolution signal by some code to be stored and transmitted. It can be done on a sample by sample basis, using table of quantization levels, where the quantization level is selected by the encoder (scalar quantizer) using for example nearest-neighbor criterion. Quantization can also be performed on a set of values often called vector (e.g. set of LPC parameters or set of consecutive signal samples), where quantized vectors are selected from multi-dimensional tables also referred to as codebooks, again the quantized vector is selected by the encoder (vector quantizer, VQ) using for example nearest-neighbor criterion. For a given number of bits, vector quantizer produces much better quality than scalar quantizer, at the cost of increased computational complexity.

37. Codebooks (and quantizers) are usually computed by training data gathered during system operation, such that the generated trained codebook minimizes some distortion measure over all the training data generated by the system. Once the system is changed, e.g. its operation order is changed, computation elements are added, removed or modified, the codebook would no longer operate optimally, and new training process is needed to be performed to achieve adequate performance. In other words, systems having quantizers and/or

codebooks may operate unpredictably once they are changed or combined with other systems.

38. Similarly, person of ordinary skills in the art knows that one cannot combine systems that were designed to operate at one sampling rate with systems that were designed to operate at a different sampling rate. Combining such system typically yields unpredictable result.

### D. Code Excited Linear Prediction (CELP)

39. Code Excited Linear Prediction (CELP) has become the most widely used coding scheme in the past 30 years. Its core idea is based on closed-loop quantization of the residual signal and remnant signal, also known as analysis-by-synthesis (AbS). In this paradigm, the excitation and the remnant signals are selected such that the resulted output speech best matches the input speech, as explained below.

40. A simplified block diagram of the encoder of such a system is illustrated in Figure 4. In the encoder, the LPC are computed and quantized for the short-term correlation filter in open-loop manner once every 20-30 ms frame. This is done the standard LPC analysis that aims to maximize the speech short-term self-prediction, as explained above.

41. Given the LPC precomputed in open-loop for the whole frame, the encoder then starts closed-loop processing of 3-5 subframes, each of 5-7 ms, where

the distortion criterion is waveform matching between the input speech and the synthesized speech, or (theoretically) equivalently minimizing the energy of weighted error between the input speech and the synthesized speech. For convenience let's consider a theoretically equivalent simplified diagram illustrated in Figure 4, although actual CELP system is implemented differently, and is more complicated.

42. The encoder consists of two components, namely the analyzer, and the synthesizer. Since the synthesizer is actually a replica of the decoder, it is also known as the *local decoder*. In the synthesizer, the excitation is generated as the output of a long-term correlation filter, which generates the pitch periodicity. The excitation is then passed through the short-term correlation, also known as the LP synthesis filter, to produce the output speech. This filter models the vocal tract transfer function. It emphasizes certain frequency regions (formants), and de-emphasizes others (e.g., valleys). The output speech is then subtracted from the input speech to generate the objective error signal. In our theoretically equivalent and simplified diagram, the objective error is passed through the perceptual-weighting filter, which models the spectral masking effect of the auditory system, to produce the subjective error [3], [5]. The weighted mean squared error (WMSE) is computed as the sum of the squares of the samples of this perceptually weighted error signal. The analyzer selects the excitation parameters to minimize the

WMSE. This procedure is known both as *closed-loop* (CL) quantization and AbS. The AbS is most effective when the excitation parameters are selected to jointly minimize the WMSE. The selected parameters are transmitted to the decoder that duplicates the same synthesis structure to reconstruct the speech.

43. Various LP based AbS speech coders have been created. They are distinguished largely by the excitation quantization scheme used. Among them are multi-pulse excitation (MPE) coding [4], regular-pulse excitation (RPE) coding [8], and code-excited linear predictive (CELP) coding [5]. The Algebraic CELP (ACELP) is a form of multi-pulse excitation, where the pulses are restricted to certain algebraic structure and hence does not require storage of the codebook entries [14].

44. The short term correlation, or the LPC synthesis filter is commonly computed via LP analysis [1], [2]. When the LP analysis is applied to the input speech, it is categorized as *forward adaptive* LP analysis. For constraints such as delay limitations, the LP analysis may sometimes be applied to the reconstructed speech. Such a scheme is known as *backward adaptive* LP analysis. This process is exactly repeated at the decoder, and therefore in backward adaptive LP based systems the LPC parameters need not be transmitted. The low-delay CELP (LD-CELP) is such a backward adaptive LP AbS coder [6].

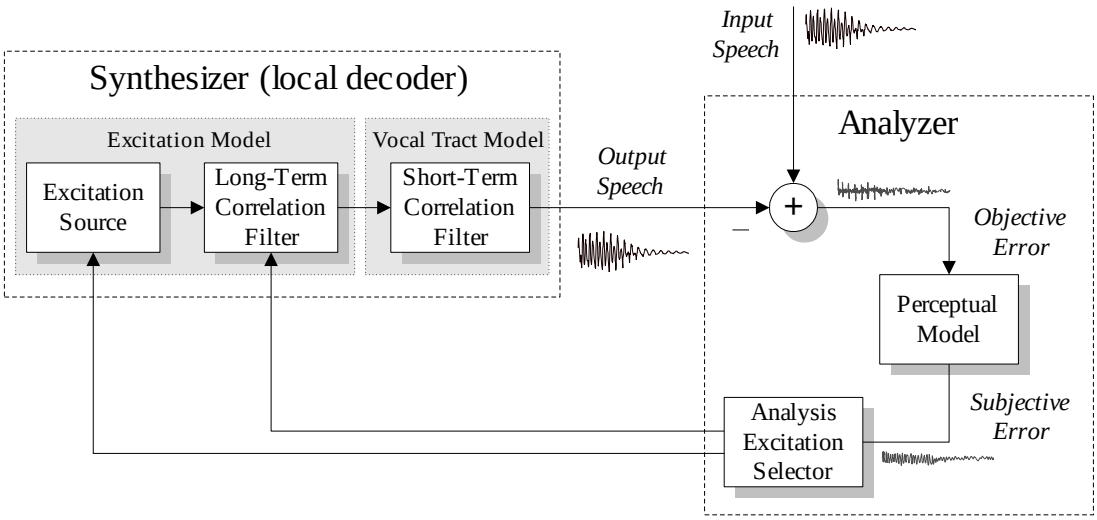


**Figure 4.** Block diagram of perception-based linear predictive analysis-by-synthesis speech coder, theoretically equivalent simplified diagram.

45. The concept of AbS was generalized such the original speech signal is first modified to produce a perceptually equivalent signal that can be coded more efficiently by the AbS system [9], [10], [11]. The speech signal is modified such that its pitch contour is linear over the coded frame, and therefore fewer bits are needed to describe the pitch. This concept was applied to a CELP scheme, and known as the *relaxed-excitation CELP* (RCELP) [10], [11] which was standardized in 1996 by the Telecommunications Industry Association (TIA) for CDMA wireless networks [7]. More about LP AbS coding may be found in [12].

46. In CELP coders, the open-loop LPC analysis is applied to the input speech once per frame to minimize the input speech autocorrelation. The computed LPC are then quantized, and used to decompose the speech and generate the residual signal for the frame. Such a decomposition allows for applying slower

LPC quantization at a frame intervals of typically 20-30 ms, and a faster residual encoding at a subframe rate of typically 5-7 ms. Then the residual signal is quantized subframe by subframe in closed-loop using AbS by matching each subframe of synthesized speech to the input speech using WMSE distortion measure.

47. The quantized residual signal is usually referred to as the excitation signal, since it is used as excitation to LPC based synthesis (short-term correlation) filter that generates reconstructed speech. The excitation is generated at each subframe using excitation source and long-term correlation filter. The excitation source is often referred to as fixed codebook (FCB), and its entries are typically subframe-long noise-like vectors. The long-term correlation filter is typically a comb filter, characterized by the lag (pitch period) and gain. It uses the recent excitation signal to predict the present excitation signal, therefore it is also referred to as adaptive codebook (ACB). The FCB and ACB parameters may be jointly determined for highest results, but for computational simplicity they are often determined sequentially, where the ACB parameters are determined first (or at least limited to a small subset), and then then the FCB parameters are determined. As explained, while the LPC are quantized for the whole frame (in open loop), the ACB and FCB parameters are determined and quantized for each subframe (in closed-loop).

48. In the context of the '521 patent, it is important to note the following differences between the LPC computation and the excitation computation:

| **Aspect** | **LPC computation** | **Excitation Computation** |
|---|---|---|
| Model | **Vocal track** (throat, mouth, nose cavities) | **Vocal chords** and/or lungs' air-flow excitation |
| Signal applied to | **Input speech** in CELP, and **output speech** in LD-CELP (backward adaptive) | **Residual** signal and **remnant** signal. |
| Segment applied to | Computed **once per frame** for windowed input **speech** frame (CELP) or output speech frame (LD-CELP) plus some lookahead (CELP) and lookback. | **Residual subframe**, computed sequentially, a few times per frame. |
| Method of calculation | **Open-loop** LPC analysis | **Closed-loop** Analysis-by-synthesis |
| Captures | **Spectral envelope** | **Fine spectral structure** (pitch harmonics and/or |

| | | |
|---|---|---|
| | | noise like) |
| Used for in encoder | **Speech decomposition** into slowly evolving spectral envelope (vocal tract), and periodic and/or rapidly evolving excitation | Output waveform matching by employing analysis-by-synthesis. |
| Objective | Maximizing **auto-prediction** over each frame of input speech (CELP) or output speech (LD-CELP) from its past. | **Waveform matching** of each subframe between synthesized speech and input speech, subject to perceptual weighting. |
| Distortion measure | Computed once per **frame**, to maximize the input speech's (CELP) or output speech's (LD-CELP) short-term auto-prediction gain, or equivalently to minimize input speech's (CELP) or | Computed once per **subframe** to minimize the Weighted Mean Squared Error (WMSE) between input speech subframe and synthesized speech subframe. Adaptive Codebook (ACB) |

| | | |
|---|---|---|
| | output speech's (LD-CELP) short-term prediction error. | component exploits the long-term prediction between past excitation pitch period and the present period. |

### E. Perceptual Weighting

49. Perceptual weighting is aimed to hide the quantization noise "under the rug" and make it less audible or perceivable. It is typically performed using the unquantized LPC coefficients, calculated for each subframe, in a filter that redistributes the white quantization noise such that it becomes higher under the speech's spectral formants (resonance), and lower in the spectral valleys. By doing so, the speech masks the higher noise regions more effectively, and less noise is present in spectral regions that the speech masking is less effective. Overall there is higher noise energy, and yet it is better masked by the speech and hence becomes less perceivable to the listener. When perceptual weighting is used, the quantization uses weighted mean-squared error (WMSE) also referred to as subjective error criteria, as opposed to objective error criteria such as Signal-to-noise (SNR) which minimizes the overall error.

### F. Long term pitch prediction and using adaptive codebook

In CELP coding the long-term (pitch) prediction is captured by means of Adaptive Codebook (ACB), which uses the recent excitation signal as adaptive codebook. Commonly the lag (essentially the period) and the gain parameters are determined for each subframe in closed-loop fashion to minimize the (perceptually weighted) reconstructed signal. For computational simplicity the lag is often first roughly estimated in open-loop before the actual closed-loop optimization of both the lag and gain. The ACB may be jointly optimized with the FCB, but it is typically quantized first, and then the FCB is quantized. The ACB is quantized by comparing the synthesized subframe to the output speech (in the WMSE sense) and selecting the lag (or period) and gain combination that yield the best matched subframe.

50. Since the comb filter has uniform harmonic over the whole spectrum while speech signal often exhibits harmonic structure over only portion of the spectrum, the ‘521 is directed to selection of different filter paths at each subframe to allow different spectral range of harmonic structure for each subframe, as part of the ACB parameters. The ‘521 illustrated the minimal energy selector 309, having input from multiple paths – each has a different filter, and determines the parameters b, T, and j that minimize the error energy. Evidently, along with the

common parameters of lag (T) and gain (b), the selected filter path (j) for the subframe is also transmitted, as illustrated in '521 Figure 1 and Figure 3.

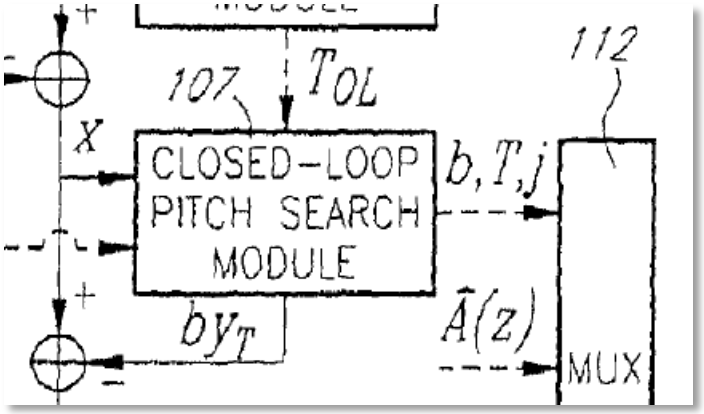


**Figure 5.** Block diagram '521 Figure 1, the selected filter path (j) for the subframe is also transmitted

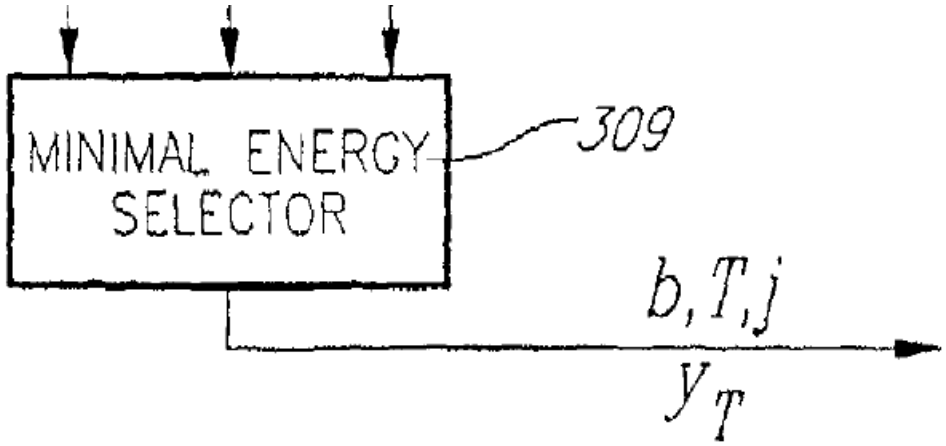


**Figure 6.** Block diagram '521 Figure 3, the selected filter path (j) for the subframe is also computed based on minimizing the error energy over multiple paths.

## G. CELP Decoding

51. A simplified block diagram of the CELP decoder is illustrated in the following figure. The received bits are decoded into three groups of parameters, namely the LPC for the whole frame (which is typically interpolated for each subframe), and FCB and ACB parameters for each subframe. The decoder generates the excitation for each subframe using the FCB, it then introduces long term periodicity using the ACB. Each excitation subframe is then passed through the LPC synthesis filter which introduces the short-term correlation, and generates the output speech. Typically the LPC are also used for perceptual postfilter, which further smoothens the speech, emphasizes the resonance, and decreases the noise in the spectral valleys, and generates a more pleasant quality speech.

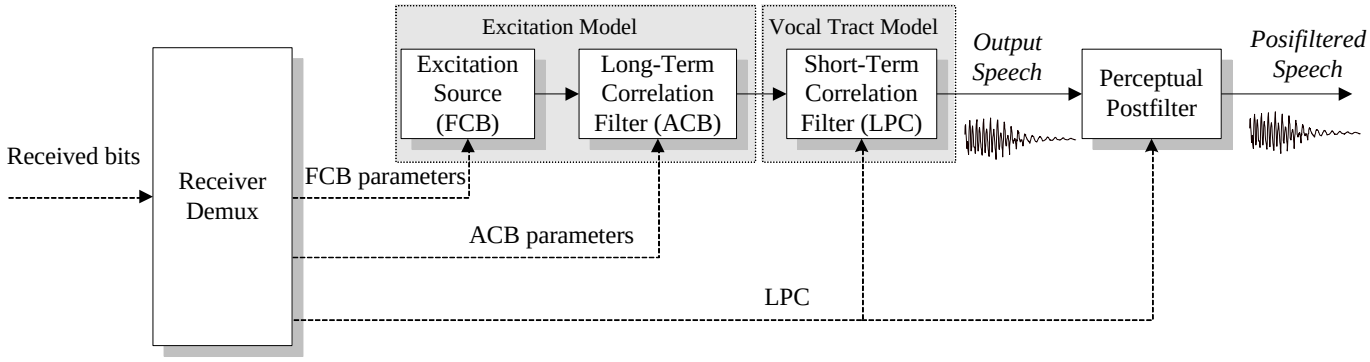


**Figure 7.** A simplified block diagram of the CELP decoder.

52. The ACB parameters commonly includes lag and gain, and internal excitation feedback to adapt its memory, i.e. its adaptive codebook. Its memory is adapted after each subframe synthesis.

53. The '521 patent is directed to CELP decoder that employs for every subframe additional received parameter (j) for selecting low-pass filter 202 for the pitch codebook 201, along with the lag (T) and the gain (b) 226, as illustrated in Figure 1.

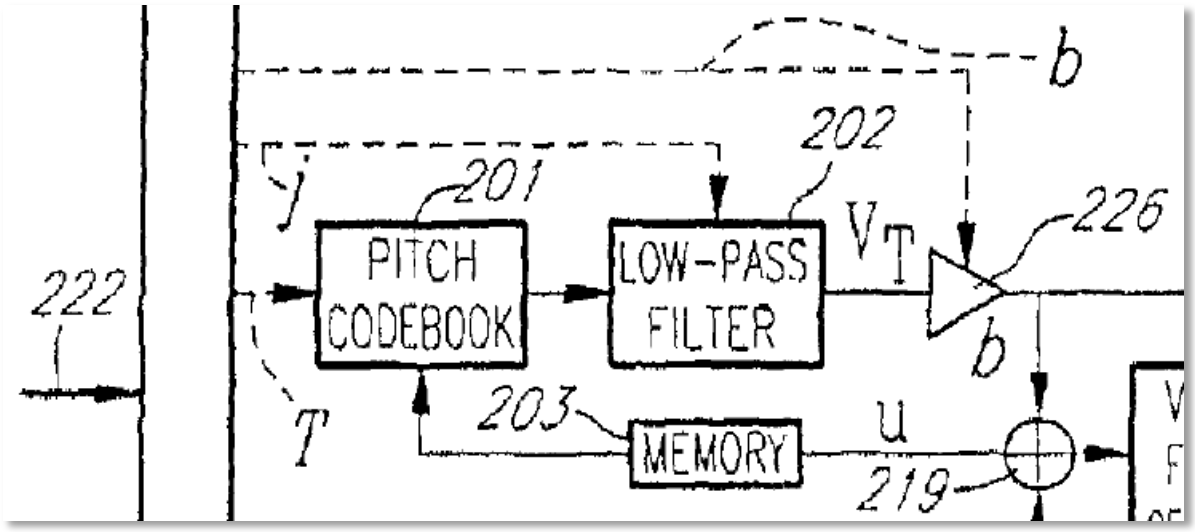


**Figure 8.** Block diagram '521 Figure 1, the selected filter path (j) for the subframe is also received

## H. Speech Bandwidth extension

54. Speech bandwidth extension is intended to generate crispier speech. While for voiced speech, much of its energy and information is located at the lower frequency range, and for unvoiced speech much of much of its energy and

information is located at the higher frequency band. There are also mixed sounds and transitory sounds. The art of speech bandwidth extension includes injecting adequate level of noise to the high frequency range, so that it add crispiness to some speech segment while not turning other segments too noisy and harsh. It has to be done such that it improves the overall quality. Different scheme may result in different quality, and artifacts, depending on the particular implementation.

## I. Speech Quality

55. Speech quality is usually measured either by bit-exactness when comparing one system to another system with known quality, or by extensive subjective tests where many listeners listen to the generated speech and grade it. Such tests are for example Mean Opinion Score (MOS) test. It is well known in the art of speech coding and speech enhancement that some material may sound good to some people and not so good to other people. It is also well known to skilled in the art that some artifacts may be more important in some languages, and other artifacts may be more annoying in other languages. For these reasons and more, extensive subjective tests are so much more important and significant than one person's or a few people's opinion.

**J. Finite precision Considerations**

56. The well known textbook "Discrete-Time Signal Processing," by Alan V. Oppenheim, Ronald W. Schafer that was partially brought in Exhibit 1035, discusses the differences between digital signal processing systems and filters that appear to be theoretically equivalent in pages that were not included in the exhibit.[1] For example, on page 346 the textbook explains: "Theoretically, the order of implementation does not affect the overall system function. However, as we will see, when a difference equation is implemented with finite-precision arithmetic, there can be a significant difference between two systems that are theoretically equivalent." There are additional similar statements on pages 357 and 365.

The block diagrams of Figures 6.3 and 6.4 have several important differences. In Figure 6.3, the zeros of $H(z)$, represented by $H_1(z)$, are implemented first, followed by the poles, represented by $H_2(z)$. In Figure 6.4, the poles are implemented first, followed by the zeros. Theoretically, the order of implementation does not affect the overall system function. However, as we will see, when a difference equation is implemented with finite-precision arithmetic, there can be a significant difference between two systems that are theoretically equivalent. Another important point concerns the number of delay elements in the two systems. As drawn, the systems in Figures 6.3 and 6.4 each have a total of $(N + M)$ delay elements. However, the block diagram of Figure 6.4 can be redrawn by noting that exactly the same signal, $w[n]$, is stored in the two chains of delay elements in the figure. Consequently, the two can be collapsed into one chain, as indicated in Figure 6.5.

57. In Section 6.6 starting on page 370 the textbook discusses in more details "Overview of finite-precision numerical effects" and explains that: "One motivation for considering alternatives to the simple direct-form structures is that

[1] A complete version of this book is in Exhibit 2002.

different structures that are theoretically equivalent may behave differently when implemented with finite numerical precision."

**6.6 OVERVIEW OF FINITE-PRECISION NUMERICAL EFFECTS**

We have seen that a particular linear time-invariant discrete-time system can be implemented by a variety of computational structures. One motivation for considering alternatives to the simple direct-form structures is that different structures that are theoretically equivalent may behave differently when implemented with finite numerical precision. In this section, we give a brief introduction to the major numerical problems that arise in implementing discrete-time systems. A more detailed analysis of these finite-word-length effects is given in Sections 6.7–6.9.

58. The inventors even discuss some of these issues in the '521 patent, for example in col.2 l.20-25: "Wideband signals exhibit a much wider dynamic range compared to telephone-band signals, which results in precision problems when a fixed-point implementation of the algorithm is required (which is essential in wireless applications)." Also in col.8, 65-67: "It also reduces the dynamic range of the input speech signal, which renders it more suitable for fixed-point implementation." Furthermore, the '521 patent uses pre-emphasis filter 103 and de-emphasis filter 207, two filters that theoretically cancel the effect of each other, and the theoretical equivalent of them is having no filter at all. Yet, for real world considerations using finite-precision processors, the inventors included the use of such filters in the invention, in spite the additional computations and memory cost which are typically translated into more expensive product having larger form

factor and having shorter battery life. This is yet another evidence for the importance of system elements and their computational order when it comes to real-world implementation considerations.

59. Person of ordinary skill in the art at the time of the patent understood these differences, and knew that the order of computations does matter when it comes to actual implementation in real world environment using processors that always have finite-precision architecture.

## IV. PERSON OF ORDINARY SKILL IN THE ART

60. I am informed that it is permissible to determine the level of ordinary skill in the art from a review of relevant prior art references. For purposes of this Declaration, I am relying on the October 27, 1998 priority date listed on the face of the ‘521 Patent for my analysis of the level of ordinary skill in the art appropriate to the ‘521 Patent.

61. In my view, the level of ordinary skill relevant to the ‘521 Patent is evident from a review of the prior art references cited above.

62. In particular, it is my opinion that a person of ordinary skill in the art for the ‘521 Patent would have had at least a bachelor of science degree in Electrical Engineering, with knowledge and experience in speech coding and speech enhancement and specifically with Code Excited Linear Prediction (CELP)

coding and alike, and at least two years of experience working with speech coding technologies. That includes deep knowledge and good understanding of speech coder design issues, speech coding artifacts, LPC or short-term prediction, long-term prediction, open-loop analysis, closed loop analysis by synthesis, frame-based processing, and subframe-based processing, optimal search and/or quantization, suboptimal search and/or quantization, auditory system properties, understanding of subjective quality versus objective measures, spectral properties of speech signal, implementation issues in finite precision (e.g. in fixed-point implementation), and error sensitivity analysis. I possess at least these qualifications, and I have considered the issues herein from the perspective of a person of ordinary skill in the art.

## V. OVERVIEW OF THE '521 PATENT

63. The '521 Patent relates to a speech decoding system that includes high frequency content recovery method and device.

64.

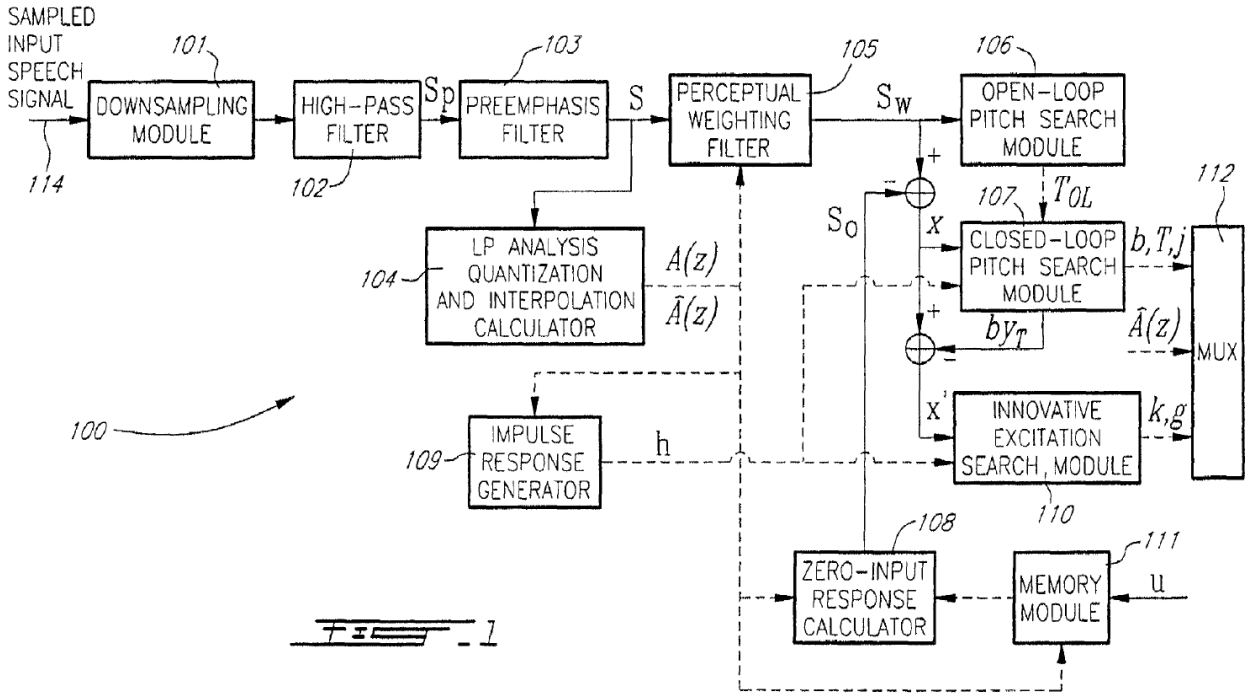


Fig. 1

**Figure 9.** '521 Patent Figure 1

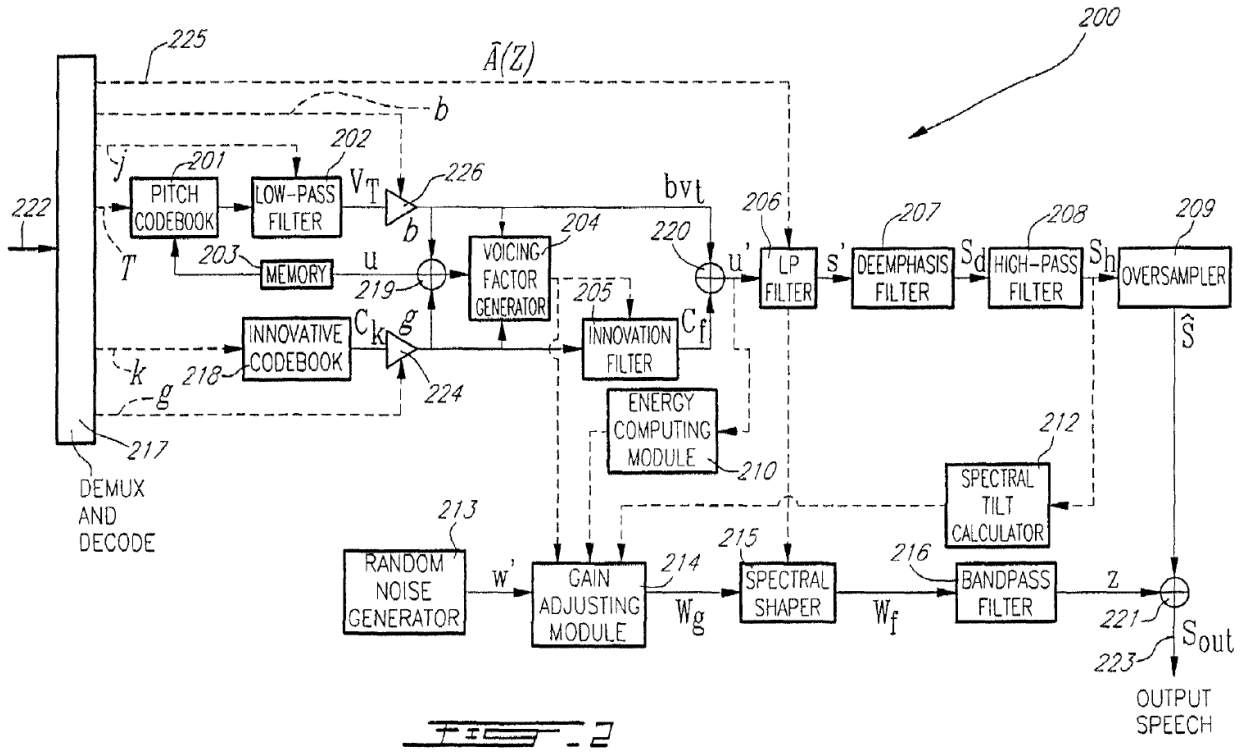


Fig. 2

**Figure 10.** '521 Patent Figure 2

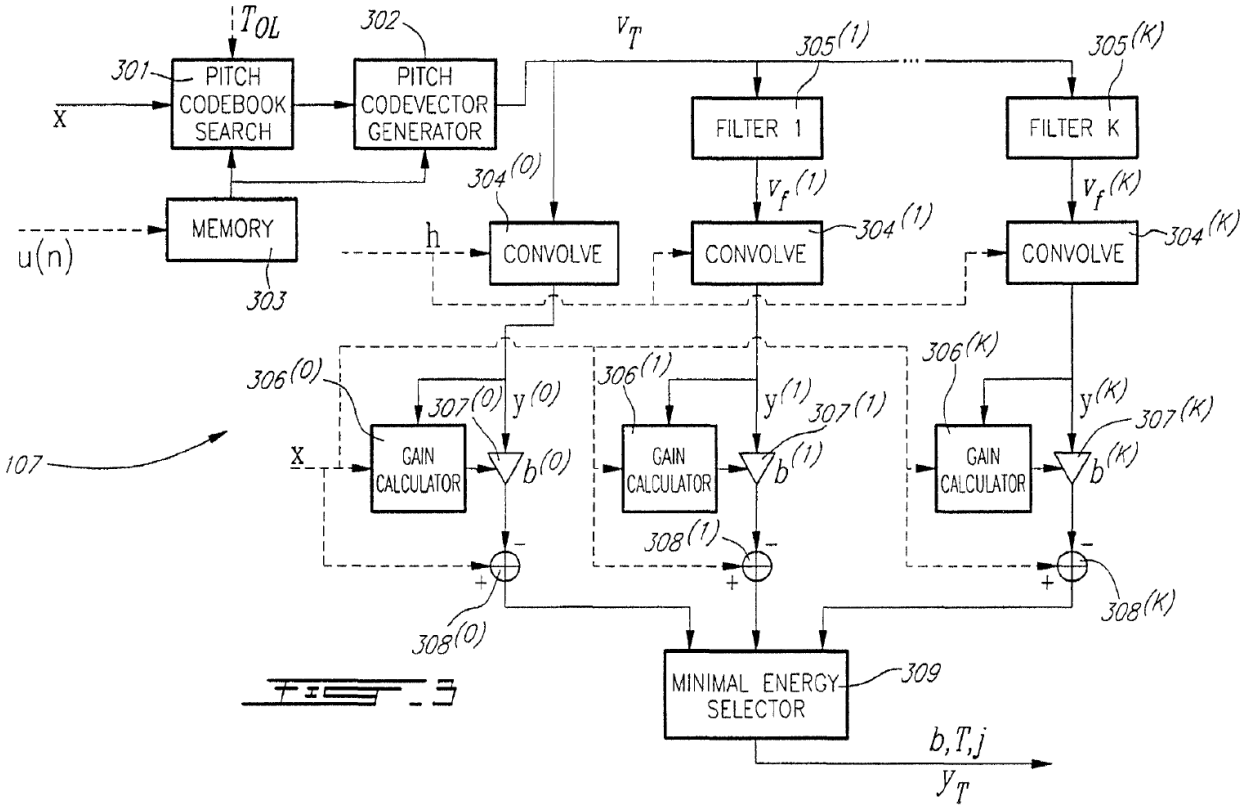


**Figure 11.** '521 Patent Figure 3

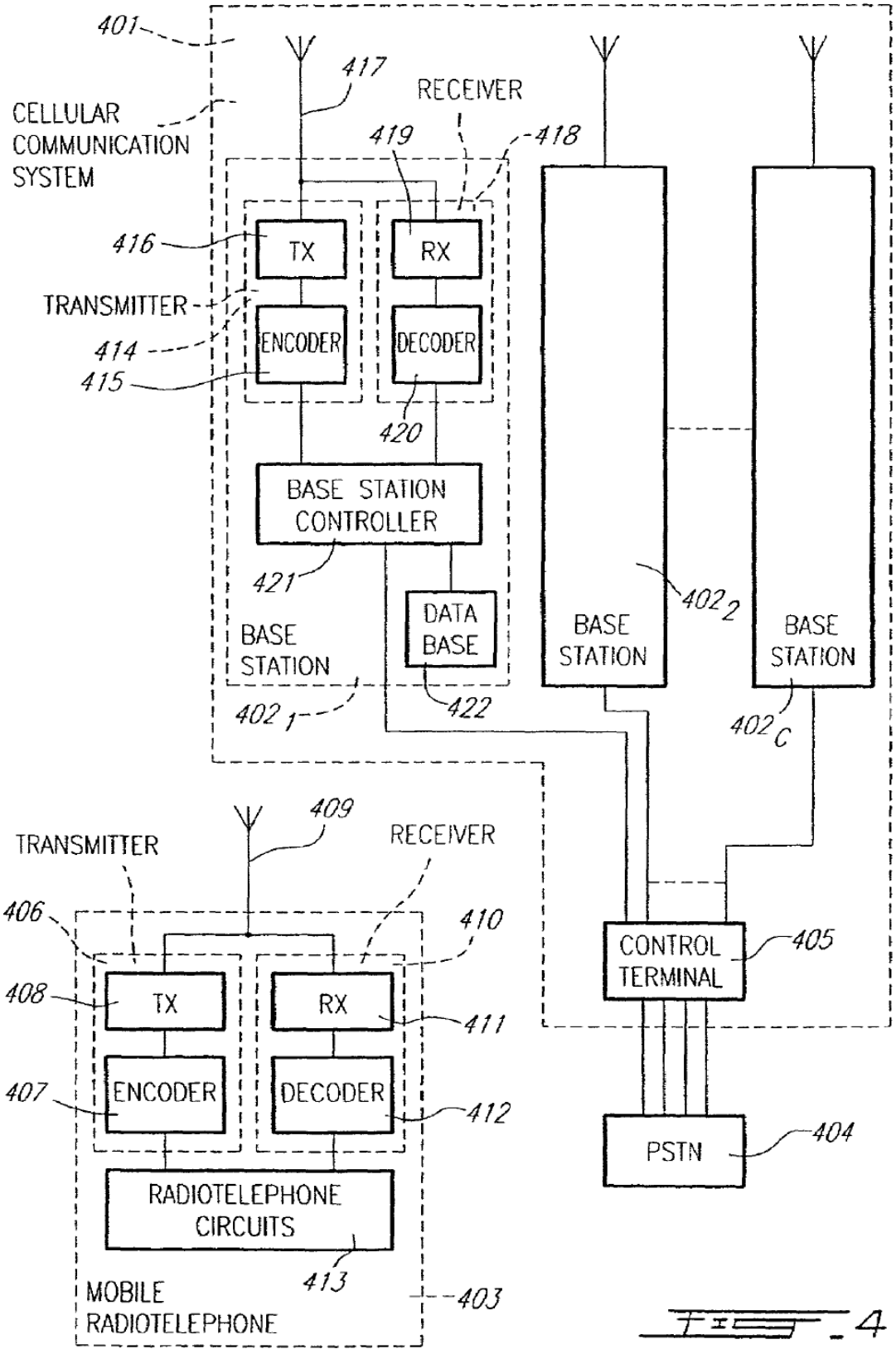


**Figure 12.** '521 Patent Figure 4

## VI. THE CLAIMS OF THE '521 PATENT

65. I understand that Petitioner challenges the validity of Claims 1-3, 8-11, 16, 25-27, 32-35, 40, 49, 50, 52, and 53 of the '521 Patent, where claims 1, 9, and 25 are independent claims and the remaining are dependent claims. Claim 9 is similar to claim 1, but focuses on "the improvement a high-frequency content recovering." Claim 25 is similar to claim 1 in the context of "A mobile transmitter/receiver unit". Independent Claim 1 is exemplary and reads:

> 1. A decoder for producing a synthesized wideband signal, comprising:
>
> a) a signal fragmenting device for receiving an encoded version of a wide band signal previously down-sampled during encoding and extracting from said encoded wide band signal version at least pitch codebook parameters, innovative codebook parameters, and linear prediction filter coefficients;
>
> b) a pitch code book responsive to said pitch codebook parameters for producing a pitch codevector;
>
> c) an innovative codebook responsive to said innovative codebook parameters for producing an innovative codevector;
>
> d) a combiner circuit for combining said pitch codevector and said innovative codevector to thereby produce an excitation signal;
>
> e) a signal synthesis device including a linear prediction filter for filtering said excitation signal in relation to said linear prediction filter coefficients to thereby produce a synthesized wideband signal,

and an oversampler responsive to said synthesized wideband signal for producing an over-sampled signal version of the synthesized wideband signal; and

f) a high-frequency content recovering device comprising:

i) a random noise generator for producing a noise sequence having a given spectrum;

ii) a spectral shaping unit for shaping the spectrum of the noise sequence in relation to linear prediction filter coefficients related to said down-sampled wideband signal; and

iii) a signal injection circuit for injecting said spectrally-shaped noise sequence in said over-sampled synthesized signal version to thereby produce said full-spectrum synthesized wideband signal.

## VII. LEGAL STANDARDS

66. I have not been asked to offer an opinion on the law; however, I understand that I am obliged to follow existing law. I have therefore been asked to apply the following legal principles to my analysis, and I have done so.

### A. Requirements of a Method and System Patent

67. I have been informed that "[i]n determining whether a patent is for a technological invention, the following will be considered on a case-by-case basis: whether the claimed subject matter as a whole recites a technological feature that is novel and unobvious over the prior art; and solves a technical problem using a technical solution."

**B. Obviousness**

68. I understand that even if a patent is not anticipated, it is still invalid if the differences between the claimed subject matter and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the pertinent art.

69. I understand that a person having ordinary skill in the art (*i.e.*, a PHOSITA) is a hypothetical person who is presumed to have known the relevant art at the time of the invention was made. I understand that the condition "at the time the invention was made" is imposed to rule out impermissible hindsight. I also understand that an expert is to analyze the prior art from the perspective of a person of ordinary skill in the art and not simply to provide his/her own personal conclusions.

70. I also understand that an obviousness determination includes several factual inquiries, including (1) determining the scope and content of the prior art; (2) ascertaining the differences between the claimed invention and the prior art; (3) resolving the level of ordinary skill in the pertinent art; and (4) taking into consideration any secondary indicia of non-obviousness.

71. I am informed that secondary indicia of non-obviousness may include (1) a long felt but unsolved need that was satisfied by the claimed invention; (2) commercial success attributable to the claimed invention; (3) unexpected results

achieved by the claimed invention; praise by experts of the claimed invention with factual support; (4) taking of licenses under the patent by others for reasons related to the alleged unobviousness of the claimed invention; and (5) evidence that competitors in the marketplace are copying the invention instead of using the prior art. I also understand that there must be a relationship, or nexus, between any such secondary indicia and the claimed invention, i.e., objective evidence of nonobviousness must be attributable to the claimed invention. I further understand that near simultaneous invention by two or more equally talented inventors working independently, may or may not be an indication of obviousness when considered in light of all the circumstances.

72. I understand a conclusion of obviousness can be based on a combination of multiple prior art references. I understand that exemplary rationales that may support a conclusion of obviousness include:

> (A) Combining prior art elements according to known methods to yield predictable results;
> (B) Simple substitution of one known element for another to obtain predictable results;
> (C) Use of known technique to improve similar devices (methods, or products) in the same way;
> (D) Applying a known technique to a known device (method, or product) ready for improvement to yield predictable results;

(E) "Obvious to try" – choosing from a finite number of identified, predictable solutions, with a reasonable expectation of success;
(F) Known work in one field of endeavor may prompt variations of it for use in either the same field or a different one based on design incentives or other market forces if the variations are predictable to one of ordinary skill in the art;
(G) Some teaching, suggestion, or motivation in the prior art that would have led one of ordinary skill to modify the prior art reference or to combine prior art reference teachings to arrive at the claimed invention.

73. I further understand that obviousness analysis recognizes that market demand, rather than scientific literature, often drives design trends.

74. I understand that if a technique has been used to improve one device, and a person of ordinary skill in the art would recognize that it would improve similar devices in the same way, using the technique is obvious unless its actual application is beyond his or her skill.

75. I also understand that practical and common sense considerations should guide a proper obviousness analysis, because familiar items may have obvious uses beyond their primary purposes. I further understand that applying common sense does not require a "specific hint or suggestion in a particular reference," only a reasoned explanation that avoids conclusory generalizations.

76. I understand that a person of ordinary skill in the art addressing a problem will often be able to fit the teachings of multiple publications together like

pieces of a puzzle. In this regard, I understand that an obviousness analysis may take into account the inferences and creative steps that a person of ordinary skill in the art would employ.

77. I understand a particular combination may be proven obvious merely by showing that it was obvious to try the combination. For example, when there is a design need or market pressure to solve a problem and/or there is a finite number of identified, predictable solutions, with a reasonable expectation of success, a person of ordinary skill in the art has good reason to pursue the known options within his or her technical grasp. I understand that if this leads to anticipated success, it is likely the result not of innovation but of ordinary skill and common sense.

78. I understand that the combination of familiar elements according to known methods is likely obvious when it does no more than yield predictable results. When work is known in one field of endeavor, it may prompt variations of that work for use in either the same field or a different one, based on design incentives and other market forces. If a technique has been used to improve one device, and a person of ordinary skill in the art can implement a predictable variation, it is likely unpatentable.

79. It is further my understanding that to be proper for use in an obviousness analysis, a reference must be analogous art to the claimed invention.

Accordingly, I understand that under the correct analysis, any need or problem known in the field of endeavor at the time of the invention and addressed by the claimed invention can provide a reason for combining the elements in the manner claimed.

80. I understand a claim can be obvious in light of a single reference, without the need to combine references, if the elements of the claim that are not found explicitly or inherently in the reference can be supplied by the common sense of one of skill in the art. For example, combining two embodiments disclosed adjacent to each other in a prior art patent does not require a leap of inventiveness.

81. I understand that a claimed invention may be obvious if it involves merely simple substitution of one known element for another to obtain predictable results. I understand further that the prior art need not be like two puzzle pieces that must fit together perfectly. For example, a claimed invention may be found obvious if a person of ordinary skill in the art would view rearrangement as an obvious matter of design choice.

82. Finally, I have been informed and understand that obviousness analysis requires a comparison of the properly construed claim language to the prior art on a limitation-by-limitation basis.

## VIII. CLAIM CONSTRUCTION

83. I understand that claims in a covered patent must be given their broadest reasonable interpretations in light of the specification. I understand that claim terms in a district court are generally given their ordinary and customary meaning.

The claim constructions set forth in the Patent Owner's Preliminary Response for certain claim limitations are consistent with how one of ordinary skill in the art at the time of the effective filing date of the '521 patent would have understood those limitations.

84. My opinions in this Declaration do not change whether the specific constructions identified are adopted or whether no express constructions are adopted. My opinions do not change if it is determined that the claim construction standard applicable in district court cases should apply to this review.

## IX. SUMMARY OF PRIOR ART TO THE '521 PATENT ALLEGED IN THIS PETITION (CASE IPR2016-00705)

### A. U.S. Patent 5,797,119 ("Ozawa")

85. Ozawa describes a narrowband speech encoder that uses the LPC ("spectral parameters") calculated for each frame (having 5 subframes), and uses them as input to mode-classifier 27 which determines the frame's mode, essentially in open-loop and based on the frame's LPC. See *Ozawa 2:52-60* (emphasis added):

> "The speech encoder includes a framing circuit 10 where a digital input speech signal is segmented into blocks or "frames" of 40-millisecond duration. for example. The output of framing circuit 10 is supplied to a subframing circuit 11 where the speech samples of each **frame are subdivided into a plurality of subblocks, or "subframes" of 8-millisecond duration**, for example."

The mode is selected once per frame out of four values based on the open-loop short-term prediction error ("minimum error power") for the frame. See also *Id 4:1-16* (emphasis added):

> "A mode classifier 27 is connected to the spectral parameter calculator 12 to evaluate the linear predictive coefficients $\alpha_{ij}$. Specifically. it calculates K-parameters that represent the **spectral envelope of the speech samples of every five subframes**. Using the K-parameters, the mode classifier determines an accumulated predictive error power for **every five subframes** compares it with three threshold values and The the error power is classified into one of four distinct categories, or modes, with a mode 0 corresponding to the minimum error power and a mode 3 corresponding to the maximum. A **mode index** is supplied from the mode classifier to the pitch synthesizer 17 and to the multiplexer 25 for transmission or storage."

Once mode index is determined per frame, the encoding continues for each subframe correspondingly to the predetermined frame mode. For mode 0 the ACB is disabled for all the subframes in the frame, and for modes 1-3 the subframe closed-loop (ACB) operation operates for each of the subframe within the frame, based on the frame's predetermined mode. See Id 4:17-19:

> "In order to determine the pitch period at subframe intervals. the pitch synthesizer 17 is provided with a known adaptive codebook."

See Id 4:24-25:

> "When the encoder is in mode 0. the pitch synthesizer produces no pitch index."

86. I understand that Petitioner alleges the ACB disabling as multiple path as claimed by the invention, I strongly disagree with that at least for the following reasons:

87. (a) The mode and hence the alleged multiple path is predetermined before the subframe closed-loop processing starts, and hence is not affected by the ACB operation and/or its used distortion, furthermore there is no switching among different paths over the 5 subframes of the frame

88. (b) Disabling the ACB can not form multiple path within the ACB, and no multiple distortion paths are used for the ACB optimization, and no multiple distortions are selected from multiple paths by a selector within the ACB are disclosed in Ozawa. There is no selection of the best path within the subframe's ACB closed-loop long-term prediction optimization (when active), since a single path is preselected for the whole frame in open-loop by the LPC short-term prediction error.

89. Therefore 1.a, Ozawa does not disclose at least two signal paths associated with respective sets of pitch codebook parameters representative of the digitized input audio data, since there is preselected single path, and there are no multiple path available during the subframe ACB search. Either a filter path is

preselected for all the subframes within the frame, or no ACB is selected, but not both. Furthermore when no ACB is selected there are no pitch codebook parameters for all the subframes within the frame.

90. Unlike commonly used in CELP, Ozawa discloses first optimizing the FCB, using equation (5) and subsequently subtracting the FCB contribution by subtractor 16.

$$D_c = \sum_{n=c}^{N-1} |Xw(n) - \beta \cdot q(n) - g_1 c_{1jz}(n) * h_w(n) - g_2 c_{2kz}(n) * h_w(n)|^2 \quad (5)$$

The remaining signal is then used for the ACB search. Ozawa does not disclose how the ACB pitch period is calculated except *Id 4.22-23* "Pitch synthesizer 17 supplies a pitch index indicating the pitch period to an excitation vector candidate selector 18." Ozawa does not disclose "pitch prediction errors calculated" as part of the pitch index calculation (by selector 18).

91. I disagree with the statement made in the petition p.17: " One of skill in the art would understand that calculating distortion is the same as calculating prediction error" made in the context of equation (6). I understand this statement by Petitioner as referred to the long-term (pitch) prediction error the ACB. Ozawa's equation (6) is a joint ACB and FCB quantization distortion that is aimed for search for optimized parameters of both the ACB's and FCB's gains (Id 5.31-33: "where, $\beta'_k$ is the gain of k-the adaptive code vector, and $g'_{1k}$ and $g'_{2k}$ are the

gains of k-th excitation code vectors of the first and second excitation vector stages. respectively". Determining the ACB's and FCB's gains by gain calculator 23 cannot by viewed as calculating "pitch prediction error"

$$D_k = \sum_{n=c}^{N-1} [X'w(n) - \beta'_k \cdot q(n) - g'_{1k}c_{1jz}(n)*h_w(n) - g'_{2k}c_{2iz}(n)*h_w(n)]^2 \quad (6)$$

92. Ozawa does not disclose how the pitch index q(n) is calculated, except that it is not indexed by the index k, hence it is not disclosed as part of the distortion optimization. Therefore Ozawa's joint quantization distortion (6) is not long-term prediction error. Furthermore, during mode 0 there is no use of ACB at all i.e. no long-term prediction at all, and the above equation is narrowed to FCB quantization only:

$$D_k = \sum_{n=c}^{N-1} [X'w(n) - g'_{1k}c_{1jz}(n)*h_w(n) - g'_{2k}c_{2iz}(n)*h_w(n)]^2 \quad (7)$$

93. For all the above reasons, Ozawa does not disclose calculating pitch prediction error.

94. I disagree with Dr. Johnson analysis Ex.1036 91-102. Dr. Johnson and petitioner mixed up "lowest calculated pitch prediction error" and short-term prediction error that is used for the mode selection. I strongly disagree with the statement (Johnson Ex1036 at 99, and Petition 18-19) "In Ozawa, the mode index

is determined by the mode classifier 27 which acts as a switching mechanism designed to allow the lowest possible prediction error." First there is no switching mechanism disclosed by Ozawa, and no such design is disclosed. Second, the use of the term "prediction error" (which in this case refers at best to the LPC short-term prediction) in the context of pitch prediction error is confusing or mixing-up/misleading. Dr. Johnson and petitioner further mesh this confusion between the frame's LPC short term prediction error minimization and the subframe's closed loop distortion in equations (6) and (7) by "This is then followed by the excitation codebook selector to expressly minimize the distortion, as described above. The net result is to create a system in which the overall prediction error is minimized, and resulting signal quality is maximized."

95. Ozawa does not disclose "pitch prediction errors calculated in said at least two signal paths" first since no "pitch prediction error" is disclosed by Ozawa, and of course that no multiple "pitch prediction errors" are disclosed by Ozawa, and no more than a single path is used per ACB search hence no "said at least two signal paths" are disclosed by Ozawa. Even the distortion is calculated for a single path at any given subframe search. Since there is no "pitch prediction errors calculated in said at least two signal paths", Ozawa does not disclose "a selector for comparing the pitch prediction errors… for choosing the signal path", no signal path is selected based on the long-term prediction errors, for the reasons

above and also since the single path is already predetermined in open-loop based on the frame LPC.

96.

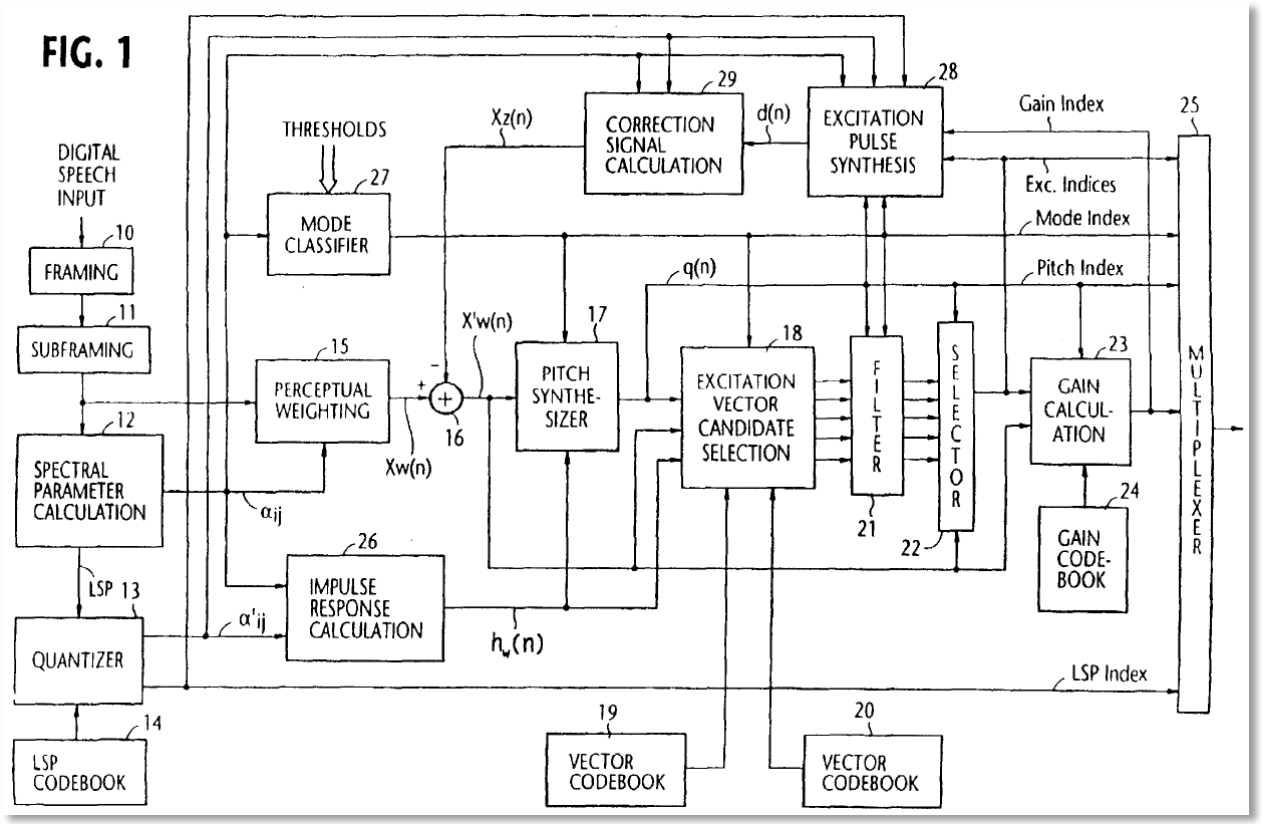


**Figure 13.** Ozawa Figure 1

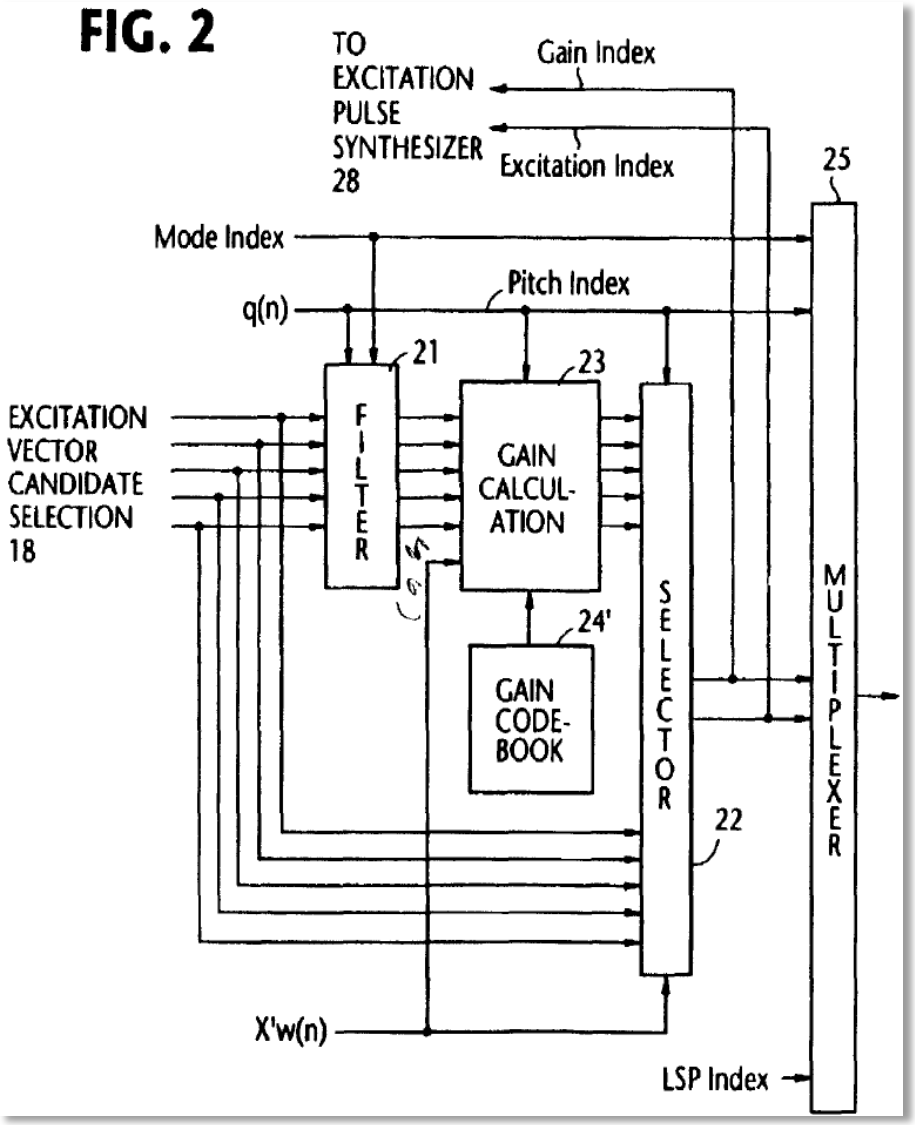


**Figure 14.** Ozawa Figure 2

97. Unlike the ‘521 where at each subframe a different filter path is optimized in closed-loop as part of the ACB long-term prediction and output waveform matching weighted error optimization, in Ozawa a single filter is selected for the whole frame in open-loop based on the frames LPC which are computed based on different distortion of maximizing the input speech

autocorrelation. It is therefore my opinion that Ozawa does not disclose the '521 invention's (Claim 1 element a) "at least two signal paths associated to respective sets of pitch codebook parameters", in Ozawa a single signal path is associated to respective sets of pitch codebook parameters. In Ozawa for all the subframes in the frame there is a single path predetermined in open-loop by the LPC and it is not associated respective sets of pitch codebook parameters that are determined for each subframe in closed loop. In Ozawa the single path is predetermined for the frame in open-loop based on the input's short-term prediction coefficients (LPC) while the '521 is directed to subframe path selection based on the closed loop long-term prediction at the output. In other words, the '521 is directed to inclusion of path selection as one of the ACB optimized parameters, and in Ozawa there is a predetermined single path that is not determined as part of the ACB optimized parameters.

98. Therefore, I strongly disagree with the '521 Petition (Case IPR2016-00705, p.16, l.2) and Dr. Johnson ('521 Ex.1036 section 91 p.35) "Ozawa discloses at least two signal paths associated with respective sets of pitch codebook parameters representative of the digitized input audio data." Similarly, I also strongly disagree with the statement'521 Petition (Case IPR2016-00705, p.16, l.12) and Dr. Johnson ('521 Ex.1036 section 98 p.37): "Ozawa discloses a selector for comparing the pitch prediction errors calculated in the at least two signal paths, for

choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook parameters associated to the chosen signal path." Skilled in the art would understand that a single filter in Ozawa is selected based on the LPC and unrelated to pitch prediction error, and later on the ACB parameters are determined by minimizing the output pitch prediction error – during which a single signal path is associated to respective sets of pitch codebook parameters.

99. The patent is also directed to (claim 1 element a.i) "each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error" while Ozawa there is a preselected single path and the ACB is computed by computing the pitch prediction error only for that preselected path. For a given frame and subframe the same path is used in Ozawa and the ACB is computed by computing the pitch prediction error only for that preselected path – at no time prediction error is calculated for more than a single signal path.

100. The '521 (Claim 1 element b) patent is also directed to "a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error."

### B. Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP) ("G.729")

101. The G.729 discloses narrow band speech coder. The G.729 discloses scheme in which the ACB is first searched and optimized, and subsequently the FCB is searched. The G.729 block diagram illustrates that the ACB gain and vectors (pitch search) are determined in closed loop before the algebraic codebook (FCB) search. This is very different from Ozawa that discloses first searching the FCB vectors and then the ACB vectors, and then joint optimization of the ACB and FCB gains. Furthermore, since the signals used for the ACB in Ozawa and the signals used in G.729 are different, the gain values are expected to be different and may not be suitable. Therefore I disagree with Dr. Johnson and Petitioner, and my opinion is that skilled in the art would understand that the G.729's ACB gain calculator that calculates one ACB gain cannot be combined with Ozawa gain calculator 23 that calculates one ACB gain and two FCB gains. As can be seen, the G.729 ACB gain contains no consideration of FCB gains like in Ozawa:

$$g_p = \frac{\sum_{n=0}^{39} x(n)\, y(n)}{\sum_{n=0}^{39} y(n)\, y(n)} \qquad \text{bounded by } 0 \le g_p \le 1.2 \qquad (43)$$

102. Petitioner stated that skilled in the art would be motivated to combine Ozawa and G.729, but did not explain why would such person be motivated and how specifically that can be done. It is my opinion, that skilled in the art would

have no motivation to combine Ozawa and G.729. These are two different algorithms, that use different signals, operate in different order, and optimizes different distortions and equations.

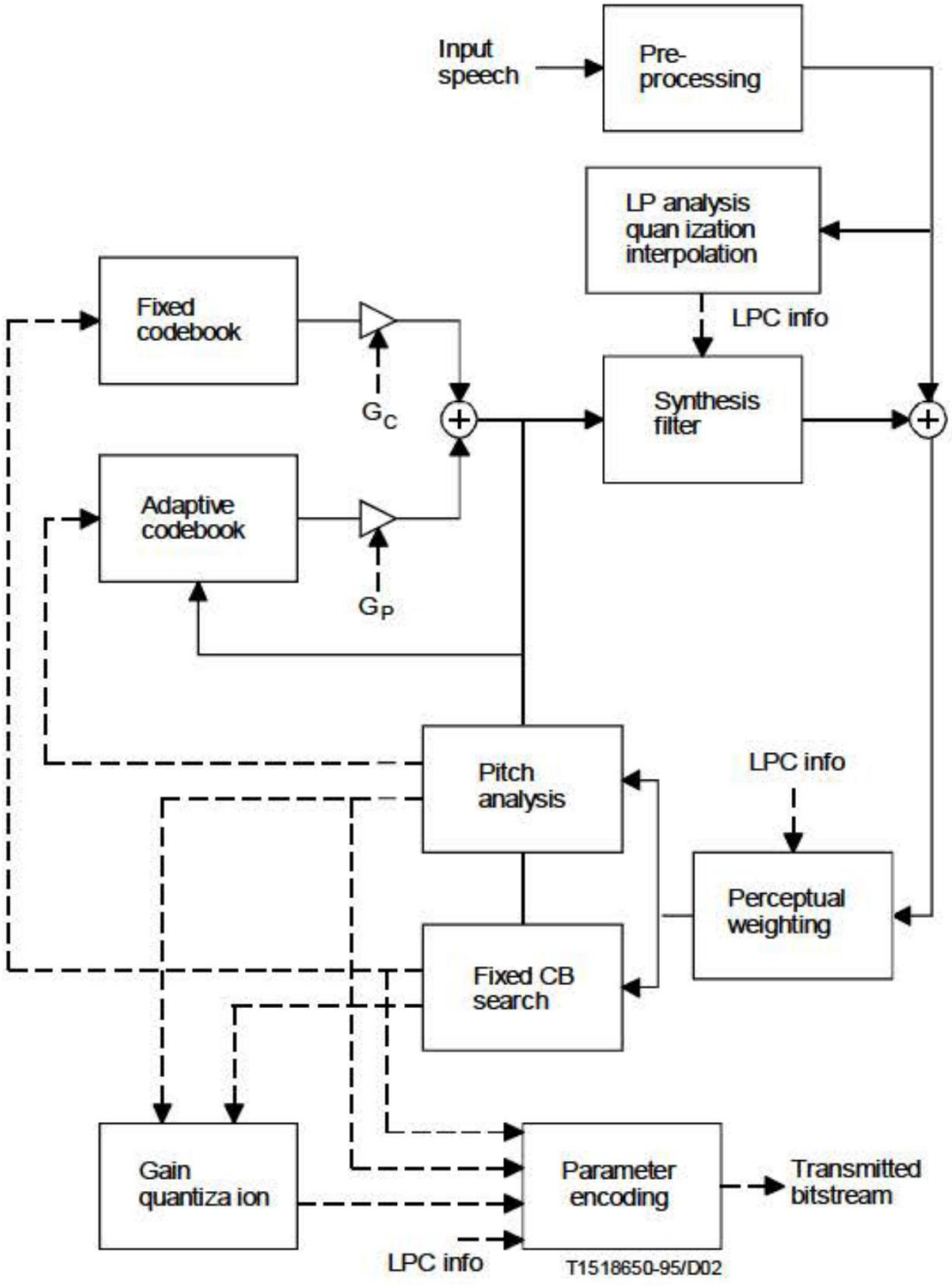


FIGURE 2/G.729

**Encoding principle of the CS-ACELP encoder**

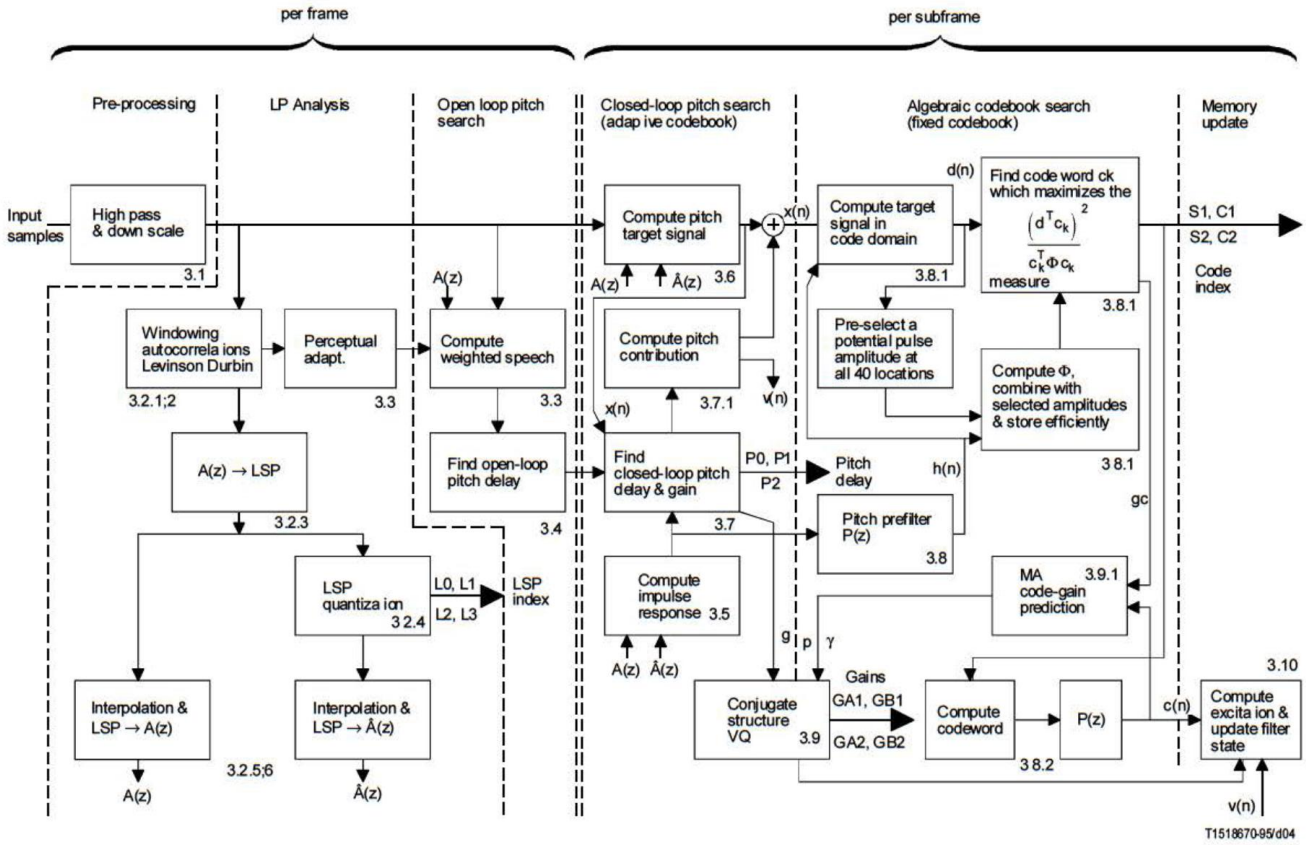


FIGURE 4/G.729
**Signal flow at the CS-ACELP encoder**

### C. U.S. Patent 5,235,669 ("Ordentlich")

103. I am very familiar with Ordentlich and Shoham work, since in 1992 I perform my M.Sc. research at Bell Labs, that was a continuation to this work. Dr. Shoham served as my M.Sc advisor.

104. Ordentlich illustrates in Figure 1 merely a possible conventional CELP on a high level, "the basic structure of conventional CELP", as part of the introduction with references to narrowband CELP systems, and then describes LD-CELP for wideband processing. Ordentlich's invention is an extension of the

weighting filter used in conventional narrowband CELP to wideband signals, and forming a specific weighting filter to be used in LD-CELP.

105. LD-CELP is very different than CELP since it operates in backward adaptive fashion using different signals and transmitting different parameters, as is illustrated by (See for example [13] O. Gottesman and Y. Shoham, "Real-Time Implementation of High Quality 32 kbps Wideband Speech LD-CELP Coder," EUROSPEECH'93, 1993) :

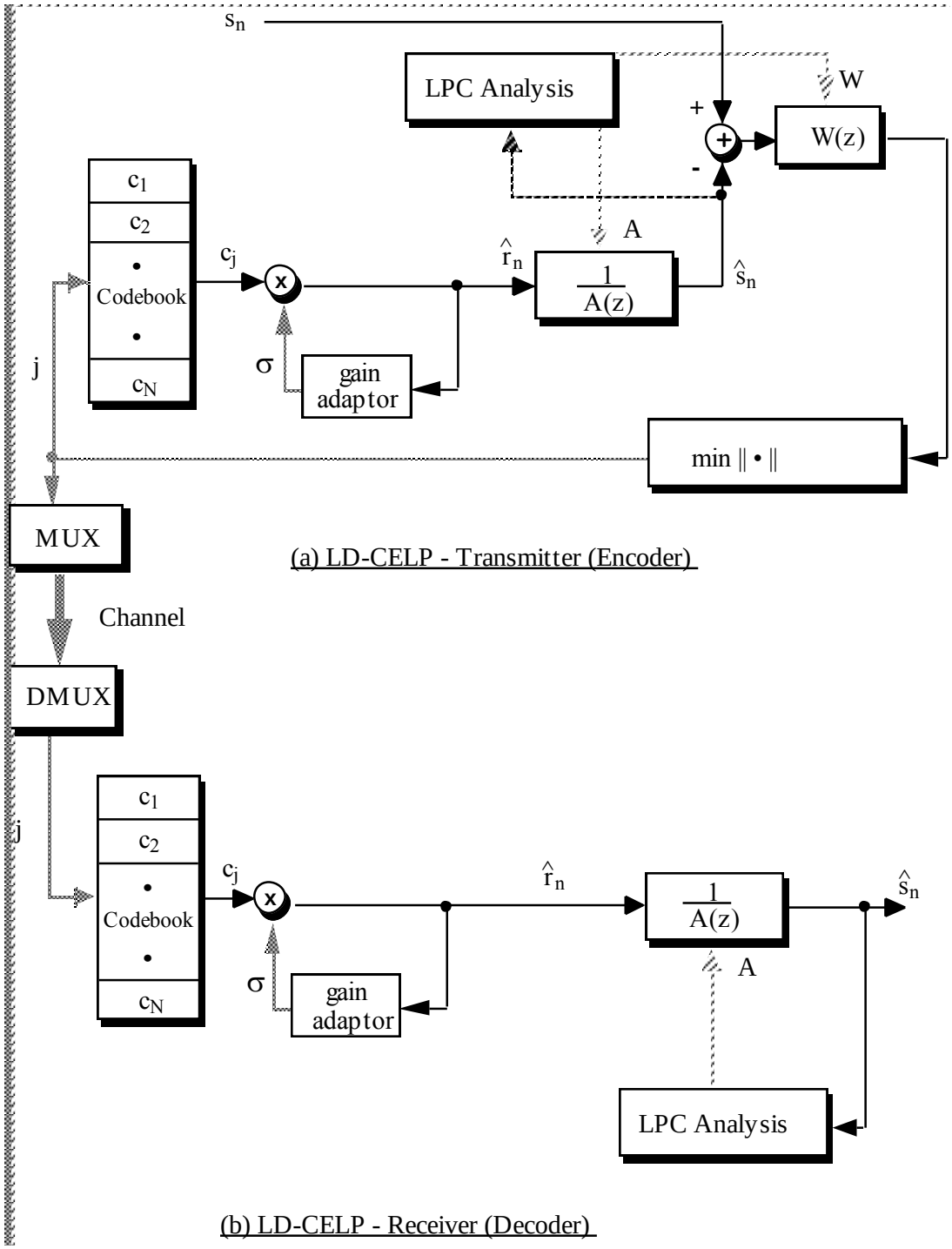


**Figure 15.** LD-CELP [13]

106. Petitioner stated that skilled in the art would be motivated to combine Ozawa and Ordentlich, but did not explain why would such person be motivated

and how specifically that can be done. Similarly to the G.729, it is my opinion that also Ordentlich cannot be combined with Ozawa since Ozawa is not a typical CELP implementation, and it calculates the FCB vector before the ACB vector, and its distortion searched for 3 gains, namely one ACB gain and two FCB gains. These are two different algorithms, that use different signals, operate in different order, and optimizes different distortions and equations. Furthermore, since Ordentlich LD-CELP operates for wideband, and Ozawa operates for narrowband, they cannot be combined. Different sampling rate means different codebooks, different dimensions, and different processing.

107. One cannot combine LD-CELP ("Ordentlich") with CELP ("Ozawa"), as these algorithms operate differently. For example, the CELP is designed to code frames of 20-30 ms with subframes of 5-7 ms, using lookahead, while LD-CELP uses extremely short subframes of less than 1 ms. Additionally, while in CELP parameters are computed for the frame using the input speech and are transmitted, in LD-CELP LPC and gain parameters are computed in backward adaptive fashion using the output signals and they are not transmitted.

108. It is my opinion, that skilled in the art would have no motivation to combine Ozawa and Ordentlich.

## X. GROUNDS FOR PATENTABILITY FOR EACH CLAIM OF THE '521 PATENT

109. In the following sections I will explain why the alleged four ground described in the Petition and Dr. Johnson's report do not disclose the claimed '521 invention.

### A. Ground 1: Ozawa Does Not Anticipate Claims 1, 2, 5, 7, 8, 55, 56, 59, 62

110. As explained above, Ozawa does not disclose Claim 1.a "at least two signal paths associated to respective sets of pitch codebook parameters representative of said digitized input audio data." The signal path alleged by Petitioner is preselected in open-loop once per frame based of the LPC analysis and based on the short-term prediction error. Only one path actually uses the ACB, and the other one bypass the ACB and has no pitch codebook parameters at all. Once either path is selected for the frame, that single path is used for all the subframes, and if ACB search path is selected, then the ACB search is performed only for that path. If the other path is selected, then the ACB is bypassed and no ACB search is performed and no ACB parameters are used. In short, only one signal path can be used and associated to respective sets of pitch codebook parameters.

111. As explained above. Ozawa also does not disclose 1.a.i "each signal path comprises a pitch prediction error calculating device for calculating a pitch

prediction error of said pitch codevector from said pitch codebook search device." Since there is at most one signal path associated with the ACB search at a given time, there could not be "each signal path comprises a pitch prediction error …" as there is at most one path for which pitch prediction error is calculated, and at least one path that pitch prediction error is not calculated. For clarity, the mode and the path were preselected based on the frame's open loop short-term prediction error, while the pitch is optimized in subframe's closed-loop long-term prediction distortion. In other words, in Ozawa the selection of the path has nothing to do with the pitch prediction error, and the pitch codebook search.

112. As explained above. Ozawa also does not disclose 1.a.i "at least one of said at least two signal paths comprises a filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device of said at least one signal path." As explained, in Ozawa there is no more than one signal path for a given subframe, only one of the possible paths has filter for filtering the pitch codevector, since there is no pitch codevector used at the bypass path and therefore there is no filter for filtering the pitch codevector at the bypass path. Subsequently, there is no pitch prediction error calculating device rather than some quantization distortion defined by equation (6) and (7). The joint ACB and FCB quantization distortion (6) is calculated only when the selected path uses the pitch codebook, and FCB quantization distortion (7) is used when no pitch

codebook is used. The distortion is therefore calculated for at most one signal path.

113. As explained above, Ozawa also does not disclose 1.b "a selector for comparing the pitch prediction errors calculated in said at least two signal paths, for choosing the signal path having the lowest calculated pitch prediction error and for selecting the set of pitch codebook parameters associated to the chosen signal path." In Ozawa no pitch prediction is calculated, rather than the joint quantization distortion of equation (6). The distortion is calculated for at most a single path, that is preselected for the frame, and for the reasons explained above there are no "said at least two signal paths". When distortion (7) is calculated the pitch codebook is not involved at all. The pitch codebook does not select path having lowest calculated pitch prediction, since it uses at most a single path or bypassed and no need there for selecting a path that was already preselected.

114. Ozawa does not disclose dependent Claim 2: "A pitch analysis device as defined in claim 1, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device." First, Claim 1 is not disclosed. Furthermore for the reasons above, for each subframe there is at most one path, and there is no "at least two signal paths". In Ozawa, the path that bypasses the filter also bypasses the pitch codebook and therefore has no pitch codevector at all,

as also illustrated in Ozawa Figure 2 (see also Figure 14 above). For the reasons explained above there is no "pitch prediction error calculating device", rather than distortion (7) that is calculated when there is no filtering, and evidently no pitch codevector is even part of equation (7). Petitioner and Dr. Johnson mixed up Ozawa's path with no pitch codevector and no filter, and the '521's "paths comprises no filter for filtering the pitch codevector" which claims that there is pitch codevector but it is simply not filtered in that path, as illustrated in the '521 Patent Figure 3 - the path from pitch codevector generator 302 to convolution block $304^{(0)}$, shown below:

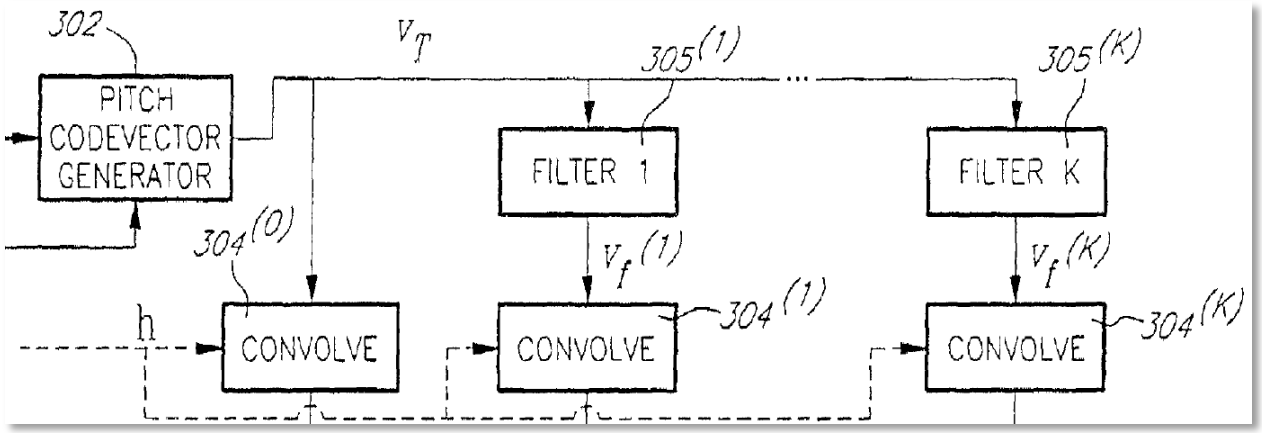


**Figure 16.** '521 Figure 3 "paths comprises no filter for filtering the pitch codevector"

115. Ozawa does not disclose dependent Claim 5 "A pitch analysis device as defined in claim 1, wherein each pitch prediction error calculating device comprises…" First it does not disclose all elements of Claim 1 as explained above.

Then, as explained above, there is no "pitch prediction error calculating device", rather than another distortion measure (6). Then, there is a single such device calculating the joint quantization distortion (6). When the other path is preselected then the FCB quantization distortion in (7) is used without any pitch codebook involved, and all the remaining elements of Claim 7 are not even remotely involved.

116. Ozawa does not disclose dependent Claim 7 "A pitch analysis device as defined in claim 5, wherein: …" Ozawa does not disclose Claim 7 since Claim 5 is not disclosed. Furthermore Ozawa does not disclose the first element "each of said filters of the plurality of signal paths" since only one path has a filter, only one path is preselected, and there is no plurality of signal paths per subframe. Also, Ozawa does not disclose " pitch code book parameters comprise the filter index" since the mode is predetermined in open-loop for the whole frame based on the LPC and it is not determined by the subframe's pitch codebook search, and hence cannot be considered a pitch codebook parameter.

117. Ozawa does not disclose dependent Claim 8, "A pitch analysis device as defined in claim 1, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for

choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error." Ozawa does not disclose Claim 8 since Claim 1 is not disclosed. Also, there is no disclosed "pitch prediction error calculating device", the joint quantization distortion (6) is not "pitch prediction error", and FCB quantization distortion (7) does not even include pitch codevector. The distortion is calculated for a single path at any given subframe. There is no choice of signal path as part of the pitch codebook search, and the path is already preselected. There is no disclosed choice of path by choosing any lowest calculated thing.

118. Ozawa does not disclose method Claim 55, "A pitch analysis method for producing a set of pitch codebook parameters, comprising…" for at least similar reasons explained above for Claim 1.

119. Ozawa does not disclose dependent Claim 56, for similar reasons explained above for Claim 1 and Claim 2.

120. Ozawa does not disclose dependent Claim 59, "A pitch analysis method as defined in claim 55, wherein calculating a pitch prediction error in each signal

121. path comprises:…" , for at least similar reasons explained above for Claim 1 and Claim 5.

122. Ozawa does not disclose dependent Claim 62 "A pitch analysis method as defined in claim 55, wherein calculating said pitch prediction error, in each signal path, comprises calculating an energy of the corresponding pitch prediction error, and wherein comparing the pitch prediction errors comprises comparing the energies of said pitch prediction errors of the different signal paths and choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error," for at least similar reasons explained above for Claim 1 and Claim 8.

123. My opinion is that Ozawa also does not disclose any equivalent of the claims or their elements.

### B. Ground 2: Ozawa In View Of G.729 Does Not Render Obvious Claims 6, 15, 42, And 60.

#### 1. Disclosure In G.729

124. As explained above the G.729 operates very differently than Ozawa, and cannot be combined with it. Petitioner stated that skilled in the art would be motivated to combine Ozawa and G.729, but did not explain why would such person be motivated and how specifically that can be done. The G.729 discloses gain calculation that is different from the one disclosed on Ozawa, uses different signals, different codebooks and quantizers, and operation order. Although the equations appear somewhat similar, the gain $g_p$ calculated in G.729 is calculated for a single path. Also, Ozawa uses pitch codebook gain during only one path, and

no pitch codebook on the other path, where G.729 has a single path that is used at all times. Claim 6 on the other hand discloses the gain calculation $b^{(j)}$ for index j where j=0, 1, 2, ... , K, and K corresponds to a number of signal paths. Ozawa and G.729 do not disclose such gain calculation over a number of signal paths.

125. Ozawa combined with G.729 does not disclose dependent Claim 15, "An encoder as defined in claim 14, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation: $b^{(j)}$ =.... where j=0, 1, 2, ... , K, and K corresponds to a number of signal paths…," for at least similar reasons explained above for Claim 1 and Claim 6.

126. Ozawa combined with G.729 does not disclose dependent Claim 42, "A network element as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation $b^{(j)}$ =.... where j=0, 1, 2, ... , K, and K corresponds to a number of signal paths,…" for at least similar reasons explained above for Claim 1 and Claim 6.

127. Ozawa combined with G.729 does not disclose dependent Claim 60: "A pitch analysis method as defined in claim 59, wherein said pitch gain calculation comprises calculating said pitch gain $b^{(j)}$ using the relation: $b^{(j)}$ =.... where j=0, 1, 2, ... , K, and K corresponds to a number of signal paths, …," for at least similar reasons explained above for Claim 1 and Claim 6.

### 2. No Motivation To Combine G.729 With Ozawa

128. Petitioner stated that skilled in the art would be motivated to combine Ozawa and G.729, but did not explain why would such person be motivated and how specifically that can be done. It is my opinion, that skilled in the art would have no motivation to combine Ozawa and G.729. These are two different algorithms, that use different signals, operate in different order, and optimizes different distortions and equations.

129. It is therefore my opinion that Claims 6, 15, 42 and 60 are not disclosed by combining Ozawa and G.729 ACB gain calculator.

130. My opinion is that Ozawa combined with G.729 also does not disclose any equivalent of the claims or their elements.

## C. Ground 3: Ozawa In View Of Ordentlich Does Not Renders Obvious) Claims 10,11,14,17, 28, 29, 32, 35, 37, 38, 41, And 44.

### 1. Disclosure In Ordentlich

131. As explained above Ordentlich disclosed LD-CELP algorithm that operates very differently than Ozawa and G.729. For example in LD-CELP the LPC are calculated differently, in backward adaptive using the output signal, and everything that is based on the LPC such as the weighting and impulse response would be calculated differently, using different signals, in different order, at different frame sizes, and using different vector lengths.

132. Ordentlich illustrates in Figure 1 merely a possible conventional CELP on a high level, "the basic structure of conventional CELP", as part of the introduction with references to narrowband CELP systems, and then describes LD-CELP for wideband processing. Ordentlich's invention is an extension of the weighting filter used in conventional narrowband CELP to wideband signals, and forming a specific weighting filter to be used in LD-CELP. Then Ordentlich introduces the wideband LD-CELP, while Ozawa is a unconventional CELP that operates on narrowband speech.

133. Ozawa combined with Ordentlich does not disclose dependent Claim 10: "An encoder having a pitch analysis device as in claim 1 for encoding a wideband input signal, said encoder comprising…".

134. Additional element 10.d.ii "said pitch analysis device responsive to the pitch codevector for selecting, from said sets of pitch codebook parameters, the set of pitch codebook parameters associated to the signal path having the lowest calculated pitch prediction error;" is not disclosed by Ozawa combined with Ordentlich, as explained above neither Ozawa does not disclose "calculated pitch prediction error". Ozawa does not disclose more than one path per subframe. The path is selected based on the LPC and the pitch codebook is not involved with the path selection.

135. Also element 10.f: "a signal forming device for producing an encoded wide band signal comprising the set of pitch codebook parameters associated to the signal path having the lowest pitch prediction error, said innovative codebook parameters, and said linear prediction synthesis filter coefficients," is not disclosed by Ozawa combined with Ordentlich, for the same reasons explained above.

Ozawa combined with Ordentlich does not disclose dependent Claim 11: "An encoder as defined in claim 10, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device.". As explained above Claim 10 is not disclosed. Additionally, for at least similar reasons explained above for Claim 1 and Claim 2.

136. Ozawa combined with Ordentlich does not disclose dependent Claim 14: "An encoder as defined in claim 10, wherein each pitch prediction error calculating device comprises". As explained above Claim 10 is not disclosed. Furthermore, Ozawa combined with Ordentlich does not disclose element 14.d "a combiner circuit for combining the amplified convolved pitch codevector with the pitch search target vector to thereby produce the pitch prediction error." The distortions (6) and (7) are not pitch prediction errors, and (7) is not even involved with the pitch codebook.

137. Ozawa combined with Ordentlich does not disclose dependent Claim 17: "An encoder as defined in claim 10, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error." First, as explained Claim 10 is not disclosed. Additionally, for at least similar reasons explained above for Claim 1 and Claim 8.

138. Ozawa combined with Ordentlich does not disclose dependent Claim 28: "A cellular mobile transmitter/receiver unit, comprising..." Element 28.a is not disclosed at least for the similar reasons Claim 10 is not disclosed.

139. Ozawa combined with Ordentlich does not disclose dependent Claim 29: "A cellular mobile transmitter/receiver unit as defined in claim 28, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device." First for the reason Claim 28 is not disclosed. Additionally, for at least similar reasons explained above for Claim 1 and Claim 2.

140. Ozawa combined with Ordentlich does not disclose dependent Claim 32.

141. Ozawa combined with Ordentlich does not disclose dependent Claim 35: "A cellular mobile transmitter/receiver unit as defined in claim 28, wherein said pitch prediction error calculating device of each signal path…." First for the reason Claim 28 is not disclosed. Additionally, for at least similar reasons explained above for Claim 1 and Claim 8.

142. Ozawa combined with Ordentlich does not disclose dependent Claim 37: "A network element, comprising: a transmitter including an encoder for encoding a wideband signal as recited in claim 10 and a transmission circuit for transmitting the encoded wideband signal...," for at least the reason Claim 10 is not disclosed.

143. Ozawa combined with Ordentlich does not disclose dependent Claim 38: "A network element as defined in claim 37, wherein one of said at least two signal paths comprises no filter for filtering the pitch codevector before supplying said pitch codevector to the pitch prediction error calculating device." First for the reason Claim 37 is not disclosed. Additionally, for at least similar reasons explained above for Claim 1 and Claim 2.

144. Ozawa combined with Ordentlich does not disclose dependent Claim 41: "A network element as defined in claim 37, wherein each pitch prediction error calculating device comprises:.." First for the reason Claim 37 is not disclosed. Additionally, for at least similar reasons explained above for Claim 1 and Claim 5.

145. Ozawa combined with Ordentlich does not disclose dependent Claim 44: "A network element as defined in claim 37, wherein said pitch prediction error calculating device of each signal path comprises means for calculating an energy of the corresponding pitch prediction error, and wherein said selector comprises means for comparing the energies of said pitch prediction errors of the different signal paths and for choosing as the signal path having the lowest calculated pitch prediction error the signal path having the lowest calculated energy of the pitch prediction error." First for the reason Claim 37 is not disclosed. Additionally, for at least similar reasons explained above for Claim 1 and Claim 8.

#### 2. No Motivation To Combine Ordentlich With Ozawa

146. As explained above, Ozawa and Ordentlich are so different systems that there is no motivation to combine them. It is therefore my opinion that Claims 10,11,14,17, 28, 29, 32, 35, 37, 38, 41, and 44 are not disclosed by combining Ozawa and Ordentlich.

147. My opinion is that Ozawa combined with Ordentlich also does not disclose any equivalent of the claims or their elements.

### D. Ground 4: Ozawa In View Of G.729 And Further In View Of Ordentlich Does Not Render Obvious Claims 15, 33, And 42.

#### 1. Disclosure In Ordentlich

148. As explained above Ordentlich disclosed LD-CELP algorithm that operates very differently than Ozawa and G.729. For example in LD-CELP the

LPC are calculated differently, in backward adaptive using the output signal, and everything that is based on the LPC such as the weighting and impulse response would be calculated differently, using different signals, at different frame sizes, and using different vector lengths.

149. Figure 1 of Ordentlich is merely "the basic structure of conventional CELP" as part of the introduction with references to narrowband CELP systems while Ozawa is not a conventional CELP. Then Ordentlich introduces the wideband LD-CELP, while Ozawa operates on narrowband speech.

2. **Disclosure In G.729**

150. As explained above the G.729 operates very differently than Ozawa, and cannot be combined with it. Petitioner stated that skilled in the art would be motivated to combine Ozawa and G.729, but did not explain why would such person be motivated and how specifically that can be done. The G.729 discloses gain calculation that is different from the one disclosed on Ozawa, uses different signals, different codebooks and quantizers, and operation order. Although the equations appear somewhat similar, the gain $g_p$ calculated in G.729 is calculated for a single path. Also, Ozawa uses pitch codebook gain during only one path, and no pitch codebook on the other path, where G.729 has a single path that is used at all times. Claim 6 on the other hand discloses the gain calculation $b^{(j)}$ for index j

where j=0, 1, 2, ... , K, and K corresponds to a number of signal paths. Ozawa and G.729 do not disclose such gain calculation over a number of signal paths.

151. Ozawa combined with G.729 and Ordentlich does not disclose dependent Claim 15: "An encoder as defined in claim 14, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation: $b^{(j)}$ =.... where j=0, 1, 2, ... , K, and K corresponds to a number of signal paths…," for at least similar reasons explained above for Claim 1 and Claim 6.

152. Ozawa combined with G.729 and Ordentlich does not disclose dependent Claim 33: "A cellular mobile transmitter/receiver unit as defined in claim 32, wherein said pitch gain calculator comprises a means for calculating said pitch gain b(j) using the relation: b(j) =.... where j=0, 1, 2, ... , K, and K corresponds to a number of signal paths…," for at least similar reasons explained above for Claim 1 and Claim 6.

153. Ozawa combined with G.729 and Ordentlich does not disclose dependent Claim 42: "A network element as defined in claim 41, wherein said pitch gain calculator comprises a means for calculating said pitch gain $b^{(j)}$ using the relation $b^{(j)}$ =.... where j=0, 1, 2, ... , K, and K corresponds to a number of signal paths,…" for at least similar reasons explained above for Claim 1 and Claim 6.

### 3. No Motivation To Combine G.729 And Ordentlich With Ozawa

154. It is my opinion, that skilled in the art would have no motivation to combine Ozawa with G.729 and Ordentlich. These are three different algorithms, that use different signals, operate in different order, and optimize different distortions and equations. As explained above, Ozawa and Ordentlich are so different systems that there is no motivation to combine them. As explained above, Ozawa and G.729 are so different systems that there is no motivation to combine them. As explained above, G.729 and Ordentlich are so different systems that there is no motivation to combine them.

155. It is therefore my opinion that Claims 15, 33, and 42 are not disclosed by combining Ozawa, G729, and Ordentlich.

156. My opinion is that Ozawa combined with G729, and Ordentlich also does not disclose any equivalent of the claims or their elements.

## XI. SUMMARY

### A. Summary of Invalidity Analysis

157. Based on my analysis of all the materials discussed above, as well as my education and years of experience in the field, it is my opinion that the references listed in the table below collectively do not disclose all elements of the claims shown therein, and that, where I cite multiple references for a single claim,

there is an appropriate basis to combine the references, all as detailed further below.

| Ground | Claims | Description |
|---|---|---|
| 1 | 1, 2, 5, 7, 8, 55, 56, 59, and 62 | Non obvious under § 102(b) over Ozawa, Ex. 1006 |
| 2 | 6, 15, 42, and 60 | Non obvious under § 103 over Ozawa in view of G.729, Ex. 1006 and Ex. 1004 |
| 3 | 10, 11, 14, 17, 28, 29, 32, 35, 37, 38, 41, and 44 | Non obvious under § 103 over Ozawa in view of Ordentlich, Ex. 1006 and Ex. 1007 |
| 4 | 15, 33, and 42 | Non obvious under § 103(A) over Ozawa in view of G.729 and Ordentlich, Ex. 1006, Ex. 1004, and Ex. 1007 |

158.

## XII. CONCLUSION

159. Based on the foregoing analysis, it is my opinion that not all the elements of each of Claims 1, 2, 5, 7, 8, 55, 56, 59, and 62 of the '521 Patent are disclosed by, or would have been an expected variant of Ozawa. It is my opinion

that not every element of each of Claims 6, 15, 42, and 60 of the '521 Patent are disclosed by, or would have been an expected variant of, the combination of Ozawa and G.729. It is my opinion that not every element of each of Claims 10, 11, 14, 17, 28, 29, 32, 35, 37, 38, 41, and 44 of the '521 Patent is disclosed by, or would not have been an expected variant of, the combination of Ozawa, and Ordentlich. It is my opinion that not every element of each of Claims 15, 33, and 42 of the '521 Patent is not disclosed by, or would have been an expected variant of, the combination of Ozawa, G.729, and/or Ordentlich.

160. A person of ordinary skill in the art would not have been motivated to combine Ozawa, G.729, and/or Ordentlich. My opinion is that Ozawa combined with G729, and/or Ordentlich also does not disclose any equivalent of the claims or their elements.

161. In signing this Declaration, I recognize that it will be filed as evidence in a contested case before the Patent Trial and Appeal Board of the United States Patent and Trademark Office. I also recognize that I may be subject to cross-examination within the United States. If cross-examination is required, I will appear within the United States during the time allotted for cross examination.

162. The undersigned being warned that willful false statements and the like are punishable by fine or imprisonment, or both, under 18 U.S.C. § 1001,

declares under penalty of perjury under the laws of the United States of America that the foregoing is true and correct.

Executed on June 17, 2016 in Los Altos Hills, California.

Respectfully submitted,

Oded Gottesman, Ph.D.

## Bibliography

[1] J. D. Markel, and A. H. Gray, Jr., *Linear Prediction of Speech*, Springer Verlag, Berlin, 1976.

[2] J. Makhoul, "Linear prediction: A tutorial review," *Proc. IEEE*, pp. 561-580, April 1975.

[3] B. S. Atal, and M. R. Schroeder, "Predictive coding of speech signals and subjective error criteria," *IEEE. Trans. Speech Signal. Proc.*, ASSP Vol. 27, no. 3, pp. 247-254, 1979.

[4] B. S. Atal, and M. R. Schroeder, "A new model of LPC excitation for production of natural sounding speech at low bit-rates," in *Proc. Int. Conf. Comm*, Paris, pp. 1610-1613, 1984.

[5] B. S. Atal, and M. R. Schroeder, "Stochastic coding of speech at very low bit-rate," in *Proc. Int. Conf. Acoust., Speech, Signal Processing.*, Paris, pp. 614-617, 1982. P. Kroon, E. F. Deprettre, and R. J. Sluyter, "Regular-pulse excitation – A novel approach to effective and efficient multipulse coding of speech," *IEEE Trans. Acoust. Speech Signal Proc.*, vol. 34, no. 5, pp. 1054-1063, 1986.

[6] ITU-T Recommendation G.728, "G.728 - Coding of speech at 16 kbit/s using low-delay code excited linear prediction (LD-CELP)," ITU-T, 1990.

[7] TIA/EIA Recommendation IS-127, "Enhanced variable rate codec, speech service option 3 for wideband spread spectrum digital systems," September 9, 1996.

[8] P. Kroon, E. F. Deprettre, and R. J. Sluyter, "Regular-pulse excitation – A novel approach to effective and efficient multipulse coding of speech," *IEEE Trans. Acoust. Speech Signal Proc.*, vol. 34, no. 5, pp. 1054-1063, 1986.

[9] W. B. Kleijn, "Generalized analysis-by synthesis coding and its application to pitch prediction," in *Proc. IEEE Int. Conf. Acoust., Speech, Signal Processing.*, Vol. I, pp. 337-340, 1992.

[10] W. B. Kleijn, P. Kroon , L Cellario, and D. Sereno, "A 5.85 kb/s CELP algorithm for cellular applications," in *Proc. IEEE Int. Conf. Acoust., Speech, Signal Processing.*, Vol. II, pp. 596-599, 1993.

[11] W. B. Kleijn, P. Kroon, and D. Nahumi, "The RCELP speech coding algorithm," *European Trans. on Telecomm., vol 4, no. 5*, pp. 573-5782 1994.

[12] W. B. Kleijn, and K. K. Paliwal, *Speech Coding Synthesis*, Elsevier Science B. V., Amsterdam, 1995.

[13] O. Gottesman and Y. Shoham, "*Real-Time Implementation of High Quality 32 kbps Wideband Speech LD-CELP Coder*," EUROSPEECH'93, 1993

[14] R. Salami; C. Laflamme ; J. -P. Adoul ; A. Kataoka, "*Design and description of CS-ACELP: a toll quality 8 kb/s speech coder*," IEEE Transactions on Speech and Audio Processing (Volume:6 , Issue: 2 ), March 1998.

# Dr. Oded Gottesman

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## A. RESUME

## 1. EXPERIENCE

**Technology Expert for Patent Infringement** September 2001 – Present

Providing world-class expertise and experience in working with leading law firms, providing expert witness services, prior art searching, and related consulting for patent infringement, trade secret disputes, and related matters. Among areas of expertise are DSP, telecommunications, cellular networks, packetized networks, VoIP, streaming systems, real-time systems, timing and synchronization in cellular and other networks, location and position systems, transcoding between links and networks having multiple timing sources and data rates, source coding, channel coding, error control coding, algorithms, speech and audio processing, signal compression, algorithms, software, hardware and implementation. Served as expert witness in several cellular network and smartphone / tablet device related cases, including network operation, protocols, layers, signal processing, user / human interface, and applications. Served as expert witness & consultant for law firms and clients including Alston & Bird, Cooley Godward Kronish, Cummins, Dechert, Echelon, Fenwick & West, Fish & Richardson, Hill Kertscher & Wharton, Kirkland & Ellis, MagicJack, Sidley Austin, Weir & Partners, Woodcock & Washburn, Microsoft, Nokia, Parts Geek, Samsung, TruePosition, and Wiav.

**Compandent, Inc., Santa Barbara CA** February 2001 – Present

Starting and running a small startup company developing for outsourcing of DSP R&D and implementation, and licensing intellectual property. Developing technologies for various communication networks, including cellular network, satellite network, telephone networks, packetized networks, and military networks. R&D and implementations of technologies related to streaming-media systems. DSP contracts for SignalCom (Microsoft company) and the Department of Defense (DoD, DARPA, NATO). DSP hardware and software. R&D and implementation DSP projects - low-rate speech coder, noise cancellation, echo cancellation, and communication for wireless applications and streaming media applications for environments that include various timing synchronizations, error control coding (such as FEC), modem (various modulations), etc.. Voice communication product development for cellular networks, wireless networks, packetized networks, VoIP, satellite communication, military and security radio devices, etc.. Exploring advanced robust positioning methods that combined methods such as GPS and cellular network signals.. Developing proprietary timing and synchronization solution for links and systems with multiple data timings. Development and implementation of adaptive algorithms, networking protocols, multi-tasking, embedded systems, timing and synchronization, etc.. Inventor of a patent for transmission of signals over degraded channel such as cellular network, and packetized network. The

company developed, manages the production, and sells line of proprietary electronic boards and standalone devices, or can be embedded in other devices or systems having different timing sources, in synchronous and asynchronous links.

**DSP Communications, Israel,** January 1994 - September 1995

- Title: *Speech Coding Group Leader*

  DSPC is a cellular technologies manufacturer of cellular phone chipset that includes the cellular network interface and the voice compression. R&D and implementation of a proprietary multi-rate speech coder at 4-8 kbps at floating-point and fixed-point. Leading the DSP development and implementation of PDC's PSI-CELP and VSELP standards, and noise cancellation & echo cancellation, on the company's DSP chipset (including FEC, and modem/modulation, masked on TI's TMS320c54x). Developing and implementation of a noise-canceller for digital-cellular phone. The PDC chipset has become the flag ship of the company. Oded's contribution was the key in turning DSPC, from a poor company (in term speech coding expertise) in 1994, into a very successful one, leading to a $1.6 billion acquisition by Intel.

**Optibase - VCON, Israel,** April - December 1993

- Title: *Audio Group Manager*

  Optibase/VCON is a manufacturer of streaming-media systems. Managing and leading DSP audio team in implementation of DSP video-teleconferencing and streaming multi-media system that combined G.728, G.722, and Acoustic Echo Canceller (AEC). R&D and implementation of a proprietary AEC for teleconferencing. The company developed and sold its self developed teleconferencing electronic boards.

**AT&T Bell Labs, Murray Hill,** January - September 1992

- Title: *DSP Consultant*

  M.Sc. degree research, and thesis writing, in the Signal Processing Research Department.

**Vibration Specialty Corporation: Philadelphia, PA**, January 1991 - August 1995

- Title: *DSP Consultant – Real-Time DSP Software*

  Development and implementation of the real-time DSP software for the SpectraViB, a 4-channel spectrum analyzer and data collector. The company developed and sold its self developed teleconferencing electronic boards.

**Efrat - Comverse Technology, Israel,** October 1989 - September 1990

- Title: *DSP Engineer of Trilogue voice mail system for telephone & cellular networks*
  - Implemented interface with telephone network.
  - Designed and implemented DSP algorithms (speech coding, echo cancellation, telephone, DTMF, etc.) using the Texas-Instruments TMS320c25, and the NEC µPD77p25 processors.
  - Managed the **hardware-software integration** cooperating with the hardware engineers and the upper software's programmers.
  - The group also implemented noise canceller that was based on Ephraim-Malah algorithm.

**Computers**

- IBM PC (under Windows 7, Win XP, Win 2000, Win NT, Win 3.1, DOS and Linux), SUN (under Unix/Solaris), Android, and Macintosh OS,
- Programming: C++, C, PASCAL, FORTRAN, various DSP Assembly codes,
- PC system manager of the Signal Compression Laboratory at UCSB,
- Computer networking,
- Programming of Data-Base applications,
- Programming of Internet protocols, sockets, TCP-IP, etc.,
- Programming of distributed systems, and embedded systems,
- Programming Android application, and iOS (iPhone, iPad)
- Web site programming including HTML, XML, JAVA, CGI, etc..

## 2. EDUCATION

**Post-Doctoral Research, Univ. of CA Santa Barbara** January 2001 – March 2002

- Title: *Post Doctoral Researcher*
  Researching in the area of Voice-over-Internet Protocol (VoIP), and issues of packetized networks.

- **Ph.D. - University of California at Santa Barbara,** October 1995 – Fall 2000
  Electrical and Computer Engineering Department,
  Santa Barbara, CA 93106, USA
  - Ph.D.: Oct 1995-2000
  - Graduate Student Researcher: Supervised by **Prof. Allen Gersho**
  - Ph.D. Research: "*Analysis-by-Synthesis Waveform Interpolative Coding of Speech at Low Bit Rates*"

- **M.Sc.E.E. - Drexel University,**
  Electrical and Computer Engineering Department,
  32nd and Chestnut Sts. Philadelphia, PA 19104, USA
  - Master of Science: Oct 1992; **Cum Laude**, **GPA: 4.0**
  - M.Sc. Thesis: "*Low Delay CELP Wide Band Speech Coding at 32 kbps*".
  - The research was performed in **AT&T Bell Labs, Murray Hill**.

- **B.Sc.E.E. - Ben-Gurion University,**
  Electrical and Computer Engineering Department,
  P.O.box 653 Beer Sheva 84105, Israel
  - B.Sc.: June 1988; **Cum Laude,** GPA: 3.6; Class Standing: **Top 3%**
  - Electrical Engineer (Majors: DSP, Communications, and Computers)
  - Dean's List: 1987-8, 1986-7 and 1985-6
  - B.Sc. Senior Project: "*A Speaker-Independent Speech Recognition System based on Dynamic Time Warping and HMM*".

## 3. AWARDS

- ✓ **Ericsson-Nokia Best Paper Award** for the paper: "Enhanced Waveform Interpolative Coding at 4 kbps" by Oded Gottesman and Allen Gersho, *IEEE Workshop on Speech Coding*, Finland, 1999.
- ✓ Drexel University: **Cum Laude** for M.Sc.E.E. study, June 1992.
- ✓ The Knesset (**The Israeli Parliament**) -The Education and Culture Committee: Honor for B.Sc.E.E. study, 1988.
- ✓ Ben-Gurion University: **Cum Laude** for B.Sc.E.E. study, June 1988.
- ✓ Ben-Gurion University: **Dean's List**: 1987-8, 1986-7 and 1985-6.

## 4. PRIZES

- ✓ **Compandent Technologies - Winner of the Business Plan Competition** of the Center for Entrepreneurship & Engineering Management (CEEM) at UCSB.

## 5. PATENTS

- *International patents in the area of speech coding, and transmission over degraded channel such as cellular networks and packetized networks.*

## 6. ACTIVITIES

- **IEEE Senior Member- Signal Processing Society, and Communications Society**
- Paper peer review

## 7. OTHER EDUCATION & EXPERIENCE

- **Selected business courses**
- Entrepreneurship,
- Marketing campaigns including, strategy development, Internet based methods, concepts, and advertising material preparation,
- Sales, including correspondence, agreement negotiation and closing deals, customer support.
- Hardware and software project management,
- Study and analysis of new problems and technologies,

## 8. SELECTED PUBLICATIONS

[1] O. Gottesman and A. Gersho, "Enhanced Waveform Interpolative Coding at Low Bit-Rate," in *IEEE Trans. Speech and Audio Processing*, vol. 9, November 2001, pp. 786-798.

[2] O. Gottesman and A. Gersho, " Enhancing Waveform Interpolative Coding with Weighted REW Paramertric Quantization," in *IEEE Workshop on Speech Coding Proc.*, pp. 50-52, September 2000.

[3] O. Gottesman and A. Gersho, "High Quality Enhanced Waveform Interpolative Coding at 2.8 kbps," *IEEE ICASSP'2000*, pp. 1363-1366, June, 2000, Turkey.

[4] O. Gottesman and A. Gersho, "Enhanced Analysis-by-Synthesis Waveform Interpolative Coding at 4 kbps," *EUROSPEECH'99*, pp. 1443-1446, 1999, Hungary.

[5] O. Gottesman and A. Gersho, "Enhanced Waveform Interpolative Coding at 4 kbps," *IEEE Workshop on Speech Coding Proceedings*, pp. 90-92, 1999, Finland.

[6] O. Gottesman, "Dispersion Phase Vector Quantization For Enhancement of Waveform Interpolative Coder," *IEEE ICASSP'99*, vol. 1, pp. 269-272, 1999.

[7] O. Gottesman and Y. Shoham, "Real-Time Implementation of High Quality 32 kbps Wideband Speech LD-CELP Coder," *EUROSPEECH'93*, 1993.

[8] O. Gottesman and Y. Shoham, "Real-Time Implementation of High Quality Wideband-Audio LD-CELP Coder at 32 kbps Using Two-DSP Hardware," *ICSPAT'93*, Newton, MA, USA: DSP Associates, pp.1379-82 vol.2, 1993.

[9] Oded Gottesman "Enhanced waveform interpolative coder," US7643996, Dec 1, 1999.

[10] Oded Gottesman,and Allen Gersho, "REW parametric vector quantization and dual-predictive SEW vector quantization for waveform interpolative coding," US7010482, Filing date Mar 16, 2001.

[11] Oded Gottesman and Allen Gersho, "REW parametric vector quantization and dual-predictive SEW vector quantization for waveform interpolative coding", US7584095, Filing date: Sep 23, 2005.

[12] Oded Gottesman "Redundant compression techniques for transmitting data over degraded communication links and/or storing data on media subject to degradation," US6614370, Jan 24, 2002.

[13] Oded Gottesman, "Dispersion phase vector quantization for enhancement of waveform interpolative coder," Provisional Application No. 60/110,641, December 01, 1998.

[14] Oded Gottesman, "Weighted distortion vector quantization of shift-removed waveform for speech and other signals," Provisional Application No. 60/110,522, December 01, 1998.

[15] Oded Gottesman, "Enhanced Waveform Interpolative Coder," PCT International Application PCT/US99/28449, Corres. to USSN 60/110,522 and 60/110,641, December 01, 1999.

[16] Oded Gottesman, "Enhanced Waveform Interpolative Coder," International Patent Application Serial No. WO 00/33297, June 8, 2000.

[17] Oded Gottesman, "Enhanced Waveform Interpolative Coder," U.S. Patent Application Serial No. 09/831,843, May 14, 2001.

[18] Oded Gottesman, "Enhanced Waveform Interpolative Coder," Japanese Patent Application Serial No. 2000/585864, 2001.

[19] Oded Gottesman, "Enhanced Waveform Interpolative Coder," Chinese Patent Application Serial No. 2000-585864 , 99815704, 2001.

[20] Oded Gottesman, "Enhanced Waveform Interpolative Coder," European Patent Application Serial No. 99962962.9, 2001.

[21] Oded Gottesman and Allen Gersho, "REW parametric vector quantization, and dual-predictive SEW vector quantization for waveform interpolative coding," U.S. Provisional Patent Application No. 60/190,371, March 17 2000.

[22] Oded Gottesman and Allen Gersho, "Analysis-by-synthesis predictive vector quantization, of parametrically represented rapidly-evolving-waveform, and of slowly-evolving waveform, for speech coding and synthesis application," U.S. Provisional patent application, March 17 2000.

[23] Oded Gottesman and Allen Gersho, "REW parametric vector quantization, and dual-predictive SEW vector quantization for waveform interpolative coding," U.S. Patent Application No. 09/811.187, March 16 2001.

[24] Oded Gottesman, "Redundant compression for degraded storage and transmission," U.S. Provisional Patent Application Serial No. 60/264,494, January 26 2001.

[25] Oded Gottesman, "Redundant compression of techniques for transmitting data over degraded communication links and/or storing data on media subject to degradation," U.S. Patent Application Serial No. 10/056,891, January 24 2002.

[26] Oded Gottesman, "Redundant compression of techniques for transmitting data over degraded communication links and/or storing data on media subject to degradation," U.S. Patent Serial No. 6,614,370, September 2 2003.

[27] Oded Gottesman, "System for automated interaction with voice/tone-enabled system or service," U.S. Provisional Patent Application Serial No. 60/361,157, March 01 2002.

[28] Oded Gottesman, "Techniques for interaction with sound-enabled system or service," U.S. Patent Application Serial No. 10/386,174, March 11 2003.

## 9. CONSULTING EMPLOYMENT HISTORY

2001-2004 . AT&T vs. Microsoft (Microsoft, Woodcock Washburn, Philadelphia, PA) , UNITED STATES DISTRICT COURT, SOUTHERN DISTRICT OF NEW YORK, 01 Civ. 4872 (WHP), Patent No. 32,580 (provided source code review, was deposed, case was settled the night before testifying in court).

2003-2004 . Lucent vs. Microsoft (Microsoft, Fish Richardson, San Diego, CA) , (provided source code review, analyzed the products and the patents)

2006-2007 Samsung: Ericsson, Inc. v. Samsung Telecommunications America LLP, et al., In the Matter of Certain Wireless Communication Devices, Components Thereof, and Products Containing the Same, Investigation No. 337-TA-583, consulting expert.

2006-2007 Nokia Corp.: Qualcomm Inc., et al. v. Nokia Corp., et al., 05-cv-2063 (S.D. Cal.), consulting expert.

2008-2009 WiAV Solutions LLC: WiAV Solutions LLC v. Research In Motion, Ltd., et. al., 3:08-cv-485 (E.D. Va.) and related cases, consulting expert.

2009 Landmark Media Enterprises: ShopNTown, LLC v. Landmark Media Enterprises, LLC, 2:08-cv-564 (E.D.Va.), testifying expert (deposed).

2006-2010 TruePosition Inc.: TruePostion Inc. v. Andrew LLC, 05-cv-747-SLR (D. Del.), testifying expert (deposed, testified at trial).

2009-2010 Parts Geek, LLC: U.S. Auto Parts Network, Inc. v. Parts Geek, LLC, 2:09-cv-4609 (C.D. Cal.), consulting expert.

2010-2011 Nokia Corp.: Apple Inc. v. Nokia Corp., In the Matter of Personal Data and Mobile Communications Devices and Related Software, Investigation No. 337-TA-710 (ITC), testifying expert (deposed, testified at trial).

2010-2011 Triangle Software, LLC: Triangle Software, LLC v. Garmin Int'l, Inc., 10-cv-1457 (E.D. Va.), testifying expert (deposed, testified at trial).

2010-2011 eTool Development, Inc.: eTool Development, Inc. v. National Semiconductor Corp., 2:08-cv-196 (E.D. Tex.), consulting expert (deposed).

2010-2014 High Point SARL: High Point SARL v. Sprint Nextel Corp., et al., 2:09-cv-2269 (D. Kan.), consulting expert.

2011-2012 Microsoft Corp.: Microsoft v. Barnes & Noble, In the Matter of Certain Handheld Electronic Computing Devices, Related Software, and Components Thereof, Investigation No. 337-TA-769 (ITC), testifying expert (deposed, testified at trial).

2012-2013 Mosaid Technologies Inc. vs. Sony Ericsson Mobile Comm. (USA), Inc. and HTC America, Inc., C.A. No.: 11-598-SLR, testifying expert (deposed).

2013 Provided declaration on behalf of Microsoft regarding In re Patent No: 6,339,780, Reexam Control No.: 95/002,267.

2013-2014 TruePosition, Inc. v. Polaris Wireless, Inc., Civ. Action No. 1:12-cv-00646 (D. Del.); Polaris Wireless, Inc. v. TruePosition, Inc., Case No. IPR2013-00323 (PTAB), testifying expert (provided source code review).

2013-2014 Core Wireless Licensing S.A.R.L v. Apple, Inc. Case No. 6:12-CV-100, testifying expert, only provided source code review.

2014-2015 NetTalk.com, Inc. v. magicJack Vocaltec Ltd., et al, Case No. 9:12-cv-81022, , currently pending in the United States District Court for the Southern District of Florida.

2015-2016 Toshiba Corp: Andrea Electronics, Corp. v. Toshiba Corp. et al. Investigation No. 337-TA-949 (ITC), (provided source code review, and deposed)

2016-Present Google: Callwave. v. Google (C.A. No. 12-cv-1701-RGA) (Delaware), (not yet testified)

2016-Present Saint Lawrence Communications LLC,: Saint Lawrence Communications LLC,. v. Motorola Mobility et al. Case No. 2:15-cv-000349-JRG (Lead Case), (not yet testified)